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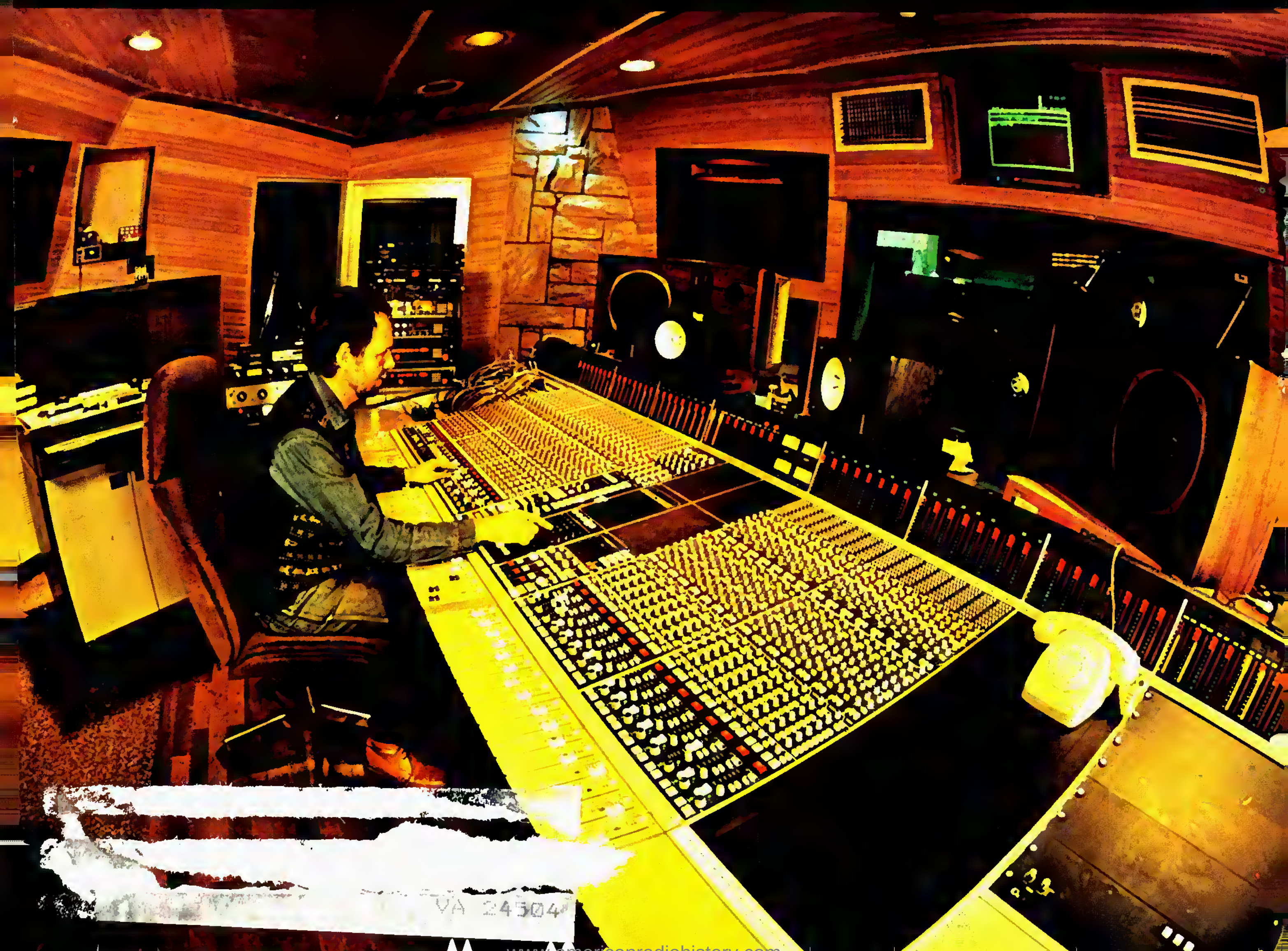

BUILT FOR THE DEMANDS OF PRODUCTION.

In the production business, quality plus speed equals success.

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—Time Code Lock keeps code coming from the Sync head, regardless of the audio monitor mode, so your synchronizer won't get confusing double messages when modes are switched.

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MARCH/APRIL 1987 VOLUME 21 NO.2

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About the Cover

Studio One's control room of AAV-Australia's Music Studios. Bill Taushke, Audio Maintenance Supervisor sits at the console. In our cover article, learn how the hit film "Crocodile Dundee" had its audio married to the picture.

About the 2 to 8trk Cover

Jan Paul Moorhead in his MIDI-controlled Pulse Audio Studio in Long Beach, California.

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CALENDAR 1987

Synergetic Audio Concepts (SynAudCon) is sponsoring a series of two-day seminars dealing with solving audio and acoustic problems.

Demonstrations will include signal alignment, measurements of %ALcons and Rasti, the fundamental differences between impulse and energy time curve measurements, and how to design loudspeaker arrays.

Atlanta, GA--April 22-23, 1987

The Presidential Hotel

Louisville, KY--May 12-13, 1987

Holiday Inn Southwest

Detroit, MI--June 2-3, 1987

To be announced.

Boston, MA--June 10-11, 1987

To be announced.

Synergetic Audio Concepts is also sponsoring a **Studio Designer's Workshop** aimed at sharing design, construction, and proof of performance technology of recording control room design. Staff for the workshop will include **Charles Bilello,** designer, and **Dr. Peter D'Antonio of RPG Diffusors.**

The workshop format will have a heavy emphasis on measurements of recording and control room acoustics, monitor loudspeakers, and diffusion will be used by the instructors to illustrate their lecture material. A live group will perform in the studio during the listening/calibration sessions in the control room. For more information, contact: **Synergetic Audio Concepts, PO Box 1239, Bedford IN, 47421. Tel: 812-275-3853**

TALKBACK

Talkback questions are answered by professionals in the media/audio industries. They all have many years experience and formal education in their respected fields. In establishing a dialogue with audio professionals, **db** invites the audio community to submit questions on audio related problems. Discoveries and experiences are also welcomed.

1) Perhaps you can confirm or disprove a rumor I've heard about using a demagnetizer to get audio heads really clean. Wetting the audio head thoroughly with head cleaner will loosen stray magnetic particles trapped in the "head gap." The demagging, while keeping the head wet with the head cleaner, will pull out the loosened magnetic particles and thereby clean the head. The evaporation time of head cleaner and the lack of physical "gap" in the head gap (where huge chunks of magnetized particles supposedly dwell) cause me to question this cleaning method. What do you think?

2) When I demag my machine, should I: A) Move in slowly, rotate around each and every head, tape guide, etc. and then withdraw slowly to six feet away.; or B) Do I move in slowly, rotate around one head or tape guide, withdraw to six feet slowly and then repeat, moving on the next head or guide until the whole path is demagged? Briefly, am I to make one trip in with my demagger or am I to wear out the carpet six feet around my machine?

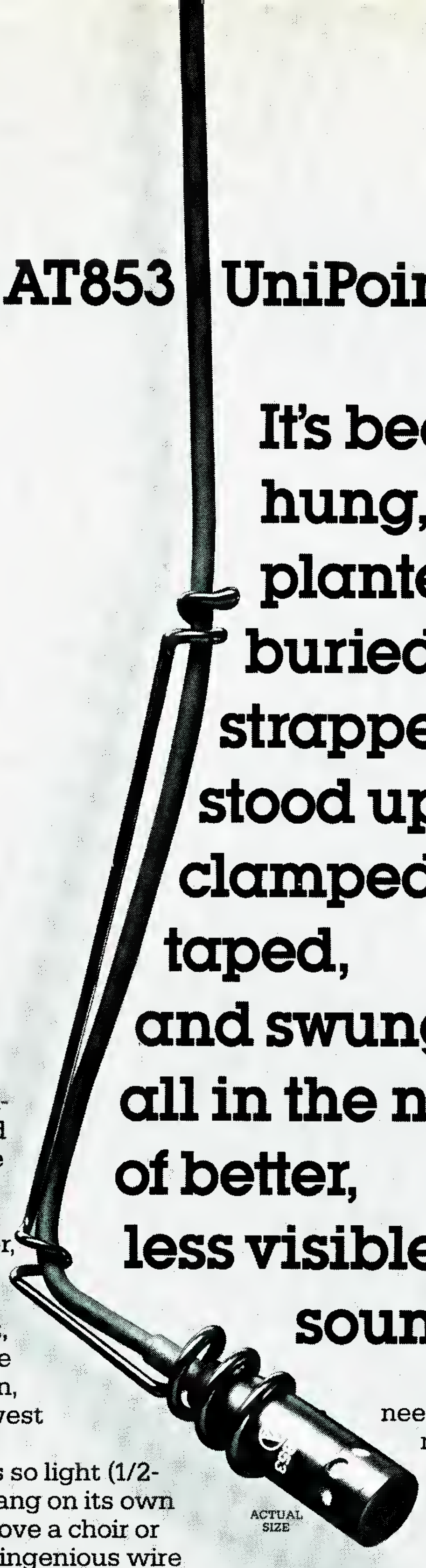
db Reply

1) Wetting the heads is fine for cleaning, however, prolonging this throughout a demagnitizing process should be avoided. When designed properly, heads do not have a physical gap where tiny magnetic men get trapped. Demagging is a dry process and should remain so!

2) The mysterious process of demagging must follow some basic rules. The speed with which one moves in to and away from a head or target object is very important. The key is SLOWLY. Move in like sneaking up on some one, then move out slowly and far away (three feet or so). Now, do it all again. Keep track of your path, trying not to repeat or cross your work. Continue to the end of the tape path. Note: On smaller devices such as cassette decks, some video equipment, etc. it is not recommended to go in and out over and over. In these cases, one trip in and out is sufficient.

The AT853 UniPoint™ Condenser Cardioid

It's been hung, planted, buried, strapped, stood up, clamped, taped, and swung... all in the name of better, less visible sound.



ACTUAL SIZE

The AT853 condenser cardioid is a remarkable microphone. Smaller than your little finger, yet with flat response from 30 to 20,000 Hz, and an effective cardioid pattern, even at the lowest frequencies.

The AT853 is so light (1/2-ounce) it can hang on its own 25-foot cord above a choir or orchestra. The ingenious wire adapter permits pointing it exactly where it's needed without support cables or stays, making the AT853 even less visible.

It also includes a neat stand adapter to instantly convert the AT853 into a desk or floor stand model. Or simply hide it in the bushes, behind props, or wherever superb sound is needed with minimum visibility.

The AT853 is operated by a single 1.5V "N" battery or phantom power. The power module also has a low-frequency rolloff option to solve rumble and room noise problems.

The AT853 is one of a family of six UniPoint ultra-miniature condenser microphones. Each with special features to solve the toughest sound pickup problems, plus professional reliability. And all from the innovators at Audio-Technica.

The AT853 may be hard to see, but it's great to listen to. Arrange for a hands-on test today.

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Circle 14 on Reader Service Card

Letters

Dear Editor,

I'm writing in reference to Jim Fiore's article, "High-Performance Direct Box" in the Jan/Feb, 1987 issue. In the article he describes the circuit well, but he doesn't give a component listing. I would like to build a few of these boxes and I would appreciate if you could send me a list.

Sincerely,
Ed Krupski
Soundworks Digital
254 W. 54th Street
New York, NY 10019

Dear Editor,

I enjoyed the Jan/Feb, 87 issue. One article in particular, "High-Performance Direct Box," caught my attention. I love building new "toys" but am not a hot technician. I'd like to try the direct box, but your writer Jim Fiore gave no resistor or cap values. I'm not that hot at claculating them myself.

Sincerely,
Doug Water
P.O. Box 870
Collegedale, TN 37315

Thanks for the inquiry concerning Jim Fiore's article. The ommission of the parts list was our mistake, not Jim Fiori's. We are sorry if that blunder has caused an inconvenience to any of our readers. Illustrated below is the complete parts list.

Dear Editor,

I really appreciated your article on building the direct box in the Jan/Feb issue of db. I would like to know if the artwork for the foil side of the board reproduced full sized? db has altered such artwork in previous articles, and the spacing of the pins for the IC look a bit too big on this one. If you would, please enclose a photocopy of the correct-sized artwork.

Mark C. Anderson
Lake Avenue Productions
2707 Irvine Ave.
Bemidji, MN 56601

In response to your question about the graphics on the circuit board... Yes, the board was reproduced to actual size. I guess you might have to stretch the legs a tad on the I.C.. So far, most of our readers have had no problem.

Dear Editor,

Love your magazine, however, I have a problem. At this point I'm not sure when my subscription expires. I'm sending along my address labels and I hope that they will help.

Sincerely,
Jim Betros

According to the subscription code on your mailing label, your subscription expires in 1992. To understand the subscription code, it is as follows: The series of numbers or letters is our "match code" which helps us find you in our computer. It begins with the letter "A" in the U.S., "TC" in Canada, "Z" in South America and "X" everywhere else.

The four digits that follow is the month of the issue (i.e. 0386 is Mar/Apr 1986), the next four digits is the expiration date of your subscription. However, it is reversed in order, so 8801 means Jan/Feb 1988 is your last issue.

Parts List

R1 1k ohm-----C1 330nF(0.33u) 50V,Polypropolyne
R2 470K-----C2 100nF(0.1u) 50V, Polypropolene
R3 470K-----C3 330nF(0.33u)50V,Polypropolene
R4 1M-----C4 1uF 16V,Tantalum
R5 1K-----C5 33uF 16V,Aluminum Eletrolytic
R6 100-----C6 1uF 16V,Tantalum
R7 10K
R8 47K-----D1-D4 1N914 small signal diode
R9 10K-----S1 DPDT (DPST if available)
R10 1K-----S2 SPST
R11 100-----IC1 NE5532 Dual Op Amp (Signetics)
All resistors 1% metal film, 1/8 watt
Misc: 1/4" jacks, 3 pin xlr jack, chassis,wire,etc.

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.

Complete PA Systems

Look to Fostex for any and all of your PA needs. Complete systems or individual components. High quality sound from input to output.

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Circle 15 on Reader Service Card

On Taxes

Mark E. Battersby

Personal Deductions for Home Studios Engineers Under TRA '86

Last fall's Tax Reform Act of 1986, drastically changed the way most musicians and sound engineers will conduct their home studio or music-related business operations. Fortunately, lower tax rates will make it less important to qualify the music-related activity as a tax business–good news in light of the newly tightened hobby/business rules.

The presumption that engaging in a given activity for profit is more difficult to meet under the new tax rules. For instance, in order for an activity to be deemed as one which is engaged in for profit after 1986, the gross income derived from such an activity must exceed the deductions attributable to that activity for three (up from two) or more tax years in a period of five consecutive tax years.

In other words, in order for the Internal Revenue Service to accept an activity as a business for tax purposes, it is now necessary to show three profitable years out of five instead of the old two out of five. Naturally, if sufficient profit years don't exist, the old ground rules for proving profit motivation or business intent remain intact.

Changes in the personal arena mean that hobby–related expenses and income will also be treated differently. A good example of these differences is provided by the fact that the itemized deductions for hobby expenses (at least to the extent of income derived from the hobby activity) will now be subject to a new two percent floor on miscellaneous itemized deductions.

This means that supply expense incurred by a person who earns hobby income from operating a home recording studio or a music-related activity is tax deductible not only to the extent of hobby income but also must be included in itemized deductions. Home studio expenses must be aggregated with other miscellaneous deductions and exceed two percent of the individual's adjusted gross income before having any impact on the taxable income of the taxpayer.

For those individuals whose studio-related activities are strictly hobbies, the new miscellaneous deduction rules will have a big impact. The two-percent floor is only a small part. Let's take a closer look at that two-percent floor on deductions itemized on the personal income tax return.

In tax years beginning after 1986, certain miscellaneous itemized deductions, including investment expenses, will be deductible only if the total amount of those deductions exceed two percent of the taxpayer's adjusted income. Fortunately (or unfortunately), both taxes and interest expenses are exempt from this two percent floor which applies to all other miscellaneous itemized deductions including hobby expenses and employee business expenses.

For tax years beginning after 1986, home sound studio owners will no longer be entitled to claim an itemized deduction for state and local taxes. Although itemized deductions for state and local real property taxes and income tax remain, the loss of this deduction for sales taxes will mean that the out-of-pocket cost of hobby equipment and supplies will increase.

Currently, a catch-all deduction is provided for taxes paid or accrued in a tax year in connection with a trade or business or producing income will be sharply curtailed. Those taxes paid or accrued in connection with the acquisition or disposition of property will henceforth be added to the book value of the acquired property. On the other side of the transaction, taxes will be subtracted from the amount realized in a disposition. Under this new provision, the amount of sales tax paid on the acquisition of depreciable property for use in a business is added to the basis of the property and is treated as part of the cost for depreciation purposes.

As already mentioned, itemized deductions for interest payments are not subject to the new two-percent floor on miscellaneous deductions. Unfortunately, after 1986, individuals face other special limitations on the deductibility of interest. Beginning in 1987, deductions for personal interest on car loans, loans to purchase studio equipment and for other personal expenditures will be partially disallowed, with total disallowance becoming effective by 1991.

Interest paid on a debt secured by an individual's principal residence or by a second residence will continue to be deductible to the extent that the debt amount does not exceed the purchase price (plus cost of improvements) of the residence. Home mortgage interest on a debt in excess of the purchase price plus improvements up to the residence's fair market value, will be tax deductible only if the debt is incurred for educational or medical purposes.

New limitations on investment interest will also be phased in over five years beginning in tax years starting after December 31, 1986. By the end of the phase-in period, an individual's deduction for investment interest will be limited to the amount of the taxpayer's net investment income.

Interest that is subject to the new investment interest limitations is defined as being interest on debt incurred or continued to purchase or carry property held for investment. Property held for investment, at least according to our lawmakers, includes any property that produces income of the following types: interest, dividends, annuities or royalties not derived in the ordinary course of trade or business.

Investment interest does not include qualified residence interest or interest that is taken into account in computing income or loss from a so-called "passive" activity. Investment interest, also, does not include interest allowable to a rental real estate activity in which the taxpayer actively participates.

Net investment income, which will be the eventual cap on the deduction of investment interest, is defined in the tax rules simply as the excess of investment income over investment expense. Investment income is the sum of interest, dividends, rents, royalties and net capital gains resulting from the dispositontion of property held for investment–but only to the extent that they are not derived from trade or business.

Interest that is disallowed because of the investment interest limitations continues to be allowed as a carryover to later years and is deducted to the extent of the limitation in the carryover year.

The new restrictions on personal and home mortgage interest and investment interest do not come into full force until tax years beginning in 1991. Until then music and sound home studio owners may deduct the full amount of interest allowed under the old law minus an increasing percentage of the interest that is disallowed under full application of the new rules.

Finally, the Tax Reform Act of 1986 adopts new language, effective for tax years beginning after 1986 that overrules two Tax Court decisions concerning the deductions for the business use of an individual's home. It should also be noted that the deduction of an employee's home office expenses is covered by the new two percent floor on miscellaneous itemized deductions.

Ordinarily, an individual may not deduct any expenses attributable to the use of his or her home for business purposes unless those expenses are clearly attributable to a proportion of the home that is used exclusively and on a regular basis:

(1) As the taxpayer's principal place of business;

(2) As a place to meet patients, clients or customers;

(3) In connection with an individual's business (in the case of a separate structure).

As an employee, an individual cannot deduct home office expenses unless the use of the home is for the convenience of the employer. In addition, the deduction is further limited to gross income derived from the individual's employment, reduced by deductions having no connection with the business such as taxes and interest.

In order to circumvent these limits on the deduction of home office expenses, some taxpayers rented a portion of their home to their employers or business. This ploy was approved by the U.S. Tax Court in 1985. The Tax Reform Act of 1986 takes this opportunity away beginning in 1987.

In addition, if a taxpayer qualifies for a home office deduction, the amount which may be deducted will soon be reduced. Beginning in 1987, the home office expense deduction will be limited to the gross income derived from the business use of the home reduced by deductions having no connection with the business (taxes, interest, etc.) and further reduced by expenditures that the individual makes that are not allowable to the use of the home, such as expenses for supplies and wages.

Although the new tax law supposedly makes taxes less of a factor in our everyday lives, it is obvious from these changes that both the hobby/business activity and the personal income tax returns will be affected. Thus, it is still necessary to keep one eye on taxes...all year long. ■

Ad Ventures

Brian Battles

I'VE MOVED!

I don't know if it looks any different to you, but this is my first Ad Ventures column in Colorado. Since moving I've noticed that the air's thinner, but I'm not. I'm blaming everything on the holidays – the Thanksgiving/ Christmas/New Year's eating and drinking binge has enough momentum to carry me along as far as the Memorial Day/Father's Day/Flag Day/ Fourth of July/Bastille Day binge.

If you've been grinding out commercials for the past few months you probably have some momentum of your own going. Isn't this stuff easy once you get started? (Isn't everything?) The only trouble with success is that it makes you busier. A common problem in business is that the most successful individuals get the most work.

While building your little company's reputation as a hot commercial production plant you will undoubtedly increase your volume of output. This is fine, except watch out for one thing: your identity. An advertisement is supposed to draw attention to the sponsor, not the producer or agency. Your job is to be transparent. Provide the means for the client to get his message out over the airwaves, but don't create a "signature" for yourself. Muhummad Ali could very easily have been describing a superior commercial recording facility when he stated "float like a butterfly, sting like a bee." Nobody should be able to pin down your style and recognize your ads, because then your client's message loses impact and you become the star of someone else's show. But your work still should stand out.

As I once mentioned, I am no composer, so you'll rarely get any technical advice on music theory from me. But I can pass along an observation: It is easier to write jingles that sound original to the average listener than it is to change your speaking voice. Therefore, to avoid being stereotyped, what you need are lots of announcers. If you are in a position to recommend, select or directly hire voice-over people, you should maintain a fairly massive stable of talent. To find these professionals, let your local grapevine know that you're hiring announcers. You'll be up to your shins in audition tapes in no time. Pop an ad in the classifieds and you will have to rent a warehouse to hold the demos. No, you won't receive cassettes from Hoyt Axton or Mason Adams (these guys have more work than they can handle, thank you. What you will get is a ton of material from amateur theater troupers, local broadcasters, and turkeys who think that their CB radio experience qualifies them to be the spokesman for local Fiduciary Bank & Trust. Have patience! If it wasn't for the latter, you would probably have to purchase blank tapes to record your demos.

As a producer, you must know a few things about hiring announcers, even if you only hold responsibility for giving advice to your client. First, good ones come cheaper than you might think. Local folks can get anywhere from $50.00 to $200.00 for a single spot. Depending on your specific market, this is usually a flat flee with no residuals. If you deal with unionized talent, there can be a complicated minimum scales, percentages based on the number of times a spot airs, and so forth. Working with an independent announcer who does this type of work to augment his or her regular job, you'll normally find that you can negotiate fairly wide ranges of compensation.

"I normally get you a hundred bucks for a sixty-second radio voicer, right? Well, La Geaque Boutique will use for their whole Back To School campaign of twenty five spots if you'll take $2,250.00."

"Gee, I don't know, after all, as a professional I set my fees based on the fact that I sound good, my voice pulls, I'm always on time for sessions, I usually only need one or two takes, and I think I'm worth the full $2,500.00."

"Okay, we can probably put you on at least half a dozen of the ads, and we'll just find some other..."

"Of course, I'm always open to discount package rates for high volume clients."

"No, no, don't compromise your integrity. I'll get you three or four spots. That's a few C-notes."

No, I insist. This project calls for someone with my experience. I'll do it for two grand."

"Eighteen hundred."

"It's a deal!"

I am not saying you should rape announcers. Heck no; I'm one of em! Just understand that you can get good people even on what you may consider a limited budget. In fact, you can't afford not to hire fresh, hot talent, because you won't go anywhere if your portfolio consists of a couple of dozen snappy, unique and sparkling jingles all wrapped around one voice.

There are other benefits of working with a variety of voice-over people. You meet some interesting folks, hear lots of juicy inside gossip, maybe pick-up a production tip or two, and find out what the market offers in terms of ability, selection and cost. You might even get a few clues about the competition to spice up your strategy file. ■

N/DYM means better sensitivity

N/DYM design means more microphone signal to cut through the electrical noise of mixers and processors for system "signal-to-noise" that equals even expensive studio microphones.

N/DYM means lower distortion

Neodymium, the rare earth supermagnet at the heart of every N/DYM mic, not only creates an enormous magnetic field within the gap, the "fringe" flux outside the gap is also intense, surrounding the voice coil with a uniform magnetic field for lower distortion even during peak SPL's.

New N/DYM™ microphones break every Electro-Voice tradition but one. Excellence

When was the last time you used a microphone that performed so well you actually did a double take? You actually said, "Wow! This thing is fantastic."

Chances are it hasn't happened in years. It hasn't happened because even though microphones have been modified and improved gradually over the years, there hasn't been any real breakthrough for over two decades.

The new N/DYM™ microphones are going to make you say "Wow!" This innovative series of vocal and instrument dynamics represents the first genuine advance in microphone performance in nearly a quarter century.

At the heart of this Electro-Voice breakthrough is N/DYM, a totally new microphone technology. N/DYM aligned design uses a rare earth supermagnet that is four times more powerful than conventional dynamic microphone magnets. The power and presence of these N/DYM microphones is anything but traditional. They convert more sound energy into usable signal than any other dynamic microphone. *That's 6 dB hotter than the most popular!*

But the proof is in performance. We know it's not the components but the sound that equates to excellence in your mind. See your Electro-Voice dealer for a demonstration before your next performance.

N/DYM means high end sizzle

With 50 percent more surface area than other designs, the larger N/D diaphragm intercepts *more sound waves* and converts this energy into *more output.* Reinforced to prevent "breakup," the diaphragm reliably couples high-frequency pressures and voice coil movements all the way to 20,000 Hz.

N/DYM means less feedback

Our supercardioid pattern rejects more unwanted off-axis sound than the usual cardioid. And the *unique geometry* of the N/DYM magnetic structure *keeps our pattern supercardioid at all frequencies.*

Electro-Voice® EV
MUSIC MICROPHONES

To learn more about N/D Series microphones, see your Electro-Voice dealer or write Electro-Voice, Inc., Dept. N, 600 Cecil Street, Buchanan, MI 49107.

a MARK IV company

Circle 18 on Reader Service Card

Recording Techniques

Bruce Bartlett

EQUALIZATION

Equalization (eq) has been called the highest example of the recording engineer's art. No other effect is used so much – or misused!

An equalizer is a sophisticated tone control, something like the bass and treble controls on a hi-fi set. Equalization (eq) affects tone quality by boosting or cutting selected frequency bands. That is, it alters the frequency response.

To undertand how an equalizer works, first we need to know what a spectrum is. A musical instrument produces a wide range of frequencies, even when a single note is sounded. These frequencies, including the fundamental and harmonics, are the spectrum of the instrument. The perception of the spectrum is called "tone quality" or "timbre."

If we electronically alter the strength of any portion of the spectrum, we affect the reproduced tone quality. An equalizer controls the level of a particular range of frequencies or frequency band, and so controls the tone quality. For example, a boost (a level increase) at 10 kHz makes most instruments sound bright and crisp. A cut at the same frequency dulls the sound.

Let's describe the major frequency ranges that eq affects. The frequencies defining each range are only approximate:

20 Hz to 200 Hz – Deep bass: Lowest fundamentals of bass instruments, sense of power, truck and air-conditioning rumble.

200 Hz to 500 Hz – Mid-bass: Warmth, fullness, body, fundamentals of many instruments.

500 Hz to 1.5 kHz – Mid-range: Nasal or horn-like colorations.

1.5 kHz to 7 kHz – Upper mid-range: Presence, edge, articulation, definition.

7 kHz to 20 kHz – High frequencies: Treble, brilliance, airiness, crispness.

TYPES OF EQUALIZERS

Equalizers range from simple to complex. The most basic is a bass and treble control *(Figure 1)*. Typically, such a device provides up to 15 dB of boost or cut at 100 Hz (for the low-frequency eq knob) and at 10 kHz (for the high-frequency eq knob) . A multiple-frequency equalizer allows boost or cut at several preset frequencies *(Figure 2)*. A sweepable equalizer *(Figure 3)* lets you tune in on the exact frequency range to boost or cut. Most complex, is a parametric equalizer *(Figure 4)*, which allows continuous adjustment of frequency, boost or cut, and bandwidth – the range of frequencies affected within the selected band.

A graphic equalizer has a row of slide pots, dividing the audible spectrum into 5 to 31 bands. It's usually used for monitor-speaker equalization.

A filter is a form of equalizer that sharply rejects (attenuates) frequencies above or below a certain frequency. For example, a 10 kHz low-pass filter (high-cut filter) removes frequencies above 10 kHz *(Figure 5)*. This reduces hiss-type noise without affecting tone quality as much as a gradual treble roll-off would. A 100 Hz high-pass filter (low-cut filter) attenuates frequencies below 100 Hz, reducing rumble from air conditioning and trucks.

Equalizers also can be classified by the shape of their frequency response curves. With peaking eq, the frequency response in the boosted range looks like a hill or peak (as in *Figure 6)*. If you apply peaking eq at 3 kHz, the response is highest at 3 kHz and falls off at higher and lower frequencies.

With shelving eq, the frequency re-

Figure 1. Bass and treble control action.

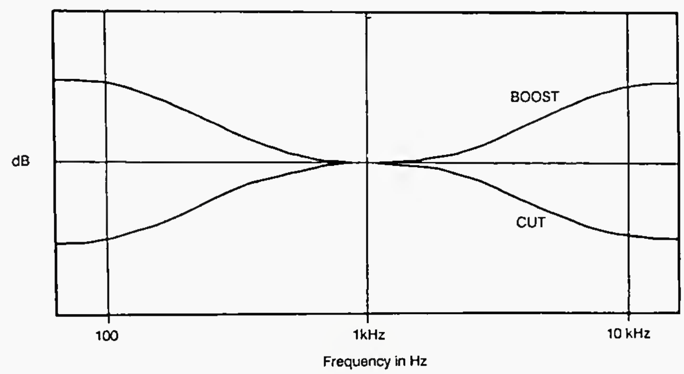


Figure 2. Multiple frequency equalization.

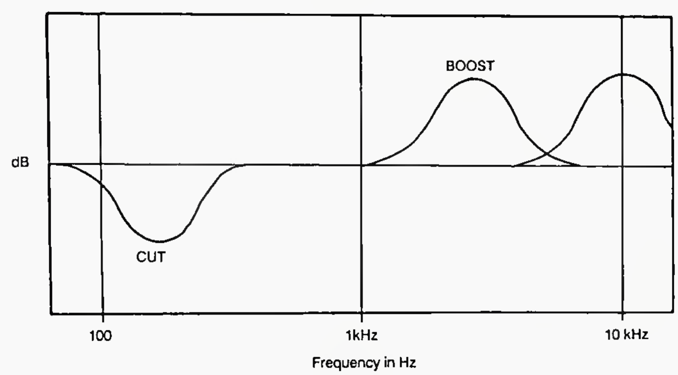
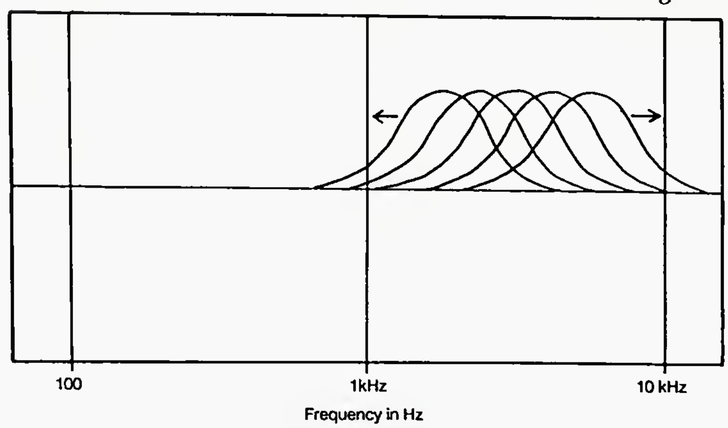


Figure 3. Sweepable equalization.



sponse in the boosted range resembles a shelf (*Figure 7*). If you apply low-frequency shelving eq, all frequencies below the selected one are boosted or cut by the same amount. Similarly, if you apply high-frequency shelving eq, all frequencies above the selected one are boosted or cut by the same amount. Shelving eq is less selective than peaking eq but is less likely to color the tone quality.

USING EQUALIZERS

If you hear a particular tone quality you want to change with eq, estimate what frequency range is emphasized or slighted. For example, if cymbals sound dull or muffled, you know the extreme high frequencies around 10 kHz are weak. So you reach for the high-frequency eq knob, set the frequency to 10 kHz, and turn it up until you hear a difference. Do the cymbals sound better? If not, try a different frequency band or a different amount of boost.

If you have sweepable (continuously variable) eq, you can apply extreme boost or cut, then sweep over a range of frequencies until you find the center frequency that sounds like the ones you want to change. As an example, suppose a kick drum has a boomy or "cardboard" sound. You guess the effect is caused by too much output in the 400-Hz area. After setting the sweepable eq to extreme cut, you sweep the center frequency from 200 to 500 Hz, and find the exact frequency range where the boomy effect disappears.

Different instruments respond differently to the same eq. A boost at 12 kHz will make cymbals sound sharp and crisp, but will not affect a kick drum other than adding noise. The reason is that the spectrum of the cymbals extends well into the highest frequencies, while the spectrum of the kick drum only extends to about 5 kHz. There is no kick-drum output at 12 kHz to be affected by a 12 kHz boost.

Similarly, the presence range is different for different instruments. Here, "presence" means articulation, definition, edge, clarity. The presence range for voice is around 5 kHz, for kick drum around 2.5 kHz to 5 kHz, and for bass guitar around 2 kHz.

When you boost a frequency band, remember that the signal level at those frequencies increases. Whenever you increase the signal level in a particular stage of a mixing console, you run the risk of overloading the following stage. Take care that this does not occur when applying extreme boost with an equalizer.

Another problem associated with ex-

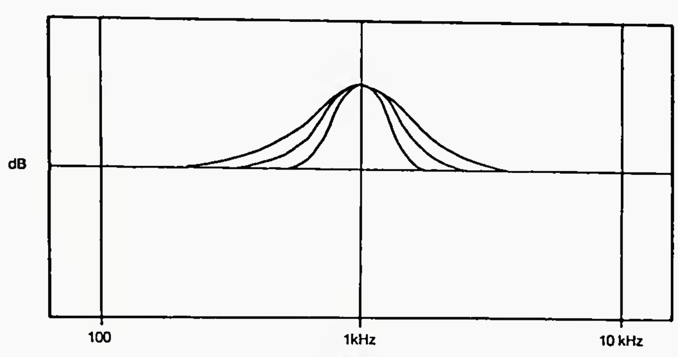


Figure 4. Varying the bandwidth of a parametric equalizer.

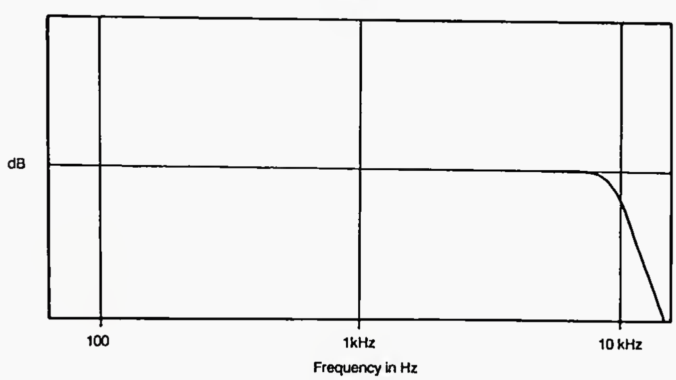
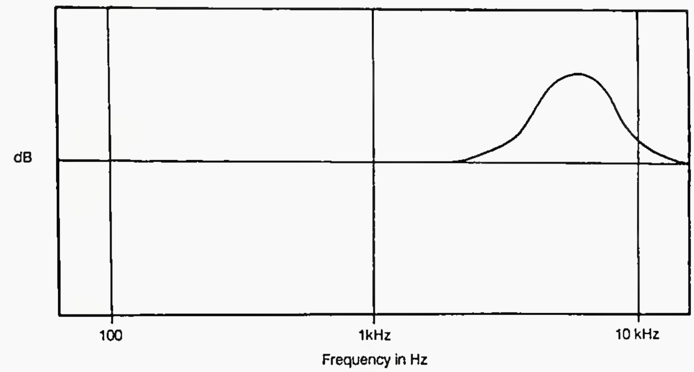


Figure 5. A low-pass filter (-3 dB at 10 kHz).

Figure 6. Peaking equalization at 7 kHz.



treme boost is phase shift. Equalization alters the signal's phase response, which can blur transients. Try to avoid excessive boost. If you need it, chances are the microphone choice was incorrect.

As an alternative to boosting eq, try cutting at another frequency. For example, a muddy sound may be due to excess mid-bass, rather than a lack of upper mid-range. Instead of boosting the upper mid-range to increase clarity, try cutting the mid-bass.

Broadband eq treats music gently–it's difficult to hear as a coloration and causes little phase shift. As a drawback, it is not very frequency selective. Narrowband eq, if overdone, sounds "colored" ; you may hear "ringing" at the boosted frequency. Use broadband eq to correct broadband deficiencies in the program; use narrowband eq to focus on a particular problem area.

Narrowband eq should be used with caution especially at low-to-mid frequencies, because this area covers the fundamental frequencies of musical instruments. The fundamental frequencies determine the pitch of notes. If you apply excessive narrowband boost around, say, 400 Hz, the loudness of the notes being played will vary as the musician goes up or down the scale in that frequency range.

During recording, you may want to apply equalization to each instrument heard individually. Filter out frequencies above and below the range of the instrument. Don't spend too much time with eq until all the instruments are mixed together, because the eq that sounds right on individual instruments may not sound right when all the instruments are heard together.

In creating the desired tonal balance, use eq as a last resort after experimenting with microphone selection and placement. Remember that a microphone is an equalizer too, either because its frequency response is not flat, or its placement exaggerates certain frequencies in the instrument's spectrum.

Should eq be applied during recording or during mixdown? If you're mixing the instruments from "live" to two-track as the music is performed, there is no separate mixdown session, so you apply equalization during recording. If you're assigning several instruments to one track, you must equaalize those instruments during recording because you can't eq them individually during mixdown. If you assign each instrument to its own track, however, the usual practice is to record flat (without eq) and then equalize the track during mixdown.

If the eq used is a bass cut or treble boost, you can obtain a better signal-to-noise ratio by applying eq during recording, rather than during mixdown. Similarly, if the eq used is a treble cut, applying it during mixdown will reduce tape hiss.

USES OF EQ

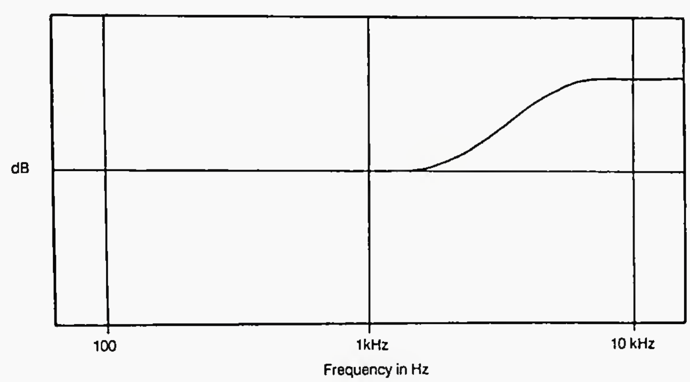
The following is a list of some ways equalization is used:

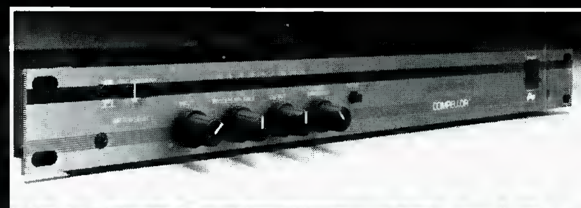
Improving tone quality. You might use a high-frequency rolloff on a sibilant singer to make him or her sound less harsh, or on a direct-recorded electric guitar to take the "edge" off the sound. As another example, boosting 100 Hz on a floor tom gives a fuller sound, or cutting around 250 Hz on a bass guitar aids clarity. The frequency response and placement of microphones affect tone quality too.

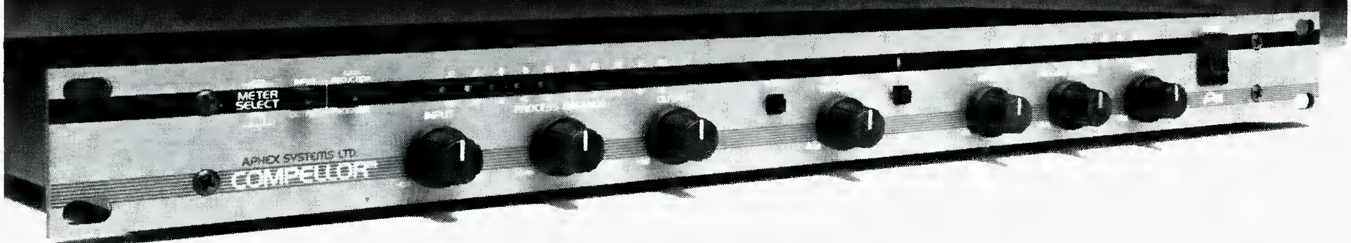
Special production effects. Extreme equalization reduces fidelity, but it also can make interesting sound effects. Sharply rolling off the lows and highs on a voice for instance, gives it a "telephone" sound. An extreme boost at 5 kHz can accent the impact of a snare drum.

Helping a track stand out. A recorded track of an instrument heard by itself may sound very clear, but when it's mixed with other tracks, the clarity may

Figure 7. Shelving equalization at 7 kHz.







disappear. Certain frequencies of the instrument can be covered up or masked by frequencies produced by other instruments. A boost in the "presence" range (say 1.5 kHz to 6 kHz) can help restore presence and clarity. Vocals are often boosted in this range to help them stand out against an instrumental background.

Compensating for response deficiencies. The microphones, tape recorder, monitor speakers, and the mixing example, a high-frequency boost on the

Low-Frequency Boost (below about 500 Hz)

Positive	*Negative*
Powerful (under 200 Hz)	Muddy
Ballsy (under 200 Hz)	Tubby (200-300 Hz)
Heavy (under 200 Hz)	Thumpy
Fat	Boomy
Thick	Barrel-like
Warm	Woody (200-400 Hz)
Robust	
Mellow	
Full	
Woody (200-400 Hz)	

Figure 8(A). Translations of equalization settings into subjective descriptions of tonal balance.

console may help restore a flat response. On the other hand, if a microphone "dies" above a certain frequency, no amount of boost can help it. A bass roll-off can compensate for a microphone's proximity effect. Bass boost can occur with most cardioid microphones when they are placed close to a sound source.

Compensate for microphone placement. Often you must place a microphone very close to an instrument to reject background sounds and leakage. Unfortunately, a close-placed microphone tends to emphasize the part of the instrument nearest to it. The tone quality picked up may not be the same as the instrument as a whole. Equalization can partly compensate for this effect. For example, a guitar mic'ed next to the sound hole sounds bassy because the sound hole radiates strong low frequencies. But a low-frequency roll-off on the console can restore a natural tonal balance.

Reducing noise and leakage. By filtering out frequencies above and below the spectral range of an instrument, you can reject noise and leakage at those frequencies. For instance, you can filter out highs above 5 kHz on a kick drum to reduce cymbal leakage. If this filtering is done during mixdown, it also will reduce tape hiss. Rolling off high frequencies on a flute can reduce breath noise. Filtering out frequencies below 100 Hz on most instruments rejects room rumble and muddy bass.

Compensating for the Fletcher-Munson effect. Fletcher and Munson discovered that the ear is less sensitive to bass and treble at low volumes than at high volumes. So, when you record a very loud instrument and play it back at a lower level, it might lack bass and treble. To restore fullness and presence, you may need to boost the lows around 100 Hz and the highs around 5 kHz to 10 khz when recording loud rock

Flat, Extended Low Frequencies

Positive	*Negative*
Full	Rumbly
Full-bodied	
Rich	
Solid	
Natural	

Low-Frequency Rolloff

Positive	*Negative*
Clean	Thin
	Cold, cool
	Tinny
	Anemic

Mid-Frequency Boost (500 Hz to 7 kHz)

(5 kHz area for most instruments, 1.5-2.5 kHz for bass instruments)

Positive	*Negative*
Present (Presence)	Hollow, muffled (500 Hz)
Punchy	Muddy, horn-like
Edgy	"Aw" sound (500-800 Hz)
Clear	Tinny, telephone-like (1 kHz)
Intelligible	
Articulate	"Er" sound (1.5 kHz)
Defined	Nasal, honky (500 Hz to 3 kHz)
Projected (2 kHz to 3 kHz)	Hard (2 kHz to 4 kHz)
Forward (2 kHz to 3 kHz)	Harsh, strident, piercing (2 kHz to 5 kHz)
	Metallic (3 kHz to 5 kHz, especially 3 kHz)
	Twangy (3 kHz)
	Edgy (3 kHz to 7 kHz)
	Sibilant (4 kHz to 7 kHz)

Flat Mid-Frequencies

Positive	*Negative*
Natural	No Punch
Neutral	"Flat" (lacking character or color)
Smooth	
Musical	

Mid-Frequency Dip

Positive	*Negative*
Mellow	Hollow (500 to 1,000 Hz)
	Disembodied (500 to 1,000 Hz)
	Muffled (5 kHz)
	Muddy (5 kHz)

Figure 8(B). More translations of equalization settings into subjective descriptions of tonal balance.

High-Frequency Boost (above about 7 kHz)

Positive	*Negative*
Trebly	Trebly
Bright	Sizzly (voice)
Crisp	Edgy
Articulate	Glassy
Etched	"Essy" Sibilant
Hot	Steely
Sizzly (cymbals)	String Noise

Flat, Extended High Frequencies

Positive	*Negative*
Open	Too detailed
Airy	Too close
Transparent	
Clear	
Natural	
Neutral	
smooth	
Effortless	
Detailed	

High-Frequency Rolloff

Positive	*Negative*
Mellow	Dull
Round	Restricted
Smooth	Muffled
Easy-on-the-ears	Veiled
Concert-hall-like	Muddy
	Distant

Overall Response

Positive (all flat response)	*Negative*
Natural	Rough, peaky, harsh,
Accurate	colored (non-flat, peaks
Neutral	and dips)
Smooth	Phasey (sharp dips)
Transparent	Cheap (narrow-band)
Effortless	Flat (lacking character—too
Musical	neutral)
Uncolored	
Liquid	

Reverberation or Leakage

Too Little	*Well-Controlled*	*Pleasant*	*Too Much*
Sterile	Clean	Warm	Echoey
Dry	Tight	Rich	Bathroom-sound
Dead		Sumptuous	Muddy
Muffled		Airy	Loose
Thin		Having depth	Washed-out
		"Live"	Barrel-like
		Spacious	Cavernous
		Open	In another room
		Full	Distant
		Bright	Trashy

Figure 8(C). The high-frequency subjective translations.

groups. The louder the group, the more boost is needed. As an alternative, use cardioid microphones with proximity effect for bass boost and a presence peak for treble boost.

Making a pleasing blend. When several instruments are heard together, they sometimes "crowd" each other in the frequency spectrum. That is, it may be difficult to distinguish the instruments by tonal differences. But by equalizing various instruments at different frequencies, you can make their timbres distinct, which results in a more pleasing blend. This procedure also evens out the contribution of each frequency band to the total spectrum, giving a mix that is tonally well balanced.

It's very instructive to spend some time using a graphic equalizer. Play wide-range music or individual instruments through it to become familiar with the tonal effects of each frequency band. Then you'll know what frequency to boost or cut to correct a tonal coloration, and you'll have a better idea what knob to turn to get a "woody" sound or a "brassy" sound.

TRANSLATING TONAL BALANCE INTO EQ SETTINGS

Often communications are poor among musicians, engineers, and producers because they speak different languages when referring to sound quality. For example, what equalization should be used to get "fat" sound or a "thin" sound? What eq can correct a "muddy" sound or a "metallic" sound?

The following chart *(Figure 8)* was made to answer these questions. It translates audio-engineering terms into subjective descriptions of sound quality. Note: These definitions are not universally agreed upon, but they are probably the most common meanings. The positive terms are used when you like the effect; the negative terms are used when you don't! ■

Editorial

This year represents the twentieth year of **db Magazine's** publication. Our very first issue was November/December 1967.

A lot has happened in professional audio in twenty years. We've been so busy chronicling those passing times that we have hardly seen them go by!

In looking over those issues, we've also noticed that even our magazine design has changed over those years. Well, this issue you now hold is no exception. We've switched to a newer type style that looks cleaner, and is much easier to read. As this year's issues go by look for further change.

What will **not** change is the dedication this publication has to the professional in audio. The articles now flowing in (these days frequently by modem or diskette)will carry forth that tradition.

We are also welcoming back into this issue Marshall King who started writing for us almost at the beginning. Marshall will now be contributing a Broadcast Audio column in each issue. As many of you know, Marshall has been with CBS-TV in Los Angeles as an audio mixer for many years. I'm sure you will find these new articles interesting, useful, and (sometimes even) provocative.

To those of you that have picked up **db Magazine** for the first time at NAB, stay with us. If your job is pro audio, this is where you will find the articles.

LZ

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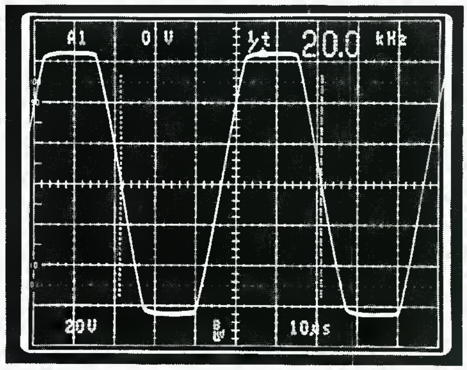

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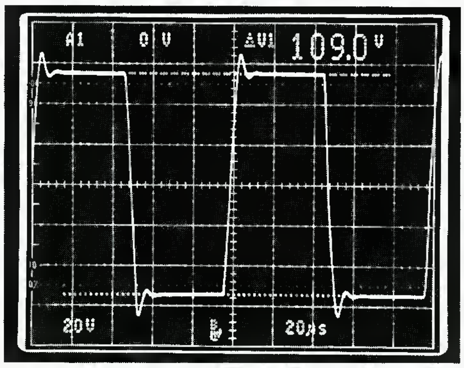

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COLLABORATION: LEGAL PROBLEMS TO AVOID

You finally get together with a songwriter you've been wanting to work with. You bounce a few ideas back and forth. Impressed with each other, you decide to write a few songs. You record rough piano-vocal demos of the songs on a portable cassette recorder. Over the next few months, these problems arise:

1. You wrote all the music for one song and half of the lyrics. The two of you decide to license the song to a television producer for a $1000.00 synchronization fee. Do you automatically get $750.00?

2. For another song, you've written only the music. Within two days after recording the song, you've gotten harsh criticism from all your songwriter friends about the lyrics. You decide to re-write the song with another lyricist, without your collaborator's approval. Can you do this without any remaining obligation to the original lyricist?

3. You want to save one of the songs for only your artist demo tape because the record company told you it's the key to a potential record deal. Your collaborator, needing quick cash to pay the rent, wants to license the song for use in National TV ads for a sexploitation picture. Can you prevent the license?

4. Your collaborator is hitchhiking across Europe and cannot be reached. Michael Jackson calls. He's in the studio with Quincy Jones and they want your song to be the "A" side of the next single, but only if you assign all publishing rights to them within 24 hours. Can you legally do it?

Unfortunately, the answer to all of the questions above is "No!" Now you can understand why the legal effects of collaboration are some of the misunderstood aspects of the business of song writing. The following information should clear up some of the confusion.

HOW IS A "JOINT WORK" CREATED?

A "joint work" under the Copyright Law is a work prepared by two or more authors with the intention that their contributions be merged into inseparable or interdependent parts of a unitary whole." A song is subject to copyright protection when "fixed in any tangible medium of expression...from which it can be perceived, reproduced or otherwise communicated, either directly or with the aid of a machine or device."

This stuffy legalese means that when you create lyrics or music, intending that someday they will be part of a song, your creation becomes part of a "joint work" when it is recorded on tape or written on a lead sheet along with someone else's lyrics or music. The song compromising the joint work is then covered by the copyright laws in its joint form, even if the completed song hasn't been registered in Washington, D.C.. (Registration does provide evidence of your claim to authorship of the song, as well as giving certain legal advantages if you sue someone for infringement.)

You don't have to be physically present when the "merging" of creations occur. In fact, it's not even necessary that you knew who your eventual collaborator would be when you composed your music or wrote your lyrics!

COLLABORATORS EACH OWN UNDIVIDED, EQUAL INTERESTS IN THE WHOLE SONG.

Unless you agree to the contrary, two co-writers each own one-half, three co-writers each one-third, etc. These fractions apply to the entire song. Separate lyrical and musical ownership rights to joint works do not exist without agreement.

There are legal cases that appear to protect songwriters from the claims of minimal contributors. The writing of a few words or composing of a few notes will not normally entitle that songwriter to an equal share, even without an agreement. It is better to be cautious if you have the slightest doubt, making it clear to such a person exactly what his or her financial and ownership interest in the song will be.

CONTRIBUTIONS TO A JOINT WORK CANNOT BE SEPARATED.

This is a source of major trouble. Without the consent of the other writer involved, one writer can't simply remove that writer's contribution and get a new collaborator to replace it, because the original writer will still own up to one-half of the new version. The ownership share of the original collaborator may not be reduced by the addition of new writers without consent. Be careful. This legal prohibition is removed in almost all publishing agreements unless restricted.

NON-EXCLUSIVE LICENSES

Each co-writer can grant non-exclusive licenses without the consent of the other writers. For example, any writer can as that writer accounts to the other writers and pays them for their share of the income received. As a practical matter, however, due to the laws of many foreign countries, most licensees want the signature of all songwriters on any license.

NO INDIVIDUAL ASSIGNEMENTS

No writer may assign an exclusive right to a song without the consent of the other writers. Most importantly, none of the co-writers can deal with the "publishing right" (i.e., ownership and administration rights) of the other writers. No assignment of exclusive rights is valid unless written, so oral agreements between the writers on this issue are unenforceable.

HOW CAN YOU PROTECT YOURSELF?

Naturally, an attorney like me is going to tell you to have a written agreement. Can a complicated agreement work? Isn't a complicated agreement a "turnoff" that could destroy a new collaboration? Definitely. Keep it short and simple.

If you feel you deserve 75% of the song, get it in writing. If you want to be able to take back your lyrics if the song isn't cut within one year, agree to it. If you want to save a song for only your artist tape, tell your co-writer up front and include this requirement in your agreement. If your collaborator likes to go wilderness camping for months at a time, get a "power of attorney" or a written, short-form assignment of all publishing rights, so you can deal effectively with the rights to the song.

Collaboration is often a friendly, casual process. That doesn't mean it has to be an ignorant process as well. ■

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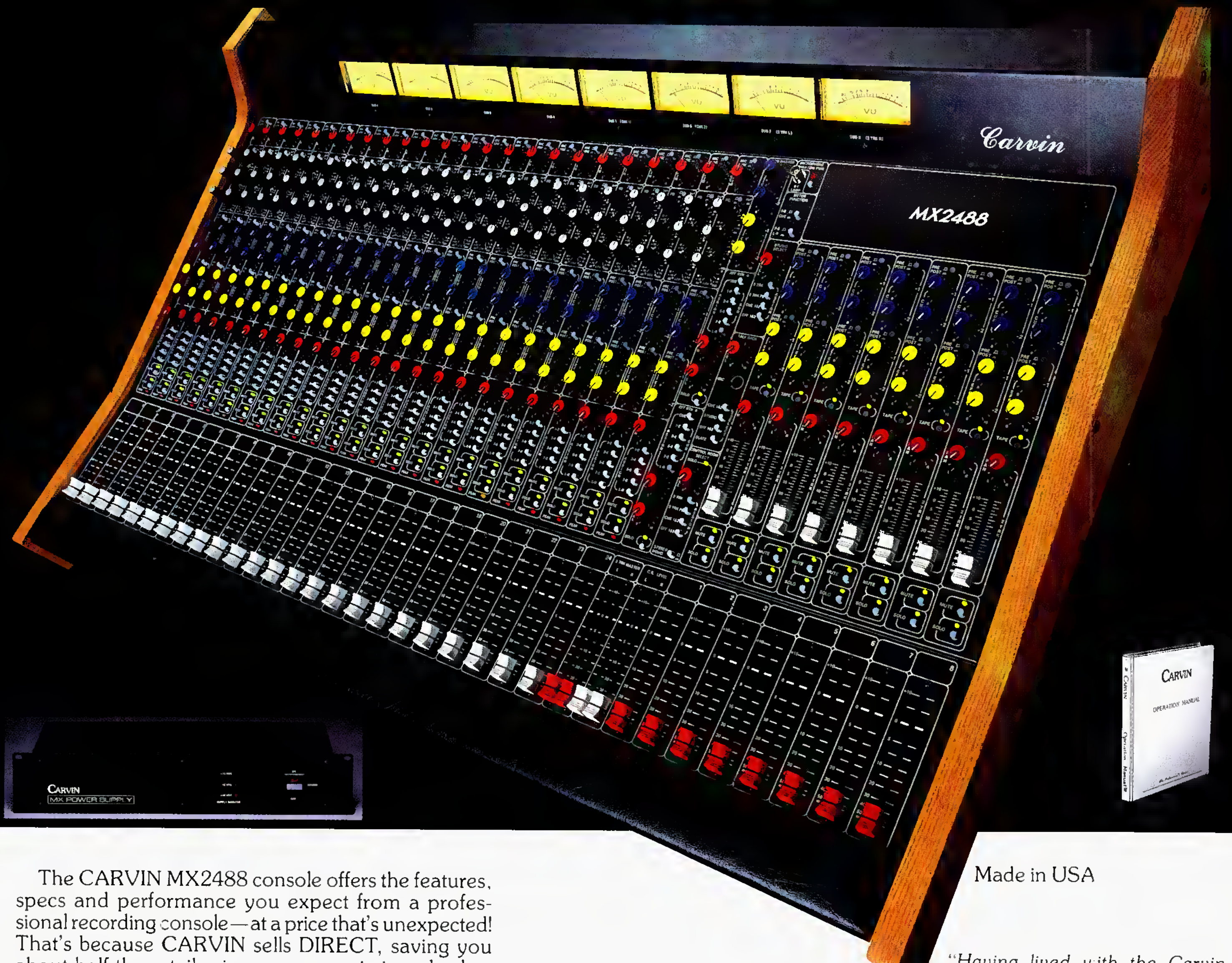

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Len Feldman—db magazine
September/October—1986

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Personal Sound— Pulse Music

Jan Paul Moorhead

The author takes us on a journey through the trial, tribulations and ultimate success of his MIDI production facility.

Upon first getting involved in recording seven years ago, I had no intention of being a producer, technologist, or member of the recordist/MIDI/SMPTE/ computer-slave cult. The initial motivation came because it was a hassle to notate my compositions, wait for an appropriate rehearsal, and then to endure them being misplayed. I was a musician looking for immediate gratification. The solution seemed to be to buy a used Teac 4 track and mixer, an RMI electric piano and a primitive drum machine. Suddenly, I was in seventh heaven. Since then, Pulse Music has gone from 4 to 8 tracks with computer control and to where we are today. The focus on personal use and composition has remained, though the gear has changed a dozen times over.

Today Pulse Music is a 16-track studio with digital mastering and is designed around MIDI, SMPTE and computer control. When personal computers started to become a major force in the world, I splurged and bought an IBM clone. This was my first rude introduction to the rate of technological change. The computer was going down in price faster than I could make payments on it. At the end of the first year you could buy a new one for less than I owed on the original computer. Now, as exasperating as this is, it illustrates a point: If you're going to play the game, you have to ante up. I had some minimal background in computers and became a proficient user quickly. This lead to consulting and computer training work which helped to finance further studio expansion and more than paid for the computer. This illustrates an economic concept mentioned at last year's Los Angeles AES convention—a new piece of gear ought to pay for itself in the first year.

WHAT!? NO BOOTHS? NO GOBOS? NO NEUMANNS? NO...?

MIDI is the primary force directing the design of this studio. We do not bring in drummers or record more than two individuals at a time. We attempt to roll tape as little as possible. Many demos and small projects are recorded "live" via PCM encoding on VHS tape. Using ten synths, two samplers and two drum machines allow us to sequence almost everything and preview or print the entire performance in one pass. To add a little bit more human feel, we have one or two humans play and sequence everything else. This approach is reflected in every aspect of the studio's design. While spending more on outboard gear, synths, samplers and drum machines, a great deal has been saved on mics and the physical surroundings. A friend of mine told me about someone he worked for who put a tremendous amount of money into his facilities, only to have the building sold out from under him. This is not a risk worth taking.

The room is as dead as possible to avoid noise coming in or out. With the exception of vocals, guitar and solo horns, almost everything is recorded by direct input. All ambience comes from the Rev 7 and a couple of delays. We put up heavy duty fiberglass insulation and then built a floor to ceiling frame of 2"x 2"s. Heavy carpets were then hung as limp absorbers. This means we have little economic commitment to the actual studio structure. All of it can be removed and most of it reused. We have to be careful to cover equipment and vacuum regularly (even the walls) to avoid problems with dust. With all the gear in a well insulated small room, air conditioning is absolutely essential. The air conditioning also does some air filtering which eliminates any problems with dust.

SMALL SPACE=LESS$ SPENT= MORE OUTBOARDS

Working in a small environment has allowed us to work with a relatively small power amp, EV Sentry 100A monitors and some small TOA 22-ME monitors. In this room we couldn't do anything other than near-field monitoring even if we wanted to. The little TOAs (similar to Auratones) are very important for getting mixes that sound as good outside the studio. To insure the portability of mixes to other environments, it would be nice to have a frequency analyzer. That is definitely on my wish list.

Because we often print everything in one pass and are working in an extremely dead environment, outboard processing is critical. Upon purchasing synths and samplers, this became an important consideration in selection. The two DSS-1s are a good example. In addition to their other merits they each have two DDLs built in. This saves having to buy more outboard gear.

The concept of "pseudo-live" MIDI recording obviously has important implications for selecting your mixer. The more inputs the merrier. My Ramsa WM-T820 has left me pleased as punch. It's a 20 input board to begin with, but it also allows the monitoring of 20 additional inputs. This additional input bus doesn't have any eq or effects sends but it still can be very useful. If you eq properly when you record and print all of your effects, you can monitor multitrack tape inputs on this bus and send the tracks back in, submixed through two line inputs, and still have 18 inputs left. These can be used for live tracks, MIDI tracks and additional processing. This way, the 20 input board becomes a 38 input board.

Typically, in a MIDI oriented studio "enough inputs is never enough", so I ended up picking up a Fostex 2050 line mixer. The 2050 has no eq or effects sends, so I use it to submix any modules or keyboards that already have on-board processing. Everything comes up on the patch bays (3 inexpensive Teac PB-64s), so if I need to, it's no problem to re-route a keyboard for special processing to the SPX-90, Gatex noise gates, Aphex Aural Exciter, Fostex compressor/ noise gate or DBX compressor.

The quantity of synths allow extensive layering to design sounds. For instance, some of the best string sounds we've achieved involve layering the TX816 with sampled strings and analog strings. If you add the right processing on top of all this, you get a string sound that is "super-real". The more you get committed to this kind of sound the more you have to print these sounds on your multitrack. Otherwise you start to feel that you need an infinite number of synths, samplers, processors and inputs.

ADAPTABILITY IS A MUST

In developing a working situation it's advisable to keep your eyes open for possible spin-off sources of income. For myself, computer and studio consulting have been helpful as well as teaching and writing for various publications. In a sense, information and how to use it is an additional product beyond just audio production or studio rental.

Another project we are involved in is developing samples for the DSS-1. You sit looking at your system, which was designed with one thing in mind, and suddenly realize that with some minor investment you are ready to pursue another source of income. For example, we'll be picking up some cases for my

PCM encoder and VCR; then my engineer Dave Pedneault and I are off to the junk yards of the universe to bang and scrape out some new sounds to amaze the world. Now, stepping out into a new area also implies that you've done your homework and searched out an effective marketer and distributor for whatever kind of racket you've created on disk. Without those two factors accounted for, you stand a good chance of wasting your time.

Finding the right people to work with is, I feel, more important than the right hardware. For one, it allows diversification of talents and technical expertise. Also, finding people you can depend upon makes your whole operation more reliable. In an industry infested with flakes and shaky operators, dependability and consistency make all the difference in your viability.

SOFTWARE

The primary computer we use is a high speed IBM clone. For sequencing, Roger Powell's "Texture" and "MIDI Ensemble" from Sight and Sound do most of the work. I am particularly enamored with the MIDI event editing on "MIDI Ensemble". It's the most flexible and powerful I have seen on the PC. Through some peculiar corporate game playing, the program was put on the back shelf when it was almost finished. However, it is still available if you call Sight and Sound. "Texture" with its drum machine approach to composition is particularly well suited to developing really fierce groove oriented music. It's also very facile for getting music out in a hurry. Another program that I am working with that is very impressive is "MESA" by Kentyn Reynolds from Roland. My first interest in this program was as a transcription program for the purposes of publishing printed music. It is also a formidable sequencer with which all your MIDI event editing can be done from traditional music notation. This is a big step up from the numeric lists that some programs use and only an accountant could love.

We are also using an Atari 1040 ST. Since we are more or less committed to IBM based sequencers, we will probably be using the ST primarily for librarian and sound editing functions. There are now some very powerful sample editing/design programs available for the ST which are of particular interest to our operation. They make the fine point editing of samples a lot easier by using detailed graphic displays. The new ones coming out will also allow the transfer of samples and sounds to different formats. This way the unique properties of the DSS-1 can be made available to other samplers. The market has increased from selling to owners of one type of sampler to owners of many makes of samplers.

STILL GROWING

Pulse Music's move toward video post production seems like a dream job. There is more music used in television and film than anywhere else. If you get to do a horror film you have the opportunity to do some really weird stuff. Honestly, in what other musical area can you get paid for using your elbows on the keys? However, there are a lot of special considerations for this kind of work. Synchronization is a whole new ballgame and FSK just doesn't cut it. We use the Fostex Autolocator with the Fostex E-16 multitrack. The Autolocator is a fine machine whose manual is as bad as the Autolocator is good. That's saying an awful lot for the Autolocator too. If you plan on using one, be sure to get the video training tape with it. The manual is hopeless but the video tape helps a great deal. So far we have avoided purchasing the 4035 controller and other synchronizer boxes that complete the Fostex video post-production system. Using many synths and samplers "pseudo-live", in conjunction with the Autolocator synchronizing SMPTE to MIDI, has proven adequate so far. One of the VCRs will output tach pulses and can be slaved to other controllers. The extra money spent for this capability, though not needed now, will pay off when we move to full synchronization of all equipment. The sequencers we use all implement MIDI song position pointer as well.

Recently, we've undertaken a major revamping of the physical organization of the facilities. There is a problem when you are involved with slow and sustained growth. The small additions start to add up until the superstructure and organization of your setup begins to become ungainly. You eventually have to revamp. Each time things become a little more efficient.

The future looks bright and the output glossy for Pulse Music if we can stay ahead of technical obsolescence, mechanical failure and operator error. ■

AAV-Australia Recording Studios

Ron Tudor

Come with us to the land down under where the largest studio in the southern hemisphere practices world-classs post production and recording.

A land down under of little rivers, crocodiles and road warriors, this may well be your impression of Australia if you took a look at some of the product originated in the recording studios of AAV Australia.

If you haven't already guessed, we're referring to the movie score of "Crocodile Dundee," the multi-hit achievements of an Australian group The Little River Band, and the now famous "Road Warrior" film series.

AAV Australia, situated in downtown South Melbourne, in Victoria, Australia, is the most comprehensive audio/video facility in the southern hemisphere, and a world leader in its ability to deal with all aspects of audio or video.

One of their most recent achievements was recording, in Studio One, the incidental theme and background music for the Australian feature film "Crocodile Dundee," starring Paul Hogan and Linda Kozslowski, which is a huge box office success across Australia and America. The soundtrack was produced at AAV Australia by renowned film composer Peter Best, and engineered by Doug Brady, one of the rising young stars of the AAV team.

This project was technically most significant for its use of a digital multitrack synchronized to pictures for the first time in Australia. We asked Doug Brady to highlight for us some of the unique circumstances and requirements surrounding the production of the soundtrack for "Crocodile Dundee."

"As you know the score for 'Crocodile' was written and produced by Peter Best, one of Australia's most prolific writers. Pete likes to prerecord the click to all cues and know they are in sync before he gets to the first band track. The click was locked to picture time code via an AXE KT 1000 digital metronome/synchronizer so that the timings Peter had prepared in scoring against his 3/4-in. Umatic video could be guaranteed right through to the final lay-up against the film. After each cue click was recorded, we played back the tape against the AXE just to verify sync."

"We then commenced recording rhythm tracks of drums, bass, guitar and piano/DX7. Traditionally, film scores using orchestral sessions are produced up to maybe one hundred pieces at once; however, Pete chose to record section-by-section as most pop albums are recorded these days. This approach allowed us to concentrate on each section independently, so the final result when we mixed it all together was precisely as Pete intended. Section by section then, the score was constructed, a forty-piece string section, followed by eight brass (trombones and trumpets) then four french horns and finally the finishing touches by the percussionists."

A NON-STANDARD INSTRUMENT

"The use of a didgeridoo (a native Australian instrument) which is featured prominently throughout the film brought to bear some unique problems. As an instrument not widely known outside Australia, we were keen to capture its natural timbre while creating the kind of presence necessary to cut through on a range of theater environments. We tried a number of different ways of micing it to get the desired effect. The didgeridoo, a hollowed timber instrument about 5 feet in length, is normally played while seated on the ground cross-legged with one end to the mouth and the other end resting on the floor. We liked the sound of the didgeridoo aimed towards a live room and after several

Figure 1. Front row L-R: Peter Best (composer, arranger), Roger Savage (engineer), Paul Hogan (Dundee Star), Peter Fairman (Dundee Director). Standing L-R: Delvene Delaney, John Cornell (Dundee Producer).

Ron Tudor is with AAV Australia.

Figure 2. AAV Australia's Music Studio Two control room.

microphone changes, settled on the combination of a TLM for warmth and a PZM for brightness. It took a bit of fiddling but we got it spot on."

"One of Peter Best's specifications for the session was that we should run in sync with picture for the entire project, and since Pete has done several sources on digital (Sony PCM 3324) it meant the new challenge of synchronizing digital to video. This was made possible via the use of the Editron 500 A (Audio) synchronizer which we use extensively in our production studios for controlling multitracks, sprocketed dubbers, video machines etc., so we were able to take this product one stage further. To preserve the maximum quality through a minimum of generations, we took the digital music multitrack into our subsidiary film post specialists Soundfirm under the supervision of Roger Savage, and mixed the score on theater surround monitoring through the Dolby Matrixing System rather than speculate as to the position changes that may occur."

When asked how he would sum up AAV Australia from a sound engineer's point of view, Doug had this to say: "AAV has a lot of excellent equipment, and it gives us extreme flexibility in what we can do. We could lock together several 24-track machines. Studio One, our major recording studio, is just fantastic to work in. The SSL and the control room are both terrific. The studio itself has an enormous amount of versatility for different ambience and recording techniques."

WHY IS THIS STUDIO DIFFERENT?

When asked what makes AAV different, Doug replied, "We're the only place in Melbourne that has digital 24-track, 1610 digital mastering and apart from that, the coffee runs hot!"

Unlike most other recording studios around the world, AAV Australia show-cases complete production facilities for all forms of electronic media. This complex houses six sound recording studios, ranging from a fully blown SSL with digital control room and adjoining studio space capable of seating a symphony orchestra, to the more intimately proportioned Studio Two catering to the needs of the smaller group's album requirements and the advertising industry, to sweetening and post-production facilities renowned for their television sound tracks and radio commercial production.

The cornerstone of the organization is its audio facilities. However, not only can AAV Australia boast its total audio set-up, but in other areas such as video production and post-production we have made investments into many millions of dollars.

The company also operates an extensive video cassette duplication plant catering for the needs of the home movie market on video tape. The main clients of this facility are CBS/Fox Video (South Pacific) Pty Ltd and RCA/Columbia Pictures/Hoyts Video Pty Ltd.

The present state of the Australian dollar creates an ideal

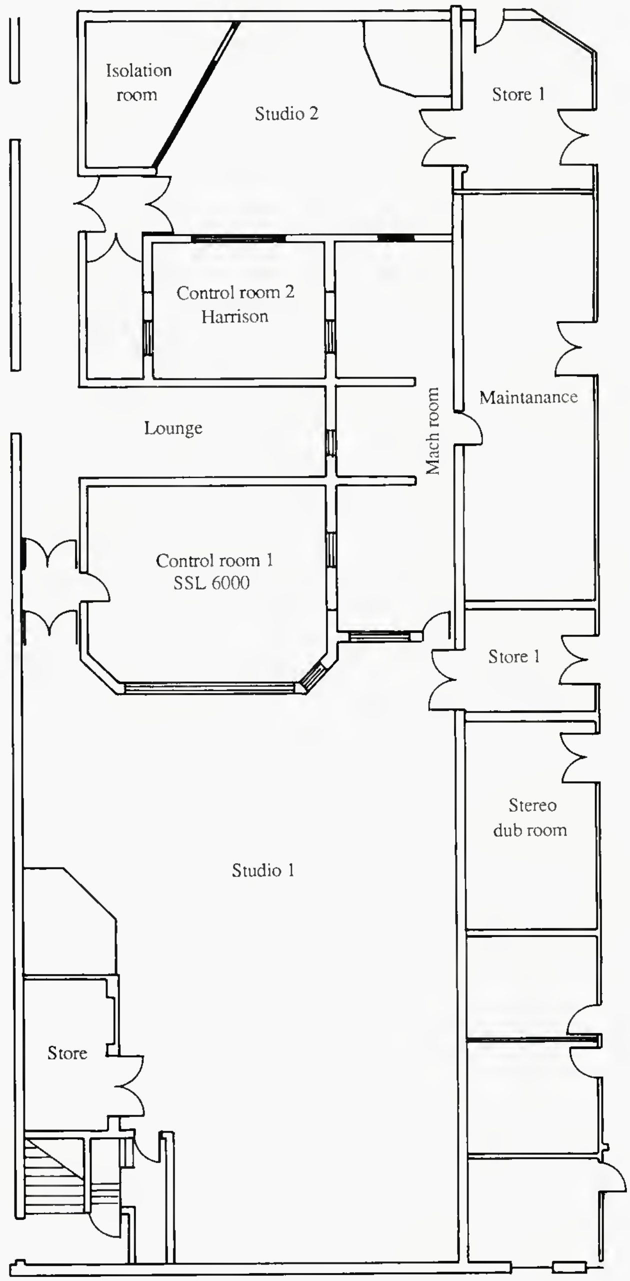


Figure 3. The studio floor plan.

opportunity for enterprising American record companies, producers and performers to go to the studios of AAV Australia for future recording projects.

RECORD DOWN UNDER TO SAVE MONEY

At the time of this report the Australian dollar was valued at approximately 64 cents to the US dollar. The significance of this, of course, is that an album project with a budget of US $100,000 could be recorded in Australia for around US $64, 000 plus air travel and accommodation, which could still amount to a considerable saving, considering the profit.

Studio One at AAV Australia boasts the very latest in recording technology including a Solid State Logic 66E series recording console and Sony PCM 3324 multitrack digital recorder. The studio is capable of operating as 24/48 track, in addition to the PCM 3324, it houses tape machines such as a Sony MCI JH24, 1/2-in., 1/4-in. ATR 2 track, Ampex MM1200, Sony PCM 1610 and a Sony PCM F1. The monitoring is designed by Sierra-Eastlake loaded with JBL components. Also available are Auratone, Yamaha NS10, AR18s and loads of outboard gear. The studio is 60 ft. x 43 ft. and the control room size is 24 ft. x 17 ft.

In addition to its fixed studio facilities, AAV Australia also operates a mobile recording unit which is considered to be the best of its type in this country. This unit has undertaken a large amount of work in the audio production area for both television and motion pictures and a great deal of its time is taken up with live recordings for the music industry. It was recently used in Sydney, Australia to record the music for a Bob Dylan film for HBO and is being used extensively for both digital multitrack recording and stereo broadcast on the recent Elton John "Tour de Force" Australian Tour.

Overseas artists who have recorded at the facility over the years include such outstanding talent as Stevie Wonder, David Bowie, Leo Sayer. Recent projects that have been completed at AAV Australia include albums by Australian Crawl, "I'm Talking" and John Farnham who has just finished a new solo album. Farnham of course is the former lead singer of The Little River Band.

Overseas producers who have used the studios recently include Fred Maher, Adam Kidron, David Kirshenbaum, Peter Henderson and David Courtney.

Other advantages of recording in Australia over the next few months is that they are enjoying the end of spring and approaching the delightful summer period. The studios of AAV Australia are surrounded by excellent shops, restaurants and hotels etc., and is in the center of most of the music and show business activity in Melbourne. High standard accommodation is available within easy walking distance of the studios.

The Audio department at AAV Australia has a tradition of breaking ground with new technology. From the mid-sixties, they were the first to use 4-track then 8-track, 16-track and 24-track and were strong supporters of Australia's first fully home-grown 16 & 24 track recorders. These machines were being designed and built by local genius Graham Thirkel through his company Optronics and after a series of custom designed mixers through the sixties and early seventies, AAV installed the first Harrison 4032 console then SSL Series 6000 currently working with the Sony PCM 3324 digital 24-track...again a first. ■

AAV Studio Equipment List

Studio 1: 24/48-track
Studio size: 18m x 13m.
Control room size: 7m x 5m.
Tape Machines: Sony MCI JH24, Sony PCM 3324 digital, 1/2-in., 1/4-in. ATR 2-track, Ampex MM 1200, Sony PCM 1610, Sony PCM F1.
Console: SSL 6000 E Series (40 in 40 out).
Monitoring: Sierra-Eastlake design with JBL components, Auratone, Yamaha NS10, AR 18s.
Outboard: AMS RMX16 reverb, AMS DMX15-80s delay, Lexicon Delta T102, Korg SDD 3000, 2 x Optro 3rd octave eq, 1176 Universal Audio Limiter, Marshall Time Modulator, 2 x dbx 160, 2 LA3A, 4 x Allison Research Gain Brains, 4 x Allision Research Kepex noise gates, Orban Parasound de-esser, EMT echo plate, Eventide Harmonizer H949, Lexicon Prime Time, 2 x dbx 160X, Tascam 122 cassette player, Technics turntable, 24-track Dolby.
Microphones: Large range available.
Instruments: Yamaha 9ft grand piano, Pearl drums, percussion kit.
Other features: Sontron Editron 500 synchronizer.
Studio 2: 24/48-track.
48-track by arrangement.
Studio size: 10m x 6m.
Control room size: 5m x 4m.
Tape Machines: Sony MCI JH24, Ampex MM 1200, Sony PCM 3324 digital, Sony PCM 1610, Sony PCM F1, 1/2-in. + 1/4-in. ATR 100 2-track.
Console: Harrison 3224 w/ Allison Research computer mixing.
Monitors: Sieera-Eastlake design with JBL components, Auratones, Yamaha NS 10, AR 18S.
Outboard: DD 1 delay, 2 x Optro 3rd octave eq, Marshall Time Modulator, Korg SDD 3000, 2 x dbx 160, 2 x LA3A, EMT echo plate, 4 x Allison Research Kepex noise gates, Orban Parasound de-esser, Tascam 122 cassette player, 2 x dbx 160X, Lexicon Prime Time, Aphex Aural Exciter, DeltaLab digital delay, additional outboard available by arrangement, 24-track Dolby pack, Optronics programmable eq.
Microphones: Large range available.
Instruments: Steinway 6 ft. grand piano, Sonor kit.
Other Features: Sontron Editron 500 synchronizer.
Mobile: 48-track
Tape machines: 2 x Optro 24-track, Sony MCI JH24, Sony PCM 3324 digital multitrack.
Console: Harrison 40/32 C Series.
Monitoring: Tannoy Little Reds, Auratones.
Outboard: Up to 12 outboard units can be run at any one time, including AMS, dbx, Yamaha and others on request, 24-track Dolby pack.
Stereo dbx 162, 2 x dbx 160, 2 x dbx 160X, spring reverb.
Microphones: Large range available.
Other features: Patch bay is bantam size, normally supplied with the Harrison C Series, including insert points for FX before and after eq. Using Sedco distribution amps it is possible to have up to 8 stereo and/or mono mix feeds. Mic lines fed to console via 27 and 16-way multicores. Communications link includes vision and T/B facilities. Powered by 3-phase power cable which can also be split to other units.

A Radio Studio Floats on a Noise-Reducing Floor

This article reached us originally as a P.R. release and has been expanded as a particularly appropriate product description for other radio stations and recording studios.

When banjo pickers and dulcimer players perform their traditional mountain music in the studios of Asheville's (North Carolina) public radio station WCQS-FM, they are insulated from the surrounding city noises by an invisible layer of "high-tech" nylon geomatrix material in the floor.

Enkasonic R noise-reduction matting was placed under the flooring in "sound critical" areas of the station's new facilities, located in a renovated building in this mountain resort's redeveloping downtown center. Ironically, Enkasonic is manufactured only a few miles from the radio station by Geomatrix Systems, a part of BASF Corporation Fibers Division in Enka, North Carolina.

Geomatrix materials, widely used for erosion and subsurface drainage control in highway and building construction, are three-dimensional, lightweight composites usually made of polymers and often combined with "geotextiles" (woven or non-woven fabrics) that provide soil filtration and stabilization in construction applications.

The striking characteristic of a geomatrix is the significant amount of open space within the matrix of fibers. The design allows the nylon matrix to absorb sound, channel water from underground structures in drainage applications, or anchor natural vegetative root systems for erosion and turf grass reinforcement.

The Enkasonic "floating floor," which separates the noise from the music at WCQS-FM, helped the station avoid more extensive renovations such as having to tear out the existing flooring according to station manager Tim Warner. Fortunately for the station (which suffers from budget squeezes as in most public radio facilitics), the best technical solution also proved to be the most cost-effective one.

"The only sound-reduction alternative I saw was to have the floor float on rubber isolating bushings," Warner said. A project that would have cost in the thousands of dollars. The bushings also would have raised the floor several inches, reducing overhead clearance at entrances and internal doorways. But as installed, the Enkasonic reduction treatment raised floors just over an inch.

Sound control was especially crucial because the building is located on a busy city street and there is a woodworking shop behind the building. Directly beneath the station is an unoccupied basement that engineers worried would act as an "echo chamber," reflecting and transmitting sounds through the floors into production and studio areas.

"We had problems with noise in the old quarters and I knew it would take a lot of work when we moved," said Warner. The floating floors in the new station were constructed in all of the sound control areas: studios and control rooms, master control room, mechanical equipment room where the air conditioner is located, the music library, lounge area and adjoining hallways.

"We used 1,122 square feet of Enkasonic," said Warner. "Our total floor area is about 1,850 square feet."

The station relied heavily on volunteer labor to install the floating floors and to renovate the building, which was provided by a supportive architect who has an office upstairs. Previously, WCQS had been located on the campus of the University of North Carolina at Asheville, where it began ten years ago as a student operated rock-n-roll station.

Now operated by a community group, Western North Carolina Public Radio, Inc., the station programs classical music, jazz, traditional mountain music, news and public affairs, and a fair amount of live performances.

Warner found the solution to his sound control problems almost by accident.

"One of the volunteer workers was familiar with Enkadrain, which he'd seen used on a local earth-sheltered home. I'd heard there was a similar product for sound control use."

We thank Jeff Renk at Price/McNabb in Ashville for his cooperation on this article.

Figure 1. The Enkasonic matting material.

Figure 2. A worker lays in the matting much as a rug would go.

Enkadrain is a sub-surface drainage control matting in the Geomatrix Systems product line, and has a non-woven filter fabric bonded to the black nylon geomatrix that is the same basic core material of Enkasonic. Warner said he called Geomatrix Systems to see if Enkasonic could solve his sound control problems.

"They described it to me and it made perfect sense, but when I first saw it, I couldn't believe it," said Warner. "It was the weirdest looking stuff I'd ever seen."

Enkasonic looks like a mass of coiled, black wire but is actually a matting of tough, nylon mono-filaments that are heat bonded together. It was specifically developed to create sound-rated floors in multi-story buildings and multi-family dwellings where testing by the Ceramic Tile Institute has shown Enkasonic to meet strict municipal building codes regulating noise pollution and sound transmission in such structures.

"Geomatrix Systems sent an applications engineer to show us how to put the Enkasonic in place, and we did the rest," said Warner.

The Enkasonic matting was placed directly on top of the old floor, which consisted of wood subfloor, wooden flooring, and a layer of Masonite intermittently covered with linoleum tile.

"It didn't even matter that the linoleum had huge chunks missing in places because the flexibility of the matting allowed it to fill the voids," said Warner.

Two layers of 3/8 inch plywood was placed over the Enkasonic at right angles to each other with their joints offset to reduce sound transmission. The sheets of plywood were glued and screwed together, resulting in a solid platform floor that rests on top of the Enkasonic matting without touching the adjacent walls. Compressed glass-fiber batting was used to fill in around the edge of the new floor to keep it from "bumping against the wall."

"The floor is constructed to keep any sound made within the room from going out of the room and coming back in," asserted Warner. "There is very little noise made when you walk on the floor, and it has just a slight give to it."

"We're pleased with the performance of the Enkasonic and I've recommended it to others. We've had no problems with it. It has done what the specifications said it would do."

"We're using a leftover piece of Enkasonic for a doormat," Warner said. "It makes a good one." ■

Television Stereo— On the Move At Last

Marshall King

This article takes a deep look into some of the problems surrounding the coming marriage of hi-fi stereo and network television.

I hear a lot of knowledgeable people say that, even though great strides have been made in the development of TV stereo, we can still expect to have the medium served up to us only in mono for a long while yet. However, if all the impressive technical strides being made by Orban, SSL, Kintek and others don't put solidly into stereo TV very soon, there is something else that will: consumer insistence.

What is causing this insistence? MTV to be sure, but probably more so is the growing awareness of home video. Almost everyone has found that you rent the finest stereophonic movies for home viewing for about a buck a pop. True, you must have a stereo VCR to play it on, but even that's no rarity anymore. If your friends don't already have one, chances are they will before long, for it's an observable American fact that no one wants to be left at the starting gate. The word is out: "Thanks to home video, stereo is both available and terrific."

You can see what this is doing to the networks, those mighty fortresses from whose R&D labs came the most binding technical parameters. Where once they were able to dole out their modicums of magic and pieces of progress on a slow timetable, understood only by their tax attorneys and various chairmen-of-the board, they now have to get moving or perish. The competition is not waiting.

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. One of such persons is Ed Greene, audio mixer par excellence of the Hollywood-based Greene-Crowe Sound Services, and another is Ron Estes, long time staffer at NBC and known for his innovative work on the "Johnny Carson Show." So much has been said and written about TV stereo development from the standpoint of transmission that it might be time to move back from the transmitter to the studio and listen to the views of those who record it in the first place.

Marshall King is with CBS-TV in Los Angeles. As a long time broadcast audio mixer he will, with this article, begin a broadcast audio column to appear regularly in these pages.

ED THINKS AHEAD!

I found Ed Greene at the Aquarius Theatre where he was about to record the weekly "Star Search" program hosted by Ed McMahon. As in most shows released by Greene, it is handled as though it will be released in stereo, whether that is the immediate plan or not.

"There is a good reason for this," he said. "Even if the show is aired the first time mono, there will almost always be re-runs, syndication, foreign release and so forth. They're going to want the show in stereo sooner or later. So I record it in stereo, on my own volition if need be. There's really no extra expense involved. Having to go back later on and reconstructing a stereo presentation out of bits and pieces that are filed away somewhere can be a full-blown disaster, not to mention unnecessarily time-consuming and expensive.

"If someone fails to think ahead in this matter, the product is jeopardized. As you know, when in the midst of a production with a whole group of people, not only in audio but in all phases of production. You interface with them all along the way, gathering information that, when put together, becomes the show. To retrieve that information six months or a year down the road, with all those people scattered to the winds, can make the stereo reconstruction a nightmare. You end up saying to yourself, 'Now wait a minute... where did I get this piece from? What roll of tape is it on? What file is it in?' To record in stereo at the outset will probably become standard procedure. At least it is with me if at all possible."

Ed Greene is talking here from recollections of near disasters where, when a show was originally recorded, no thought was given for later release in stereo. At the least, this means it should have been laid down in multitrack so that at a later time through pan pots, it can be divided to left and right channels. Otherwise, to produce stereo out of a simple composite mono track necessitates processing it via synthesizing, which he feels, does not at this stage of the game give the true effect. "The market is waiting for those programs that can provide good stereo," he said. "For those that can't, money is kissed goodbye."

"I'll give you an unbelievably good example of a tragedy in this area," Ed told me. "Each week for seven years I laid down on 24-track a show called "Don Kirschner's Rock Concert." He had every notable group of the 60's and 70's

Ed Greene (right) and his recordist Larry Greene prepare to record the Star Search program on 24-track audio tape for both mono and stereo release. Photo by Marshall King.

on that program... Manilow, Benitar, you name it. The standing order was that, once the show aired we'd erase the 24 tracks in order to conserve tape stock. You can see what happened. Any further use he might have had for re-packaging this dynamic collection of music... "The best of Rock," "Rock Concert Goes Stereo, was lost."

STEREO RE-DEFINED

Methods for recording stereo demand a clear-cut definition of what stereo is, and even this may be cause for *dissension in the ranks. All anyone can say for sure is that, since we have two ears, we perceive sound in the sense that it has direction, that it comes from three-dimensional sources with respect to our own personal location in space. Does this mean that anything we do to reproduce this perception can be considered to be stereo?

Many say the answer is "yes" and that to discuss which method is "proper" is irrelevant. If so, we have at least four ways to create stereo sound: two ways at the time of original recording and two ways after the event has happened. For the former we can use a conventional placement where the mics are more widely spaced left and right amid the sound source and fed either to the left or right output channels.

For the two post-recording methods, we can direct the mono tracks of a multitrack machine to either left or right channels via pan pots, or we can (at the time of broadcast transmission) process our mono tracks into synthesizing amplifiers referred to earlier, where the effect is usually achieved through band splitting and phase displacement. The test as to whether any of these systems is viable has so far been two-fold: First, does it give a "feeling" of special sound that does not contradict the picture. Second, does it leave the mono transmission unharmed?

MIC PLACEMENT AND ASSIGNMENT

Which method of recording stereo is most often used depends on circumstances. For a show broadcast "live" (or live-to-tape, as Johnny Carson) the mics are invariably assigned to either left or right channels, even though they may be going discreetly to multitrack at the same time for updated processing (another instance of planning ahead for later rebroadcast). We'll take an inside look at Ron Este's handling of the Carson show next issue. In the meantime, Ed Greene notes that the method he uses depends somewhat on the program material.

"One of the things I've liked to do during these years when the market has been so small, is to experiment before the hard-and-fast rules come in. Most of us have learned that the dialogue should be kept in the center track. We've seen movies where it was panned across the screen as the actors moved, but in television it just doesn't work for them. It's too hard to get the voices properly assigned; the camera cuts alone make it unworkable in television.

"Otherwise, I try to make some really nice, wide stereo if I can. You have to be careful, remembering that people are sitting in front of a 19 to 25-inch screen with speakers anywhere from six inches to a foot or more on either side. So whatever you may do to enhance the sound in stereo, it can't be so disconcerting that it will not go with the picture.

"I've concluded that dialogue should pretty much be left in mono, as the movie people are doing. The only deviation from this is a thing I've played around with in the last few months: a stereo fish pole or stereo boom. It's a European technique called M/S, or sometimes X/Y, where close-spaced microphones are used. I've been using a pair of Schoeps for this because they're physically small and allow me to mount two mics on one fish pole. They call this "coincidence stereo," originally used for orchestras. First, we mount a cardioid microphone facing forward, then a second microphone at ninety-degrees to that which is a bi-directional facing left and right. These are called the mid and the side mics. We matrix these two together electrically, taking the mid and the plus of the bi-directional mic which becomes the right side. This particular single-point pickup is an extremely effective stereo source. It works just fine, so I've started to use it on some dialogue. Also, we did an Amy Grant special for NBC where there was a song and a couple of other pieces where it worked beautifully."

This technique may require some caution by the mixer who sees it as a panacea for all his stereo problems. Two of the mainstays of television, the sitcom and the soap opera, may not lend themselves to a pair of coincidence mics on each boom. The abrupt changes in actor positions on stage, combined with the fact that the mixer is constantly cutting from one boom to the other, would surely result in a nightmare of a perspective for the viewer.

"You really have to go with the standard two-boom operation for this," Ed points out, "or RF mics where needed. If I want to get a feeling of the room, sometimes I'll put up a pair of PZM's either in the room, or on the walls, or even above it to get a feeling of the area. I'll mix that as stereo ambience with the center monophonic dialogue mics, which is pretty effective.

"One of the little problems in stereo sitcoms that you really don't find in mono is that, if you're not careful, you'll sometimes hear three audiences. You'll hear the stereo audience as picked up by the live mics that the mixer put out, you'll hear the live audience as picked up on the boom mics from spillover, and then sometimes you'll hear the stereo sweetening which is done by an operator playing audience reaction cartridges to fill in the holes. This means that you're hearing an audience with three different sound qualities and it's very disconcerting. This is just one of the new problems that stereo has introduced us to, and we're learning to avoid it."

If "sweetening" the audience reaction has been a question mark in mono broadcasting there is even more vigilance in stereo. Not only does the operator have to have good value judgment (match the laugh to the joke!), but the nature of his tape library has to be such that it blends with the live audience the mixer is dealing with. So difficult is the task that only a handful of operators are regularly called upon. And, as in most areas of the business, if you find someone you feel comfortable with, you invariably use him for everything to make your own life easier. Ed Greene is no exception.

"We normally use the work of Carroll Pratt," Ed told me. "He and his people all have stereo cartridge machines to sweeten with. During taping I try to go with the live audience as much as possible, which we pick up in stereo. But later during editing, a show is often chopped up and we can't use our live audience strictly as recorded. Or, let's face it, sometimes the laughs simply need to be augmented. For instance, we just finished doing the Kennedy Center Show in stereo for CBS. Carroll Pratt took over our live audience, cut it together and puts it into his cartridge machine so that he has the same audience that we had. He just does it by feel and it matches perfectly. Those guys are so good and they've been doing it for so long that they become part of the show itself."

Having come to television from the record industry, Ed has been on the leading edge of audio for a long time. He seems to enjoy dealing with the frustrations as well as the triumphs.

AN ENGINEER WHO MAKES AN ARTISTIC STATEMENT

"I feel fortunate to be part of it" he told me. "As you know, audio is often a labor of love. Sometimes too often, that is one of the disadvantages. Yet, that can be an advantage, too, because the producers are so preoccupied with so many other areas that they don't even want to know about audio. Unless, of course, there's a problem. And even then they don't want to know about it, they just want someone to take care of it. And as for stereo, they don't even begin to know what to tell you to do; it's something we have to bring to them! So, what's the bottom line? They leave us alone to do our thing, and I appreciate that. I try to return that respect, that privilege, by giving them the best product possible."

One of the frustrations for anyone who contributes heavily to a show is finding that what is aired is far removed from the original work. Greene is quick to admit that he is sometimes critical of these results, especially if they appear to be unnecessary. "I know that AM radio is still having its growing pains with regard to standards for stereo broadcasting, but for television the formats have pretty well been worked out. What I often find annoying is the way those standards are abused.

"Unfortunately, many of the stations, including NBC which is now a full-network stereo operation, will synthesize a program in stereo if it has not been originally recorded in true stereo. Worse than that, the stereo synthesizers think they can delineate between a mono and a stereo signal. I've heard them switch right in the middle of one or the other. That is, they'll cut away from stereo program to do a monophonic commercials, and about ten seconds into the commercial – Bam! – it switches to the synthesizer. It's the most disconcerting thing you can hear."

STEREO SYNTHESIZERS: HELP OR HINDRANCE

No one can blame a manufacturer or an innovator for wanting to be the first one out of the gate, but there should be no awards for prematurity. Until agreed upon methods for implementing a new idea becomes widespread, the user often has to put up with many annoyances. The introduction of synthesized stereo may be a fine example. It isn't that the designers don't know how to make it flawless in theory, it's that in order for it to work with a variety of existing distribution systems (such as the several kinds of transmitters and the many types of detectors in the audio chain that claim to distinguish mono from stereo) they are sometimes obligated to incorporate compromises not of their own choice. Even then, what will work with the studio A equipment may not work with studio B, for they, too, are cut from different patterns. So in the first stages of development we are often left with a variety of definitions, methods and results.

Perhaps it is too easy to be critical before all avenues have been explored. The designers are still pulling it all together, and two of their aims are to come up with sharper recognition circuits and to make their systems operator proof. With regard to the former, this touches on the problem just mentioned, whereby a program may contain both stereo segments (the main body of the show) as well as mono segments (commercial inserts or archival film clips). For the broadcast station equipped with circuitry to create synthesized stereo, we want these circuits to "recognize" whether or not it is dealing with a signal that has been recorded for stereo or mono. If it's stereo, we want it to pass through untouched, for the effects of synthesizing a signal that is already stereo can be very disturbing to the ear. This happens when a "recognition circuit" interprets the center-channel dialogue of a stereo movie as being mono simply because the left and right channels are, for the moment, identical.

Conversely, a listening disturbance occurs when and if the overall program pops back and forth from stereo to mono as when, as Ed Greene noted, the circuits react sluggishly or at improper times. One reason for this and other undesirable results is not so much flaws in design as it is a bad judgment call by the operator on duty. Designers reason that operators should be given certain latitude in circuit use, and corresponding controls are provided. Just as a noise gate threshold is set at the convenience of the operator, so might a stereo synthesizer have flexible parameters. But experience has shown that, in the absence of total error-free automation, an old rule might well apply to stereo synthesizers: less is more. If it reacts too early or too late, too often or not enough, the effect is less than adequate. Another problem is if the mixer has set the left and right channels for an abnormally wide separation, the device itself becomes an animation and is a distraction. Good taste and subtlety may be the key words for everyone involved. With mixers, their discretion prevails. Yet at the transmitters many station engineers reportedly have standing orders for operating personnel to keep hands off all adjustments.

The prevention of false triggering between mono and stereo is being given attention by such leaders in the field as Orban. This would entail some cooperation by the producers (editors) of the program, whereby each segment would have inserted into the vertical interval preceding it an industry-standardized burst that would identify that segment as being either mono or stereo. The recognition circuits of the transmitting station would treat the program in accordance with the last burst received, so that the switching from one to the other would be reliable and automatic.

The key phrase in the above approach is "industry standardized." An often heard complaint against the world of commerce is that competing companies in the same industry sometimes fail, or refuse, to agree on common formats that would benefit all. Yet, there may be hope here, for recent examples abound whereby even giants can't always go it alone. No one knows how much money was lost (or not earned) because Beta won't talk to VHS or because CP/M is hostile to MS/DOS. It is hoped that these lessons have not fallen on deaf ears and that standard methods for delivering workable synthesized stereo are on the horizon.

IN SUMMARY

It is obvious that Ed Greene has given a lot of thought and time to the direction of good audio for television, for his closing remarks again reveal his dedication to the medium.

"There's more to the audio business than meets the eye," he said. "There was a study done on people watching TV and what effect audio had on their response. Generally, when people say they liked the show they rarely mentioned the sound. They'll say, 'Gee, what a good looking show!' What came out of the study was that, when a group looked at a show that had mediocre sound, their response was just 'okay.' If they looked at the same show and the sound was really fine mono or stereo, they almost always said, 'What a much better-looking show.'

"The other part of the test was when the sound was left alone but the picture was tarnished. The viewers kept looking at the show although the picture went from color to black-and-white, it developed horizontal lines and other kinds of interference, but they still watched the show. However, when the sound was cut off, even during the time the picture was perfect, they simply got up and left. So the point we have to make with producers is: good sound is an important part of the picture.

"I happen to have a VCR at home and I rent movies. The good stereo that is available for our living rooms is absolutely astounding. That is our competition. If we don't take care of business, we're going to find ourselves with a diminishing audience. I'm not saying that sound is all of it, but I saying that people are becoming aware of its possibilities in video, so we must be more diligent than ever. For years audio people have suffered from the paranoia felt by a shunned stepchild, but now that we're being courted we have to be ready." ■

Dolby SR: Practices and Principles

Ralph Hodges

Is this new Dolby system the best of the old and the new?

It's gratifying and ego-building to report that my prediction in these pages of some years ago has been fulfilled: Dolby Labs has at last created professional noise reduction based on the sliding band principles previously employed only in the consumer B and C systems.

Dolby engineers have lusted after applying this technology to professional for many moons, but have been balked by considerations both practical and economic. However, with digital making almost anything analog look relatively cheap, and with huge investments already committed to analog now in jeopardy, the time has evidently become ripe for another Dolby leap into an uncertain future. It has turned out to be a cleverly calculated leap, involving an integration of sliding-band processing plus several other things that were lying around the workbench, and a closely reasoned re-examination of signal-processing possibilities.

Dolby Spectral Recording (SR) is in fact an analog noise-reduction system, although the company would prefer you to think of it as something more: A "signal-fitting" system that anticipates the capacity of a tape-recording channel, filling it entirely. The concept is easily grasped through an inspection of the MOL curves of any studio mastering tape. There is room for a lot of good stuff in the mid-range, somewhat less room at the bottom end (and a noise problem that can get you if you don't watch out), and depending on the bias point chosen, a limited to inadequate capacity near the very top. Fortunately, these MOL curves track the spectral energy distribution of most music fairly well. But if they tracked it perfectly, we might be able to listen to the top 40 without gagging or wishing for an instant onset of deafness. The new Dolby system attempts a closer approximation to that ideal tracking than we've seen before, endeavoring to cram the tape with everything it can take at all times, and then undoing the havoc raised with the signal spectrum via the sort of complementary decode process familiar to all who have studied compander-type noise reduction. The immediate difference is that, besides noise reduction, the SR concept attempts to maximize recording headroom in a new way not to be confused with Dolby HX, which is a bias-shifting technique implemented very differently, and involving no decoding process. Hence, Dolby SR seeks to make an analog tape recorder do everything (except not suffer from transport vagaries) that anyone has ever wanted it to do.

Ralph Hodges is a free-lance technical author.

OF FIXED BANDS, SLIDING BANDS, SIDE CHAINS AND MORE.

The Editor having warned that unfathomable complexity will not be welcomed in these pages, I must discuss SR in a rather selective way, glossing over most of the subtleties that make it, apparently, a worthwhile and musical thing to use. So we begin with a catalog of the gross mechanisms of Dolby processors through the ages(they all figure into the new design), and take what account we can of the refinements in a wrap-up.

Speaking for the moment of the complete (encode and decode) action of Dolby noise reducers, we note that A-type operates independently in four fixed frequency bands spanning the full audio spectrum, and assumes that when any given band has to open up to pass a signal of interest, the proximity (in frequency) of the signal to all others in the band will mask any noise present. The other bands remain shut down–at least to the extent of roughly 10 dB of attenuation–and curb noise that might be otherwise unconcealed by signal. B-type has a sliding, frequency/level control band that shifts its turnover point to pass what seems to be desired information, and attenuates (by the same 10 dB or so) anything above the frequency of that information.

This seems as though Dolby processors somehow "listen" to music and act accordingly, but of course, they don't. They merely boost, at the tape-recorder input, frequencies that are under utilized in terms of recordable level, and cut them down to their original level at the output. The tape recorder noise gets the cut, but not the preliminary boost, and hence it tends to disappear.

So much for the basic principles; now for the headaches of the implementation. Jacking around the levels of a complex music signal in this way is not to be undertaken

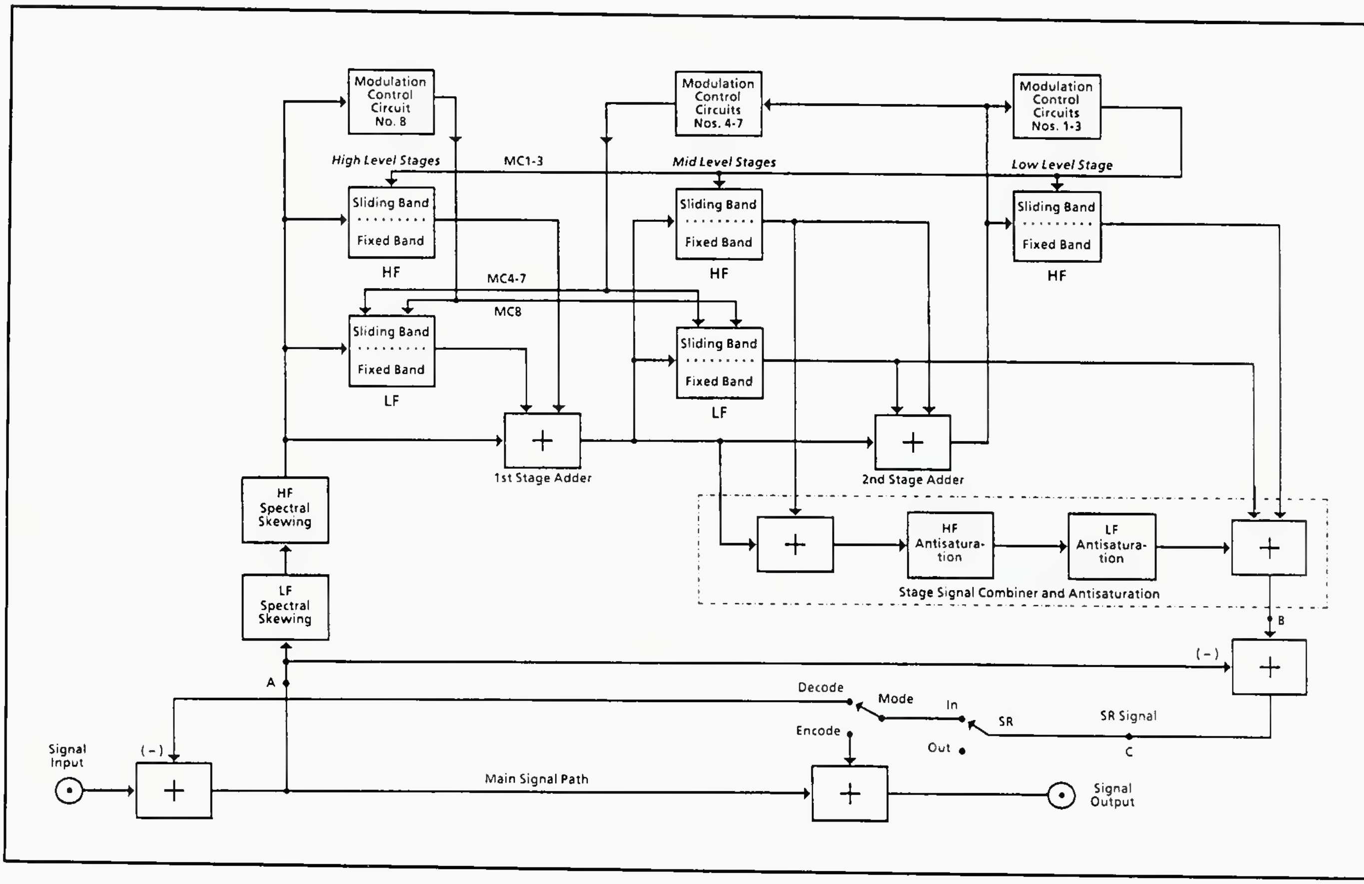


Figure 1. A basic block diagram of the SR processor. The more mysterious blocks are explained in the text.

lightly. Dolby generally prefers to avoid attempting it in the main signal path, and instead relies on "side-chain" processing. The additional gain required for low-level signals prior to recording comes from a circuit loop(s) encompassing the main signal path with an output that combines additively with the main signal during encoding and subtractively during decoding. The advantage is that high-level signals not needing any boost/cut processing, the side chain goes out to lunch and behaves as if it doesn't exist.

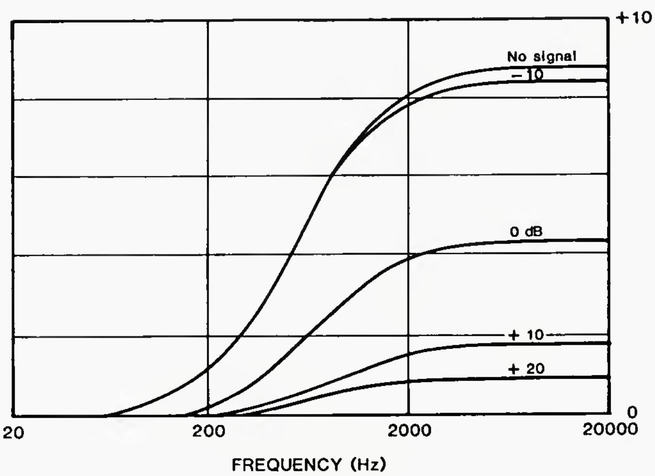
However, as with all compander systems, there is the matter of turning off the encoding boost when a sudden big blast comes along quickly enough to prevent gross overload of the recording medium, and so we find in previous Dolby systems a hard-limiting or "clipping" function. Clipping is just what you don't want happening to your signal, but the theory states that the decode function will restore the integrity of the signal upon playback.

OF 20 dB AND MORE

When Dolby embarked on the design of processors to afford more than a nominal 10 dB of noise-reduction effect–20 dB plus in the case of SR–some considerable elaboration was needed. Mainly, frequency-response errors in the recording system and infrasonic and ultrasonic noise components had to be kept out of the compander works. While their mistracking effects on level manipulations of 10 dB might be tolerable, they would not be at 20 and above. Hence, the consumer C-type system inaugurated the concepts of spectral skewing and anti-saturation, the former being fixed equalization, involving a cut during recording and a boost during playback, instead of the other way around–and the latter refining and reinforcing the effect dynamically. In C-type, the action takes place at or near the extreme top end of the audio range, but in SR it goes on at both the top and bottom.

C-type also introduced "staggered-action" sliding-band stages, leading many to the faulty conclusion that it was merely two B-type processors stacked one on top of the other. SR goes to three sliding-band stages for high frequencies and two for low. Ray Dolby's reasoning is that "linearity" is better preserved if several processors each do a little than if one is asked to do a lot. With multiple processors, each can be designed to make its contribution to the combined effect a bit differently than the others.

Furthermore, each sliding-band processor is accompanied by a fixed-band processor, linked up to work with it in a fashion Dolby called "action substitution." Why provide both types of action? Simply, the sliding-band arrangement is highly frequency selective, and tends to magnify any response errors present in the recording channel. A Fixed-band treats all frequencies within its band the same. Ideally,



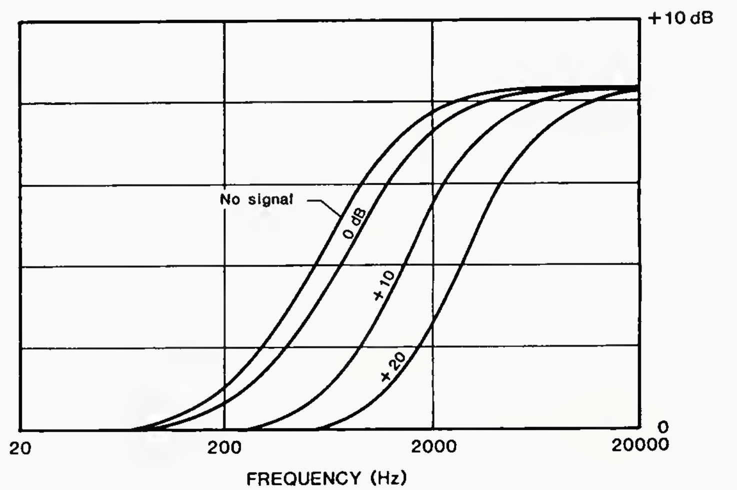
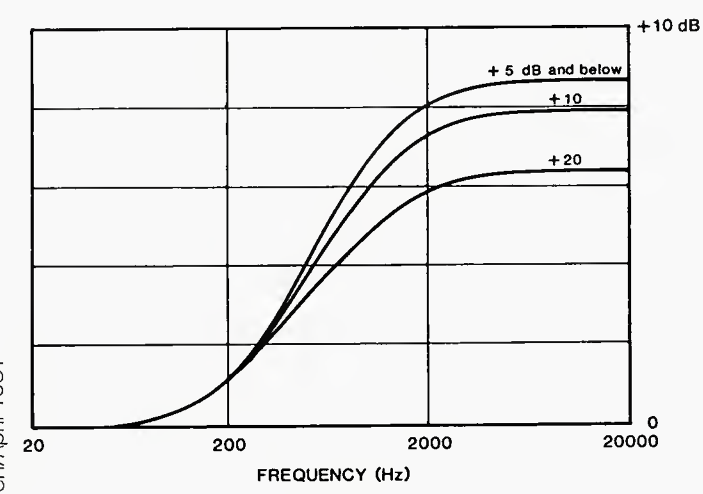
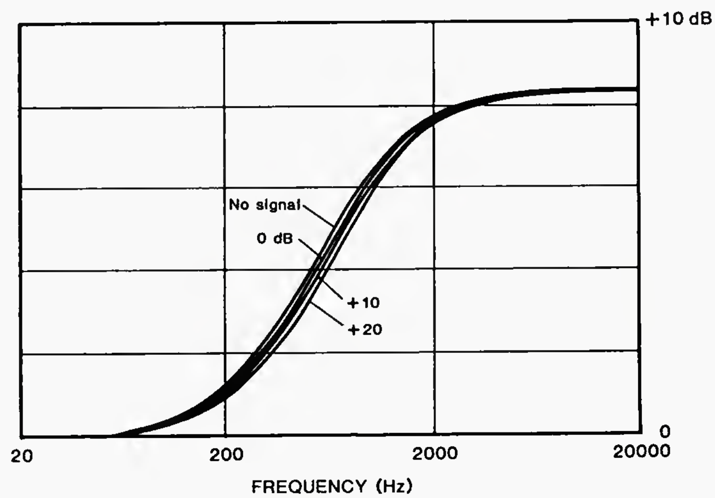


Figure 2. The desirable effect of modulation control on a fixed band compressor. Note that the knees of all curves (below) now coincide, and the compression action remains strong over a wide range of signal levels.

Figure 3. Modulation control for a sliding-band processor. Above:No modulation control;Below:With modulation. This provides maximum compression.





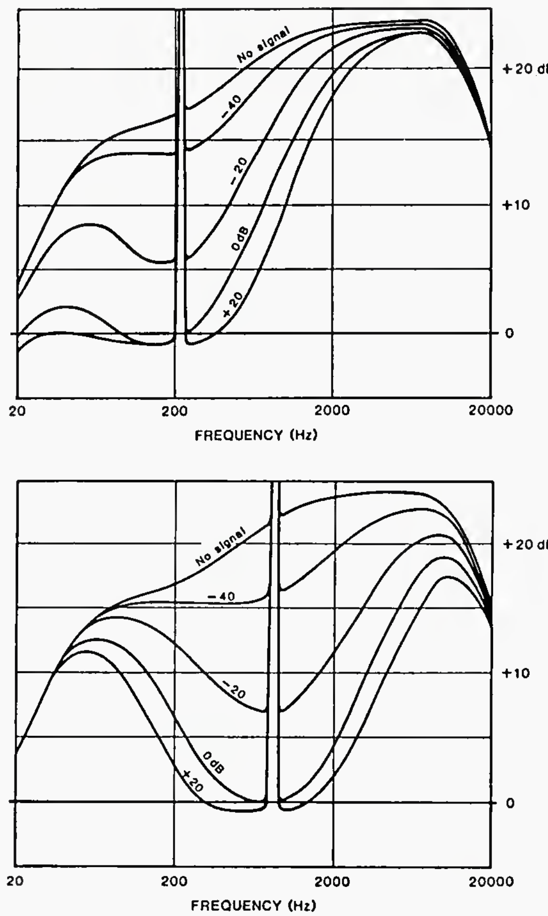


Figure 4. The total effect of the combined actions. (above-200 Hz; below:800 Hz.) Signal frequencies are passed with no impairment and with high resolution. Compressor action is strong above and below, except at the frequency extremes,which are deliberately attenuated to avoid overload of the recording medium.

you'd like a sliding-band effect operative over any frequency range that contains no desired signal (but which is likely to acquire noise via the recording process) and a fixed-band action when there is in-band, low-level information that must remain intact. Dolby SR is said to be able to substitute one type of action for the other whenever signal conditions warrant it.

SR is furnished with "modulation control," a scheme complex in implementation by which the compander is prevented from overreacting to signal conditions that might scare it into reducing compression for fear of blundering into overload. Modulation control, as applied in the SR context, appears delightfully sexy, but even a superficial explanation of its action would overload this paragraph.

Finally, when overload conditions would become inevitable (as they would occasionally with compression action involving such high gain, unless something were done about them), a faster response time called "overshoot suppression" steps in. This is also complex in its organization, with varying thresholds controlled by averaged-over-time signal conditions, frequency dependent time constants, and two-stage action. Overshoot suppression is also divorced from the main signal path, influencing only the control signals that regulate the side chains.

NEW CALIBRATION

The Dolby argument that SR is relatively tolerant of in-band response errors and out-of-band signal intrusions seems well-founded. The Dolby argument that calibration should be as precise as humanly possible is equally apropos. Therefore, in place of the familiar Dolby tone, SR processors internally generate a pink-noise signal 15 dB below reference level and interrupted periodically by "nicks" to serve as identification.

The first four seconds of pink noise is recorded on tape. The second four seconds bypass the tape and are fed directly to the monitor amplifier. The four second segments alternate for as long as you wish, so that the recording system can be trimmed by ear or instrument for some measure of resemblance between input and output. As always, it is strongly recommended that at least one segment of the noise signal remain recorded at the head of the tape as a reference for subsequent transfers.

LO, IT APPARENTLY WORKETH

It is never a pleasure to see an already complicated system become more complicated still. Nor is it gratifying to pull already expensive circuit cards out of a rack in anticipation of replacing them with even more expensive cards. But we know how it goes, and the pay back is that SR can sound worth it: clean, astonishingly dynamic, and tailor-made for multiple transfers with a minimum of intermediate processing.

Dolby believes that the fewest audible modulation effects when an encoded signal is listened to takes you most of the way to a successful companding system, and by this criterion the SR concept scores high. Not that an encoded signal isn't recognizable as such; quite the contrary, it might just prevent the occasional past embarrassments when undecoded A-type material went straight to the cutter head for mastering of the LP. But it has a coherent character suggesting that nothing more serious has gone wrong than an inept hand on a console compressor. Perhaps the worst to be said about SR is that it will leave tape-guidance problems and microphone preamp noise dreadfully exposed. The best is that it will give those old multitrackers, if properly maintained, a new lease on life.

GRATEFUL DEAD RECORD WITH DOLBY SR

One of the first–if not the first–major multi-track pop albums recorded and mixed with Dolby SR will be a new Grateful Dead release on Arista. Instrumental tracks were recorded by the Le Mobile remote truck during unique "live/studio" sessions at the Marin Civic Auditorium north of San Francisco, with overdubs and mixing done at the Grateful Dead's own studio nearby in San Rafael.

The Dead have always had trouble capturing the spontaneous ensemble improvisations of their live performances in the structured atmosphere of the recording studio. On the other hand, live recording in front of an audience imposes obvious technical limitations. Looking for a middle ground, the band decided to book the auditorium for several weeks so they could work together on stage while still taking the time and effort needed to make the best recording possible.

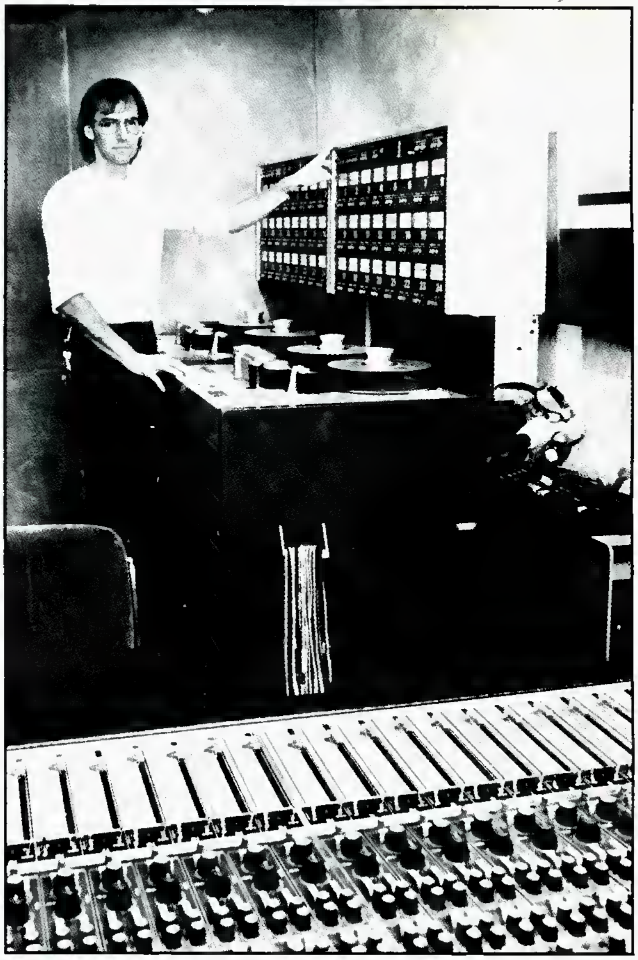
The band had previously worked with Le Mobile owner/engineer Guy Charbonneau on a video project, and when called for the album recordings he immediately suggested using the new Dolby SR system.

"I had done some recordings with SR several months before the Grateful Dead called," recalls Charbonneau. "I did a video project with Journey and some extensive tests with the Pat Benatar band. I was very pleased with what I heard–and what I didn't hear! Basically, what I heard off the tape sounded like what was coming off my Neve console."

Charbonneau communicated his enthusiasm to the Grateful Dead's recording engineer, John Cutler, who in turn contacted Kevin Dauphinee at Dolby Laboratories in San Francisco.

"John said he wanted to run down a few of this own tests before committing to the project," says Dauphinee. "I loaned them four channels of SR so they could do simultaneous encode/decode test, comparing what they heard on tape to what was coming off the board. I understand they did some vocals, Mickey Hart did some drums, and Jerry Garcia did some guitar. They were very pleased with the performance and contracted with Le Mobile to have SR installed for the project."

It should be noted that the Dead have always been



Owner/engineer Guy Charbonneau inside Le Mobile.

notoriously demanding when it comes to audio quality. For example, most recordings of the band done at their studio have been on the 2-inch 16-track format, using two interlocked Studer A80 decks. The slight but audible improvements in noise and crosstalk performance were important to Cutler and the band. Therefore, on the previous video shoot, Charbonneau had outfitted Le Mobile's dual Studer A800 machines with 16-track headblocks. But that sometimes led to problems because 16 tracks were insufficient and, with both machines running locked together, the tapes could not be overlapped; that raised the possibility of tape running out during an inspired instrumental jam.

Dolby SR to the rescue! "It was just what we needed for this project," says Charbonneau. "With SR we had performance better than with the 16-track blocks and we could overlap the machines. Also, we found we could run at 15 in/sec instead of 30 in/sec, which meant we had longer running time plus better bass response."

Charbonneau says he is still experimenting to determine the relative merits of 15 and 30 in/sec recording with SR. "In a normal studio situation, I might want to use 30 in/sec if we were doing a lot of bouncing, overdubs, slave tapes, and editing," he says, "but with this project it was laid down pretty much like a live recording. I expect there are some tradeoffs involved between the two speeds, but as far as I can tell noise doesn't seem to be one of them."

After the basic instrumental tracks were completed, Cutler contracted with Dolby for a total of 34 channels of SR to complete the project. Mixdown would require 24 plus 2 for the stereo master, and 8 additional channels were provided so band members Mickey Hart and Bob Weir could do overdubs in their home studios.

Although the Grateful Dead album was the first major Dolby SR project for Le Mobile, Charbonneau expects many more to come. "I feel I have the best recording truck on the continent," he says, "and it does not come cheap. I have held off on digital because I did not want to pass on that extra cost to my clients. With Dolby SR I can give them that same level of performance without the extra charges. A lot of people who were groaning over digital costs are now breathing a sigh of relief." ■

Automating Studio Management

Arthur Stoppe

Computers Come out of the Control Room and Into the Office

Contrary to what most people outside of the business would believe, much of the day-to-day business of running a recording studio is boring. Not only are there the usual things that are a part of running any business–scheduling, billing, bookkeeping and inventory–but there are also many tasks unique to the studio. Take, for instance, giving a price quote to a customer prior to his booking studio time. Say you have a client who has to do the audio post-production for a half-hour TV show. You get out your calculator, make an estimate of how much time the client needs, multiply that by the hourly rate, add in supplies, tax, etc., and scribble all this down on a scrap of paper. If you're lucky, you won't end up losing this paper just in case the client calls back and asks for the figures again. Or take the case of the record producer who, in the middle of recording an album project, needs a figure on just how much of his budget he has spent so far. You gather together all of his invoices, get out the calculator again and start working. Certainly, this is not the glamorous life of recording studio management that you once dreamed about.

Wouldn't it be great if you could get a computer to do these and many other tasks necessary to the management of an audio recording studio? Computers in some form or other have been at work in recording studios for some time now. They have been helping us mix, control our consoles and lock our tape recorders together. Isn't it time for them to become involved in the automation of day-to-day studio management? There are many tasks necessary for the operation of an audio studio that could be handled more efficiently through the use of computers. Scheduling, inventory, billing and bookkeeping, all of these could benefit from automation. Well, that's what we thought at Sigma Sound Studios, and we decided to do something about it.

If you are a typical business, you can computerize much of your work with available, off-the-shelf programs. Indeed, much of the work of running a studio could be handled by existing software or modified versions. You could assemble commercially available database, word processing and accounting programs and come up with ways of using them. This approach, however, could prove to be at best, a compromise. There are many specialized tasks unique to studios that could not be handled efficiently and adequately by such an assemblage of software. Besides, who would want to deal with the complex task of getting several different programs from different software publishers to exchange data and work together as an integrated studio management system? This job alone could be as involved as developing a suitable program from scratch.

With this in mind, Sigma Sound set out to develop its own Automated Studio Management System. Of course, since none of us are accomplished computer programmers, we would employ a professional to do that part of the job. However, we still had to be involved in defining the overall "architecture" of the system, since we knew best what we needed to do, and how we would prefer to do it. We also had to be closely involved in the testing and debugging of the program.

HARDWARE CHOICES

If you are going to develop your own software, you are of course, free to choose what kind of hardware you are going to run it on. If you are hoping to perform a lot of different, complex tasks simultaneously, your first impulse might be to base your studio management system on a centralized minicomputer connected to a number of satellite terminals, instead of using desk top personal computers. we considered this course of action for a while, but decided to go to the personal computer route for several reasons. First of all, while minicomputers are more than capable of doing the job, they are also fairly expensive. At the same time, personal computers are becoming less expensive while making substantial advances in computing power and speed, as well as in the area of multi-user utilization (via the necessary hardware and software add-ons). The multi-user question was of importance because what we wanted to do could be accomplished from a single, stand-alone desk-top computer, while we recognized that there would be advantages in being able to perform several different tasks at once in different locations. For instance, the Traffic Manager would be able to schedule studio time another employee was generating

Arthur Stoppe is the chief engineer of Sigma Sound of Philadelphia.

invoices.

What kind of personal computer should one choose? We decided on the IBM PC and compatible family of computers, primarily because of their wide acceptance throughout the business computing world. The IBM PC/XT and its clones have the hard disk storage necessary for our projected system's large data files, and the newer PC/AT has the speed and other features necessary for multi-user implementation. The newest of the PC class of machines, based upon the Intel 80386 microprocessor, offer even greater computing power and more extensive capabilities in the multi-user area.

OUTLINING THE PROGRAM

Defining the architecture of a computer program involves spelling out in painstaking detail every step of the task that you wish to automate, so that a programmer can write the appropriate software. This process includes articulating things that you do instinctively, or normally take for granted. For such a multifaceted task as that envisioned for our studio management system, this proved to be very complex undertaking.

The first thing that we did was to come up with the following "wish list" of things that we wanted the program to do:

* Maintain a customer data file of addresses, phone numbers, contact people, etc.

* Permit flexible pricing for each client and recording project, for cost quotations as well as for actual billing.

* Keep track of the day-to-day studio schedule, session dates and times, setups, equipment requirements, and the like.

* Generate client invoices based upon the studio services and supplies used on sessions, plus the previously determined pricing data.

* Maintain customer account data, based upon invoicing and subsequent client payments, and interface this data with a complete studio accounting package (Accounts Payable, Accounts Receivable, General Ledger). Also to be included as part of bookkeeping would be employee payroll.

* Keep track of supplies inventory.

* Maintain a tape library.

* Generate reports based upon the various types of data in the system, such as "Which clients have done the highest dollar amount of business with us this year?" "How much 1/4-in. and 2-in. tape do we have in stock?"

An interesting problem in defining just what the program should be was the fact that it had to be used by Sigma Sound in our New York and Philadelphia studios. While both

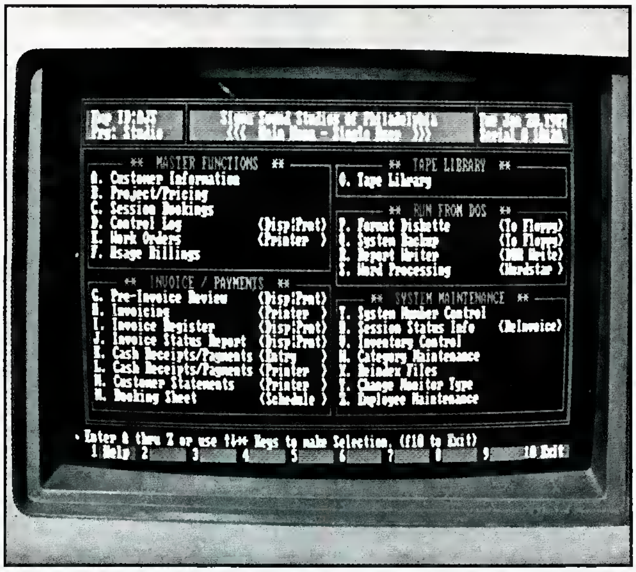


Figure 1. The main menu.

locations are part of the same company, and generally have similar operating procedures, there are some differences between NY and Philadelphia in the way in which certain things are done. For Example, New York books more time for clients in "blocks" of days or weeks at a time, while in Philadelphia most studio time is scheduled on a session-by-session basis. Each of these two ways of doing business calls for different features to be incorporated into the software. The resulting program has to accommodate both NY and Philadelphia.

TESTING AND DEBUGGING

Once the actual writing of the program was underway, we had to be closely involved in the testing and debugging of the software. There are two things that need to be done during this process. First, actual logical flaws or "bugs" in the program need to be found so that they can be fixed. Second, to find out if the program, as it is written, really addresses our needs and meets the goals that we set for it. We also needed to answer such questions as "Is the program easy to

Figure 2. The session booking screen.

```
Emp ID:AJS      Sigma Sound Studios of Philadelphia        Tue Jan 13,1987
Prg: Session          <<<  Session Review  >>>             Serial # 1012A

Project   2505  PSYCHIC CHEERLEADERS for OBSCURITY RECORDS
Studio    01    Date  01/13/87  Start    07:00-P To    03:00-A Must  04:00-A
Producer  P. R. O'ducer         Engineer  AJS    Assist SEM    Other

Type => R  Source    Mix 1     Mix 2               Titles
Size         /         /         /       How's the Weather Up There
Tracks       2                           Girls Just Want to Have Lunch
Speed       30
Digital
Noise
NWB/M       370

   New Prj   O/T   Vid   TC   New 2*

Confirm by AJS    on 01/13/87       P.O. #  67789         Workorder  <Fkey>
                                                           Number     Edit
Release by AJS    on 01/13/87  NO   Deposit  2500.00       86-1486    Mode

Use fkeys below to enter Selection.
 1 Help  2 Prev  3 Next  4Confrm 5Cancel 6Equip  7 Edit  8Setup  9 Find 10 Main
```

```
Emp ID:AJS      Sigma Sound Studios of Philadelphia        Tue Jan 13,1987
Prg: Cust          <<<  Customer Information  >>>          Serial # 1012A

Company          OBSCURITY RECORDS                     Cust. No.    OBSCUR
Contact          P. R. O'DUCER
Title            PRESIDENT
Address          12999 SOMEBACK ROAD                   Sales Tax     6.000 %
                 SUITE 7777                            Tax ID #
City/State/Zip   OUT IN THE PINES     NJ  08888-0000   Discount          0 %
Telephone        609 999-1234                          Unapl Credit   0.00

Terms  2  (0=COD 1=Csh 2=N10 3=N15 4=N20 5=N30 6=1N10 7=1N30 8=2/10 9=2/15)

YTD Sales   2138.94 Balance      0.00 On Order      0.00 Credit Limit     0

Region              Type Cust  RC     Bad Debt        Last Payment  12/17/86
Salesman            Mail Code         Fin. Chg        Last Session    /  /

Use fkeys below to Search Client File.
 1 Help  2 Prev  3 Next  4Review 5Print  6 Add   7 Edit  8Delete 9 Find 10 Menu
```

Figure 3. The client information screen.

use? Is it user friendly?" These questions can best be answered by running the program through its paces.

At first, the testing process involves a lot of "play" . This part of the debugging can often be fun. You create imaginary customers, and schedule fictitious sessions for them. You do a great amount of "what if" thinking, and create complicated test scenarios for the program that utilizes all of its features. Doing this problem will be brought to light, and that there will be few hidden bugs.

Once this initial testing is out of the way, the process becomes more serious. The next step is the actual use of the program in your everyday studio management tasks in parallel with your manual studio paperwork. Eventually, the program will have to stand on its own. It is at this stage of the testing that you really find out if the software, as it has been written, really suits your needs and your usual ways of doing things. If any additional program bugs should surface at this stage, things can get rather exciting and hectic, as these problems will interfere with getting your everyday work done. Any changes that must be made to the program at this stage, either to fix bugs or add new features, bring with them increased chances for more bugs to surface.

THE ACTUAL SOFTWARE

It is interesting to compare the organization of the actual program that we have had written to our original "wish list." The Automated Studio Management program that is the result of our efforts has the following hierarchy (order of flow) of information within the system:

1) Customer Information
2) Project
3) Session Bookings
4) Work Order
5) Usage Billings
6) Pre-Invoice Review
7) Invoicing and Accounting
8) Auxiliary and Miscellaneous Function

A "Project" is a set of rates for a particular set of sessions which cannot be quoted to a client unless information on that client (name, address, etc.) has been entered into the appropriate system data file. Recording sessions cannot then be booked for that client unless a project has been assigned and priced. This ensures that every session booked will be billed at the correct rate. A Work Order is then issued for each session. The Work Order lists the data on the session's setup and other requirements, and provides space to record the actual studio time and materials used during the session. This information is then entered into the system in the "Usage Billings" module of the program. The supplies used by each session are automatically subtracted from the studio inventory as this is done. At this stage, a session can be invoiced, or an "Pre-Invoice Review" can be generated in case adjustments to the charges being billed need to be made before the actual invoicing. Payments by clients can then be entered against their accounts as they are received. This data is automatically logged into the accounting portion of the system.

Other features of the program, such as tape library, report generation, and maintenance and housekeeping tasks, are handled as auxiliary functions of the system. Exactly how the various functions or modules of our Automated Studio Management System have been organized can be seen in the main menu screen display. *(Figure 1.)*

USER INTERFACE

The Automated Studio Management System that we have developed is a very large and complex piece of software, but every effort has been made to make it as easy to use as possible. The program is what is termed "menu driven," that is, extensive use has been made of menus, the computer keyboard function keys, and on-screen prompts to guide the user through the system. Of particular significance is the built-in help function, which makes instructions and advice instantly available at the touch of a key. Much thought had to go into the design of the many data entry and display screens to make them logical and easy to use. Here are a couple of sample screens from the system. *(Figures 2 and 3.)*

THE ACTUAL SYSTEM HARDWARE

Our Automated Studio Management System is now in use in both Sigma Sound Philadelphia and New York in multi-user mode. In both locations, the hub of the system is an IBM PC/AT connected to satellite terminals or computers via high speed (up to 115K bits per second) RS-232 lines. The use of the RS-232 format permits the utilization of simple twisted pair wiring as opposed to the multi-pair or coaxial cable necessary for other methods of data transfer. In our particular case, this type of networking is made possible by software and hardware add-ons such as Lan-Link, Multi-link, Connect-Com, etc., produced by Software Link, Inc. There are many other suitable networking products on the market.

The New York System consists of four IBM PC-compatible terminals (Kimtron KT-7s) connected to a PC/AT acting as server to them. Since the KT-7s are what are referred to as "dumb" terminals (simple data input and display devices with no built-in computing power of their own) all of the actual information processing is done in the PC/AT which divides its computing power among itself and the terminals. All the system and program files are stored on the server's hard disk drive.

The Philadelphia hardware setup is somewhat different. It consists of a central PC/AT with both dumb terminals and other PCs as satellites. Three terminals, along with a PC/XT type computer (which can act as a terminal, but is also available for independent use) are connected to the central server, which handles only the session scheduling and client billing functions of the system. Another PC/AT class computer is connected to the system to take care of bookkeeping (all of the bookkeeping for Sigma sound as a whole is handled in Philadelphia).

Although the actual networking necessary for multi-user implementation of our Studio Management System is made possible by the previously mentioned hardware and software add-ons, provisions did have to be made in writing our software to allow for this type of use. For instance, what happens if two users attempt to alter the same data file simultaneously? If no allowances are made for this, the result could be that one of the users will be locked out of that data (which is annoying rather than a serious problem), or the whole system could crash (which is a serious consequence, since valuable data could be lost). Data file handling within the system had to set up to properly accommodate such situations.

DOCUMENTATION

Another interesting aspect of this software development process is as follows: Initially, you will have to test and then use your program without any sort of instruction manual. At the same time, you do need some sort of instructional material in order to train your staff to participate in the program testing and use. So, you must set out to write the user's manual for the program as the software is being developed, and since the program is still in a continual state of change, your manual needs to keep changing in order to keep the manual up to date and accurate. In this dilemma, though, computers can come to your rescue. A good word-processing program can not only be an aid to writing your program documentation, but it also makes revisions to that material quick and easy. Also, with the program manual in electronic form, it also becomes easy to incorporate material from that document into an on line "help" function built into the software. Instructions and advice on program use can then pop up on the computer screen whenever needed.

FUTURE POSSIBILITIES

Initially, we set out to develop our Studio Management software for our own use, in part because there were few, if any, programs on the market like it. As the development of our program progresses, and we compare what we have to what is available, we are considering offering our package for purchase by other studios. Our plans for this have not been finalized.

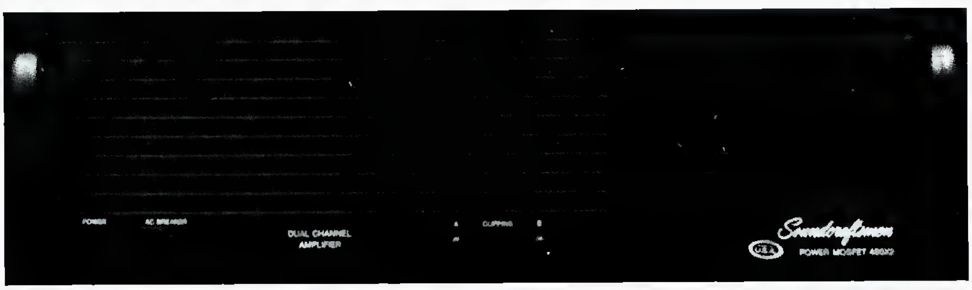
CONCLUSION

There were many aspects of developing our own Studio Management software package that we did not foresee when we started the project. Now that the process is drawing to a close, the question inevitably must be asked, "Would we do it again?" The truthful answer would have to be this: we would probably have some additional second thoughts about undertaking a similar project, knowing what we know now about just what actually is involved. At the same time, though, computer are becoming more and more a part of all aspects of our daily lives, especially in the studio (anyone with a current automated console knows just what we mean). Sigma Sound has always prided itself on attempting to be at the forefront of all areas of recording studio technology. With this in mind, the answer to the question, "Would we do it again?" would have to be *probably.* ■

Len Feldman

Lab Report

Soundcraftsmen 450X2 Stereo Power Amplifier



The Soundcraftsmen 450X2 amplifier.

GENERAL INFORMATION

Soundcraftsmen is a California based audio products company that has been turning out superb power amplifiers, preamplifiers, equalizers and other assorted professional and home audio components for many, many years. Their newest series of power amplifiers, like all of their products that I have tested in the past, are designed with a consistent philosophy which has been repeatedly stressed in their brochures and ads: "...to design and produce moderately-priced separate components with audio performance fully comparable to any competitive audio component, regardless of price. Performance is #1, and the objective of maintaining a moderate price level must be accomplished by innovative circuit design and a practical, economical approach to fabrication, given the obviously limited volume of this type of item."

The Model 450X2 represents another Soundcraftsmen innovation and advancement in power supply design. The new circuit is called Phase Control Regulation (PCR), but it has nothing to do with the audio phase response of the amplifier itself. Rather, PCR is a highly efficient means of controlling the average power supplied to the output stages of an amplifier. It involves a process of rapid switching which connects and disconnects an a.c. supply to a load for an optimized fraction of each a.c. cycle. With very light signal demands, only a small fraction of the a.c. cycle is used to provide power. With heavy demand, the full cycle is used.

Physically, the amplifier is rather simple. The standard rack-width front panel extends beyond the width of the chassis, so that the unit can be rack mounted easily. Thermal protection is provided by multi-sensor PCR (explained later on) as well as by a two-speed fan which provides forced air cooling. Air flow is from rear to front. In addition to providing high levels of power in the stereo mode into either 8-ohm or 4-ohm loads, the amplifier can operate continuously at load impedances as low as 2 ohms. The amplifier can operate in the "bridged" or mono mode by simply moving a slide switch on the rear panel. In mono mode, the amplifier is rated at 600 watts into an 8-ohm load.

CIRCUIT HIGHLIGHTS

To fully understand the benefits of PCR, consider the block diagram *(Figure 1)* which depicts a more conventional amplifier power supply. a.c. line rectifier, supplying pulsating d.c. which is then filtered by capacitors to provide a d.c. voltage at the output. *Figure 2* shows a block diagram of Soundcraftsmen's Phase Control Regulation circuit. Input a.c. voltage is applied to the power transformer for line isolation and input voltage adjustment. The a.c. voltage from

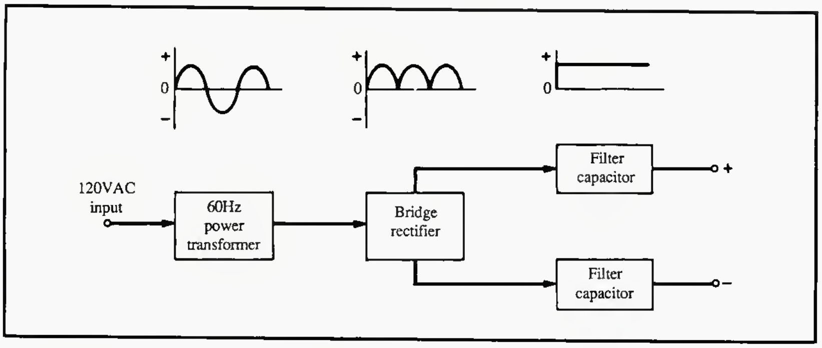


Figure 1. A conventional power supply.

the transformer's secondary winding is rectified by silicon controlled rectifiers (SCR) and smoothed by filter capacitors to provide output d.c. voltage. During operation of the amplifier, the output level of the d.c. voltage is constantly compared to a reference voltage and an error-eliminating signal is applied to the Phase-Control Regulator. It is this regulator that controls the conduction time of the SCR's, maintaining the output voltage at a precise fixed level. The positive and negative output voltages are the supply rails to the amplifier stages.

This method of regulation of an amplifier's power supply has a lot to do with its power output capabilities. Most conventional amplifiers have poor supply voltage regulation. As the load demands more power, d.c. supply voltages tend to drop. This can significantly reduce power output during high volume level operation, especially when musical dynamic peaks of more than a few milliseconds are encountered. Soundcraftsmen's approach differs from others in that they believe continuous power to be of greater importance to accurate musicality than peak power. The dynamic headroom test which I perform for all amplifiers that I test involves the use of a 20 millisecond burst of 1 kHz signal, followed by 480 milliseconds of lower-level (-20 dB) 1 kHz signal. This is supposed to replicate what happens to an amplifier when it is handling short-term musical peaks, and almost invariably, most amplifiers can deliver power levels well above their continuous ratings for such a short period. Soundcraftsmen's engineers maintain that accurate musical reproduction of the complex content of a very loud musical note requires far more than a 20 millisecond time frame. Thus, in their high-powered amplifier designs (of which the 450X2 is certainly one), the emphasis is on maintaining high output power over the long term, rather than for 20 milliseconds as dictated by the IHF Dynamic Headroom test.

All of this serves to explain why the 450X2 exhibited a rather minimal dynamic headroom during my tests, but more than met its continuous power output ratings at both 8-ohm and 4-ohm load conditions. Since 205 watts per channel is more than adequate for driving most modern speaker systems well beyond lifelike loudness levels of musical reproduction in most indoor environments, there's really no need to be concerned with short-term power peaks beyond that high power level.

Turning to the output stages, they utilize power MOSFETs which are particularly suited to the high-current requirements that have been recognized in recent years as being of such importance when reproducing today's new digital program sources. The use of MOSFETs eliminates the need for conventional current limiting protection circuitry. Each channel is protected from overheating by a system of thermal sensing devices that control the speed of an internal cooling fan and, if necessary, reduce the a.c. input to the amplifier. MOSFETs cost more than bipolar devices having similar power ratings, but Soundcraftsmen has evidently been able to incorporate them into this and other high-powered amplifiers by combining them with the cost savings in the power supply brought about by the PCR circuit and by somewhat simpler mechanical packaging.

CONSTRUCTION AND CONTROL AND PANEL LAYOUTS

The heavy duty front panel of the 450X2 contains an a.c. power switch/circuit breaker, a pair of clipping indicator LEDs and a pair of rugged carrying handles. The main chassis is constructed of 14-gauge welded steel. XLR and barrier terminal strip as well as 1/4-inch phone jack inputs are found on the rear panel for balanced or unbalanced input connections. There are input level controls for each channel, each of which is adjustable over a range greater than 26 dB. The mono/stereo slide switch is located beneath the input barrier strip terminals, and when it is flipped to the mono or "bridged" setting, a single load must be connected between the positive-labeled (+) speaker output five-way binding posts. The right third of the rear panel is given over to the air-intake slots for the internally mounted cooling fan.

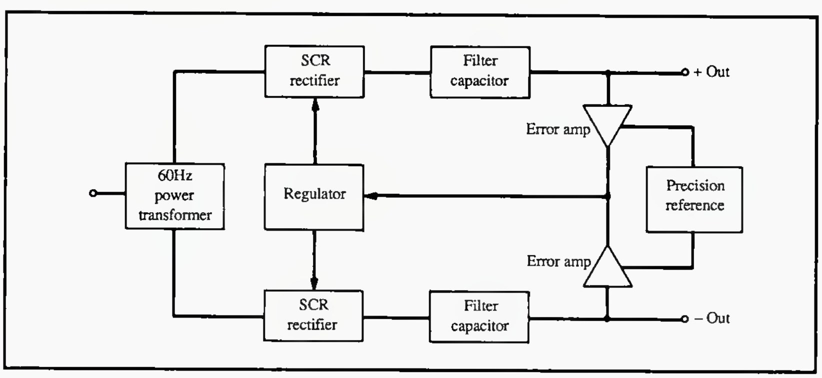


Figure 2. The Soundcraftsmen Phase Control Power Supply.

A grounded three-wire power cord is provided.

LAB MEASUREMENTS

Results of most of my measurements of the Soundcraftsmen 450X2 are summarized in the VITAL STATISTICS chart at the end of this report. Connected to 8-ohm resistive loads, the 450X2 delivered 235 watts per channel of continuous power at 1 kHz for its rated THD-plus-Noise level of 0.05% as against 205 watts claimed by Soundcraftsmen. For a 20 Hz test signal, output power rated THD+N was 219 watts, while at 20 kHz,output power reached 231 watts per channel for the same level of THD+Noise. All measurements were made with both channels driven. At the rated power output level of 205 watts per channel, THD+Noise measured a very low 0.004% at mid-frequencies, 0.005% at 20 Hz and 0.018% at 20 kHz. SMPTE-IM distortion measured only 0.005% at rated output and the amplifier could be driven to an equivalent output of 235 watts per channel before the IM reading increased to its rated value of 0.02%. *Figure 3A* is a graph showing how distortion, under 8-ohm load conditions, varies as a function of power output at four key frequencies. *Figure 3B* is a similar representation, except that the loads were changed to 4 ohms.

Soundcraftsmen does not provide an "FTC Power Rule" type of power rating for 4-ohm operation. They do state, however, that 1 kHz power, at its respected THD level, with 4-ohm loads would occur at 315 watts and that IHF Dynamic Power at 4-ohms was 420 watts. Comparing these figures with those provided for 8-ohm clipping level and 8-ohm dynamic power (stated as 225 watts and 256 watts respectively) we surmised that the amplifier should be able to deliver around 300 watts per channel at 4-ohms for the same low rated THD+Noise level of 0.05%. It did just a shade better than at mid-frequencies, delivering 305 watts per channel, but couldn't quite make the 300 watt level for a distortion reading of 0.05% at 20 Hz or at 20 kHz. Distortion for 300 watts of output at 20 Hz measured 0.3%, while at 20 kHz it increased to 0.4%. To reduce THD to the rated 0.05% at those frequency extremes, I had to back down until power output was 275 watts at 20 Hz and 260 watts at 20 kHz.

Damping factor at 8-ohms, referred to a 50 Hz test signal measured 205, allowing for some minimal resistance of our heavy gauge speaker cables. Signal-to-noise ratio measured 82 dB referred to 1 watt output. Soundcraftsmen chose to quote S/N with respect to full rated output. Since 205 watts is exactly 23.2 dB greater than 1 watt, that would put our reading relative to rated output at 105.2 dB, or almost exactly the figure quoted by Soundcraftsmen. Input sensitivity for 1 watt output was 72 millivolts, referred to 8-ohm loads.

CCIF-IM (twin tone intermodulation distortion) measured an almost imperceptible 0.0025% at 8-ohms; 0.003% at 4-ohms. Frequency response of the amplifier extended from 5 Hz to 60 kHz for the -1.0 dB roll-off points, and 2.5 Hz to 125 kHz for -3 dB cut-off points. As I expected, Dynamic Headroom was very low: Only 0.7 dB at 8 ohms and a slightly greater 1.0 dB at 4 ohms (based upon my assumption that rated output at 4- ohms is actually 300 watts per channel). Dynamic Headroom is an informational specification and not a qualitative one. The low dynamic headroom of this amplifier is in keeping with Soundcraftsmen's avowed design philosophy of "stiff" well regulated power supplies.

COMMENTS

For some reason which I have never been able to fathom, Soundcraftsmen, a company that offers a wide variety of professional and home audio equipment, has never gotten the credit and recognition that it deserves. Perhaps it's because management of the company refrains from the kind of flamboyant advertising and promotion that is practiced by less technically advanced companies. Did you know, for example, that it was Soundcraftsmen's engineers who first designed, developed and marketed "Class H" signal-tracking, power supply circuitry back in 1976. Ralph Yeomans, the President of Soundcraftsmen, and Paul Rolfes, its Engineering V.P. have on more than one occasion, outlined the basic philosophy of their company to me: "...to design and produce moderately-priced separate components with audio performance fully comparable to any competitive audio component, regardless of price. Performance is #1, and the objective of maintaining a moderate price level must be accomplished by innovative circuit design and a practical, economical approach to fabrication, given the obviously limited volume of this type of item."

The 450X2 is a rugged amplifier that operates reliably in

Figure 3(A).Distortion plotted against power at 8 ohms.

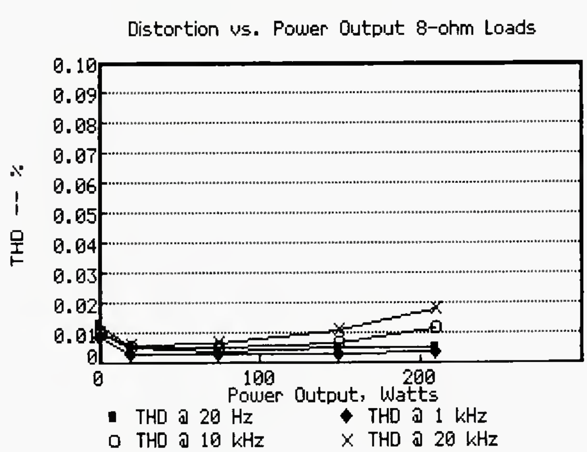
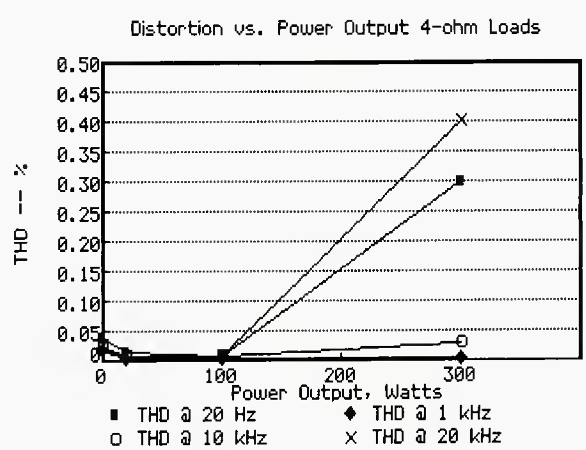
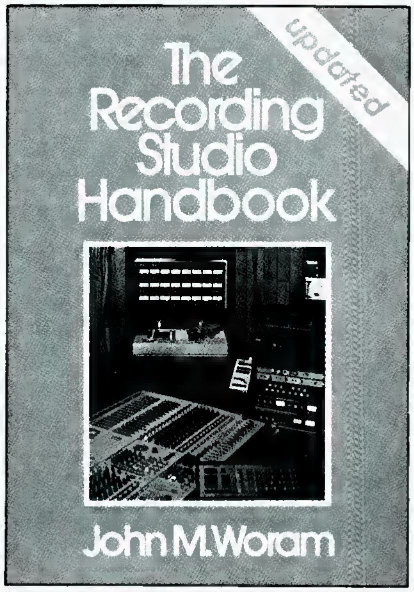


Figure 3(B). Distortion plotted against power at 4 ohms.



For Your Audio Library

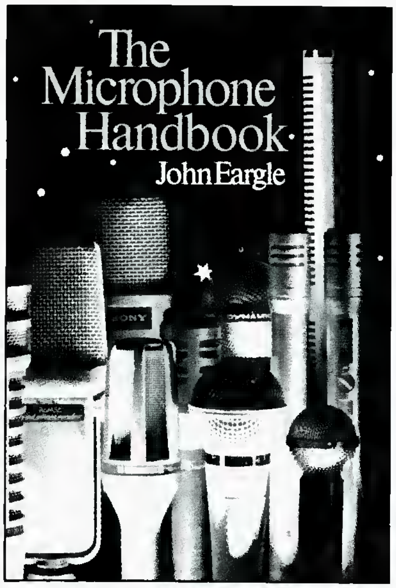
The Recording Studio Handbook



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.

__________ Both books at $67.50 plus $3.00 for postage and handling. (Over 5% savings!)

Outside the US, checks must be made in US funds drawn on a US bank.

Mastercard ☐ Visa ☐ Exp. Date / __ / __

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Name ____________________

Address ____________________

City __________ State ______ Zip ______

Signature ____________________

Send to: ELAR Publishing, 1120 Old Country Rd., Plainview, NY 11803 a fixed studio location. I listened to the 450X2 using a variety of loudspeakers, from inexpensive small monitoring types (being careful not to overdrive them with the awesome power available from this amp) to low-efficiency, highly accurate "reference" speakers that I use in my lab. I can best describe the performance of this carefully designed amplifier by stating that it brought out the best of the loudspeaker system with which I tried it. I sensed an effortlessness about some of the musical crescendos as reproduced from some of my recently acquired wide dynamic range CD spectaculars. But having confirmed that great gobs of dynamic headroom are not necessary if you have adequate continuous power available, I settled down to more musical fare and confirmed, in my own mind, that the chief difference between this amplifier and some of the high powered amps costing several times as much is price! In my view, you can spend two, three, even five times as much as the Soundcraftsmen 450X2 costs, and you won't get a better, more reliable, or more accurate sounding power amplifier.

VITAL STATISTICS

MAKE & MODEL: Soundcraftsmen Model 450X2

SPECIFICATION	MFR'S CLAIM	db MEASURED
Power Output for Rated THD		
(8-ohms, 20Hz-20kHz,Watts)	205	235
THD at Rated Power(%)	0.05	0.004
Power at Clipping (Watts)		
8 ohm loads	225	240
4 ohm loads	320	330
2 ohm loads	350	350
Dynamic Power (20ms burst,2/sec.)		
8 ohm loads (watts)	256	275
4 ohm loads (")	420	420
2 ohm loads (")	552	550
Dynamic Headroom		
8 ohm loads	0.9 dB	0.7 dB
4 ohm loads	1.2 dB	1.0 dB
Frequency Response		
for 20 Hz to 20 kHz	+0,-0.3 dB	+0,-0.1 dB
for 5 Hz to 100 kHz	-1.2 dB,-1.9dB	-1.0 dB,-1.5 dB
S/N (A-weighted, re: Rated Output)	105 dB	105.2 dB
IM Distortion,Rated Output,8 ohms	0.02%	0.005%
Damping Factor (50 Hz, 8 ohms)	200	180
Slew Rate (Volts/microsecond)	50	Confirmed
Input Sensitivity for 1 watt out	72 mV	71 mV
Dimensions (HxWxD, inches)	5-1/4x19x11-3/4	Confirmed
Weight	30 lbs.	Confirmed
Suggested Retail Price		
Model 450X2	$699.00	
Model 450X2M (W/LED Meters)	$799.00	

■

Circle 60 on Reader Service Card

Buyer's Guide: Power Amplifiers

Understanding the Specs-Power Amplifiers

The specification charts that follow are pretty well standardized throughout the industry.

The first three specification columns ask for the power of the amplifiers under carefully specified standards: In all cases we want both channels of the amplifier driven to full output, but we ask for the power of one channel. Dual channel amplifiers do not work in "live" situations any other way, though it is true that they seldom are driven (nor should they be) to full output. Still, this is a standardized test requirement for comparison purposes.

The three columns represent 8, 4, and 2-ohm loads. Again these are for standardized measurement purposes. Under actual working conditions, an amplifier may be connected to any number of parallel/series speakers, thus presenting a load that might actually be anywhere between (or even outside) our 2-8 ohm bench-test requirement.

Power Bandwith is the spec that follows the general power requirements. The continuous specs mentioned above are usually measured at around 1 kHz. It is common for a power amplifier to provide less power at frequency extremes, so power bandwidth provides a frequency range around which the amplifier will provide the specified full power.

The four following columns request distortion measurements. We ask for one watt and full power for both intermodulation distortion (IM) and total harmonic distortion (THD). Not all manufacturers use these specific standards for their internal measurements, so it cannot be assumed that a lack of entry by a manufacturer in one or more, means any more than they use different measurement standards.

Following the distortion columns we ask for the frequency response at one watt. This is an industry-standard way of giving frequency response, since it accurately represents the frequency bandwidth of their units. As part of the spec we ask for the response plus or minus so many decibels. That is the number in the column below the bandwith.

Sensitivity, the next column asks for the input voltage (from the mixer or whatever)that is required to drive the amplifier to full output.

Next, the **Dimensions, Weight,** and **Price** columns are self-explanatory.

Finally, there is the **Features** column. Here, we have invited each manufacturer to give us a few of his own words on the special qualities of his product. This information has been submitted at the discretion of the manufacturers.

You may also notice that one or more companies may be missing from these listings. We have extensive lists of who makes what, and do try to get everyone. But sometimes we miss. And sometimes, as many times as we do write and even call, return material is not forthcoming.

Model	Number of Channels	Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-kHz.	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for Full Output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
AUDIO MEDIA RESEARCH															
PMA-200	2	100	n/a	n/a	5-100k	0.01	n/a	n/a	0.05	5-100k 3	1	5.25 19 12.2	29	399.50	Has automatic thermal shutdown and DDT compression.
BOULDER AMPLIFIERS															
500	2	500	250		0.015-100k	.001	.003	0.0	.005	20-20k 0.04	1.75	7 19 15.5	51	2,875.00	Two-stage design amplifier with stereo mono switch and balanced inputs.
160	2	160	80		0.015-100k	.001	.003	0.0	.005	20-20k 0.04	1.1	3.5 19 15.5	40	1,190.00	Same as above.
100	2	30	40		0.015-100k	.001	.003	0.0	.005	20-20k 0.04	0.77	1.75 19 14	28	750.00	Same as above.
BGW SYSTEMS															
750E	2	350	600	850	20-20k		0.01		0.03	10-100k 3	1.42	7 19 13	50	1,699.00	Has toroidal power transformer, 60 dB LED display, status indicators, and bridge mono operation.
250E	2	100	150		20-20k		0.02		0.1	10-100k 3	1.41	5.5 19 11.75	40	1,099.00	Has multi-color VU meter, complete loudspeaker protection and optional input transformers.
8000	2	225	350	600	20-20k		0.05		0.1	10-100k 3	1.23	5.25 19 13	49	1,399.00	Has toroidal power transformer, complete loudspeaker protection, optional input transformers.
7500	2	200	300		20-20k		0.05		0.1	10-100k 3	1.15	5.25 19 13	44	999.00	Has complete loudspeaker protection and optional input transformers.
6500	2	100	130		20-20k		0.02		0.19	10-100k 3	1.23	3.5 19 12	30	749.00	Has thermal circuit breaker.
85	2	35	45		20-20k		0.03		0.19	10-100k 3	0.84	1.75 19 11.5	19	519.00	Has toroidal power transformer and front headphone jack.
150	2	50	75		20-20k		0.02		0.1	10-100k 3	1	1.75 19 11.5	22	799.00	Has toroidal power transformer, short circuit protection and optional input transformers.
SPA-3	3	250	400		20-20k				0.19	10-100k 3		5.25 19 13	43	2,499.00	Three-channel signal processing amp with 24 dB/octave crossovers high-pass filter and parametric eq

Model	Number of Channels	Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-kHz.	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for Full Output, V	Dimensions, H/W/D, In.	Weight, lbs.	Price, $	Features
CARVER															
PM-175	2	175	250	250	20-20k	0.1	0.1	0.5	0.5	5-80k 3	1.5	3.5 19 11.5	19	649.00	Slow startup, muted inputs, onboard modules-i.e. comp/limiter, crossover.
PM-350	2	350	450	450	20-20k	0.1	0.1	0.5	0.5	5-80k 3	1.5	3.5 19 11.5	21	849.00	As above.
PM-1.5	2	450	600	535	20-20k	0.1	0.1	0.5	0.5	5-80k 3	1.5	3.5 19 12	21	1,050.00	Used for touring.
PM-20X	2	465	600	450	20-20k	0.1	0.1	0.5	0.5	5-80k 3	2.2	3.5 19 10.8	10	1,495.00	Has slow startup, muted inputs and 1000 watt (at 4 ohms) output.
CARVIN															
DCA-800	2	200	300	400	20-20k	n/a	n/a	0.05	0.01	5-60k 3	2	5.25 19 12	44	579.00	Designed for heavy-duty touring or high fidelity requirements.
DCA-300	2	100	150	300	20-20k	n/a	n/a	0.05	0.01	5-60k 3	2	5.25 19 12	37	419.00	Low-power version of the DCA-800.
DCM-301	1	100	160	300	20-20k	n/a	n/a	0.05	0.01	5-60k 3	1	5.25 19 12	35	349.00	High-power mono amplifier designed for monitor systems. Has graphic equalizer built-in.
DCM-151	1	70	100	150	20-20k	n/a	n/a	0.05	0.01	5-60k 3	1	5.25 19 12	30	299.00	Low-power version of DCM-301.
CREST AUDIO															
8001	2	600	1000	1300	20-20k	0.01	0.05	.005	0.05	1-50k 1	.775	5.25 19 14.5	64		
6001	2	450	700	1300	20-20k	0.01	0.05	.009	0.05	1-50k 1	.775	5.25 19 14.5	58		
4000/ 4001	2	325	550	800	20-20k	0.01	0.01	.003	0.03	1-50k 1	1.21	5.25 19 13	58	2,079.00 2,279.00	4000 has LED output meters. 4001 has signal present LED.
3000/ 3001	2	240	430	640	20-20k	0.01	0.01	.003	0.03	1-50k 1	1.1	5.25 19 11.5	46	1,589.00 1,789.00	3000 has LED output meters. 3001 has signal present LED.
2501A	2	200	325	500	20-20k	0.01	0.01	.006	0.06	10-100k 3	1	3.5 19 13	38	1,279.00	
2001A	2	125	200	300	20-20k	0.01	0.01	.006	0.06	10-100k 3	0.8	3.5 19 13	32	1,059.00	
1501A	2	75	125		20-20k	0.04	0.04	.009	0.03	10-80k 3	0.7	1.75 19 10.5	17	839.00	

Model	Number of Channels	Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-kHz.	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for Full Output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
1001A	2	40	70		20-20k	0.04	0.04	.009	0.03	10-80k 3	0.42	1.75 19 10.5	17		
FA-800	2	210	335		20-20k	0.01	0.01	.007	0.05	10-40k -0.2	1	3.5 19 13	32	792.00	
Power-line 400	2	290	450		20-20k	0.01	0.01	.007	0.05	0-30k -0.2	1.2	3.5 19 13	38	1,239.00	
Power-line 300	2	220	325		20-20k	0.01	0.01	.007	0.05	0-30k -0.2	1.05	3.5 19 13	32	939.00	
CROWN															
Delta Omega 200	1	730	1300	2000	DC-45k	0.05	0.05	0.05	0.05	DC-45k 0.1	1.7	8.75 19 16.5	92	3,195.00	Compensates for load resistance to improve low-frequency and transient response.
PSA-2X	2	220	350		1-20k	0.01	0.01	.002	.002	20-20k 0.1	2.1	7 19 4.75	57	1,699.00	Has 2-speed fan, XLR input option, filters, tone generator, compressor and distortion indic.
PSA-2DX	2	220	350		1-20k	0.01	0.01	.002	.002	20-20k 0.1	2.1	7 19 4.75	57	1,929.00	Same as above with signal strength LED display.
Micro-Tech 1200LX	2	320	470	600	20-20k	0.05	0.05	0.1	0.1	20-20k 0.1	2.2	3.5 19 16	44.2	1,369.00	Same as MT-600LX, 1200 watts mon
Micro-Tech 600LX	2	220	275	300	20-20k	0.05	0.05	0.1	0.1	20-20k 0.1	2.2/ .775	3.5 19 16	39.2	1,169.00	Has distortion indicator.
DC-300A	2	155	250		1-20k	0.05	0.05	.001	.001	DC-20k 0.1	1.75	7 19 9.5	45	999.00	Same as D-75 with distortion indicator.
Power Base-1	2	200	200		20-20k				0.1	20-20k 0.1	2.2/ .775	3.5 19 16	30	729.00	Low-cost model for musicians' PA systems. 400 watts mono.
D-75	2	40	55		20-20k	0.05	0.05	.001	.001	20-20k 0.1	.812	1.75 19 9	10	524.00	Fully protected against shorts, mismatches, opens, and RF burn-outs.
DAVID HAFLER															
P500	2	255	400		10-40k	.007	.007	0.04	0.04	4-160k -3	1.55	7 19 13	53	995.00	Includes balanced inputs, front panel controls and clipping indicators. Available as kit.
P505	2	255	400		5-40k	.007	.007	0.04	0.04	4-160k -3	1.55	7 19 13	48	895.00	Has rear panel controls, unbalanced inputs and 3-speed cooling fan.
P225	2	115	175		6-60k	.005	.005	0.02	0.02	2-160k -3	1.55	5.25 19 10.5	28	525.00	Has rear panel controls with optional balanced inputs and XLR connectors available.
P125	2	63			10-40k	.005	.005	.009	.009	4-200k -3	1.1	3.5 19 9	19	395.00	Same as P225.

Model	Number of Channels	Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-kHz.	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for Full Output, V	Dimensions, H/W/D, In.	Weight, lbs.	Price, $	Features
HILL AUDIO															
DX-300	2	200	200	n/a	20-20k	0.01	0.01	0.01	0.02	20-20k 0.05	1.55	2rack 19 8.5	16	669.00	Features transformer coupled driver stage, full relay shutdown protection and toroidal transformers.
DX-800	2	250	400	n/a	20-20k	0.01	0.01	0.01	0.02	20-20k 0.05	1.55	2rack 19 13	29	899.00	Same features as the DX-300 with variable speed, servo-controlled cooling fans.
DX-1500	2	300	500	750	20-20k	0.01	0.01	0.01	0.02	20-20k 0.05	1.55	2rack 19 13	34	1,099.00	Same as the DX-800.
JBL PROFESSIONAL															
6215	2	35	45	n/a	20-20k	0.1	0.1	0.1	0.1	20-20k	1.1	1.75 19 9	10.5	576.00	Full-power frequency response is 20-20k, and it is a UL approved single power supply stereo amp.
6230	2	75	150	n/a	20-20k	0.1	0.1	0.1	0.1	20-20k	1.1	5.25 19 11	26.3	618.00	Same as above.
6260	2	150	300	n/a	20-20k	0.1	0.1	0.1	0.1	20-20k	1.1	7 19 11	44.5	870.00	Same as above.
6290	2	300	600		20-20k	0.1	0.1	0.1	0.1	20-20k	1.1	7 19	63	1,299.00	Same as above with front panel headphone jack.
PANASONIC/RAMSA															
WP-9220	2	200	300	n/a	10-85k	0.05	0.05	0.05	0.05	20-20k 0.5	1.23	5.68 18.8 15.1	38.6	899.00	Has bridge mode, 2-speed fan remote monitor output, and reative load stable.
WP-9110	2	100	150	n/a	10-85k	0.05	0.05	0.05	0.05	20-20k 0.5	1.23	3.9 18.8 15.2	28.6	699.00	Same as WP-9220.
WP-9055	2	50	75	n/a	10-85k	0.05	0.05	0.05	0.05	20-20k 0.5	1.23	2.3 18.8 13.2	19	475.00	Same as WP-9220 but has convection cooling.
PEAVEY ELECTRONICS															
DECA 424	2	125	200		20-20K	0.1		0.1	0.15	10-40K -3	1.0	3.5 19 14	27	699.50	Fan cooled, digital energy conversion, ddt compression.
M-4000	2	120	200	120	10-30K			.005	0.05	10-40K -1	1	5.25 18.87 12.38	35	599.50	Fan cooled, ddt compression, plug-in crossover option, transient-free turn on.
M-7000	2	200	350	200	10-30k			.005	0.05	10-40k -1	1.4	4.25 18.87 12.38	42	749.50	As Model M-4000 above.
CS-1200	2	350	450	350	20-40k			.005	0.05	10-40k -1	1.4	7.5 19 18	70	1299.00	As Model M-4000 above.
CS-900	2	250	450	350	220-40k		.005	0.05	10-40k	1.4 -1	5.25	50 19 15	50	899.00	As Model M-000 above.

Model	Number of Channels	Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-kHz.	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for Full Output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
DECA1200	2	300	600		20-20k	0.1		0.1	0.15	10-40k -3	1.3	3.5 19 15	26	1399.00	As Model M-4000 above plus digital energy conversion amplifier.
DECA 724	2	225	350		20-20k	0.1		0.1	0.1	10-40k -3	1.0	3.5 19 15	35	999.50	As Model DECA 1200 abbve.
QSC AUDIO															
1080	2	35	50	60	20-20k	.012	0.01	0.1	0.1	20-20k -1	.83	1.75 19 8.7	12	488.00	Features dual power supplies, extensive speaker and ampifier protection and accessory inputs.
1200	2	100	150		20-20k	.012	0.01	0.1	0.1	20-20k -1	1	5.25 19 9.5	24	548.00	Same as above.
1400	2	200	300		20-20k	.012	0.01	0.1	0.1	20-20k -1	1	5.25 19 9.5	34	768.00	Same as above.
1700	2	325	500		20-20k	.012	0.01	0.1	0.1	20-20k -0.5	1	7 19 10.8	57	1,098.00	Same as above.
3200	2	110	140		20-20k	.012	0.01	0.1	0.1	20-20k 0.1	1	1.75 19 14.6	26	958.00	Features completely independent audio channels, recessed gain controls and status indicators.
3350	2	200	300	450	20-20k	.012	0.01	0.1	0.1	20-20k 0.1	1	3.5 19 15.9	41	1,248.00	Same as above.
3500	2	300	450	700	20-20k	.012	0.01	0.1	0.1	20-20k 0.1	1	3.5 19 15.9	50	1,488.00	Same as above.
3800	2	375	600	1100	20-20k	.012	0.01	0.1	0.1	20-20k 0.1	1	5.25 19 15.9	75	1,958.00	Same as above.
RANE															
MA-6	6	100	150	150	20-20k	0.1	0.1	0.2	0.2	5-80k	.775	5.25 19 11.5	44	1,299.00	Has 6 independent channels and it is autobridgeable to 300 watts at 8 ohms with built-in limiters.
HC-6	12	.450 450mW	n/a	n/a	5-24k	0.05	0.09	0.09	0.09	5-24k	n/a 19	1.75 8.5	8	349.00	Six-way stereo headphone amp for either distributed stereo signal or 6 independent programs.
SOUNDCRAFTSMEN															
PM860	2	210	315	450	20-20k	0.05	0.05	.008	0.05	20-20k 0.1	1.2	5 8.5 14	20	599.00	Has high current design to allow stability with 2 ohm speaker loads

Model	Number of Channels	Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/-dB	Sensitivity for Full Output, V	Dimensions H/W/D in.	Weight, lbs.	Price $	Features
450X2	2	210	315	450	20-20k	0.05	0.05	.008	0.05	20-20k 0.1	1.2	5.25 19 11.75	28	749.00	High current MOSFET amp with balanced or unbalanced inputs.
900X2	2	375	675	900	20-20k	0.05	0.05	.008	0.05	20-20k 0.1	1.22	5.25 19 16.5	59	1,499.00	Same as above.
RA7501	2	275	420	320	20-20k	0.05	0.05	0.05	0.05	20-20k 0.1	1.21	7 19 15	47	899.00	Class H signal tracking design for maximum efficiency.
SUBSTRATE ENGINEERING															
A-600	2	150	290	400	20-20k	0.05	0.15	0.01	0.09	10-100k	.775	5.25 19 12	30	699.00	Has built-in limiter, DC offset protection, thermal protection, and internal fan cooling.
UREI															
6500	2	275	450	600	20-20k	0.05	0.1	0.1	0.1	20-20k	1.1	7 19 16	84	2,996.00	Dual amplifier with pull-out channel modules, twin multi-speed fans and rear panel BNC connectors.

Manufacturers' Addresses

AMR
Rt.2,Highway 503
Decatur, MS 39327

Boulder Amplifiers
Silver Lake Research
3101 Third St
Boulder, CO 80302

BGW Systems, Inc.
13130 S. Yukon Ave.
Hawthorne, CA 90250

Carver Corp.
PO Box 1237
Lynwood, WA 98046

Carvin Mfg. Corp.
1155 Industrial Ave
Escondido, CA 92025

Crest Audio
150 Florence Ave
Hawthorne, NJ 07506

Crown International
1718 W. Mishawaka Rd
Elkhart,IN 46517

The David Hafler Co.
5910 Crescent Blvd.
Pennsauken, NJ 08107

Hill Audio, Inc.
5002B N. Royal Atlanta Dr.
Tucker, GA 30084

JBL Professional
8500 Balboa Blvd.
Northridge,CA 91329

Panasonic/Ramsa
6550 Katella Ave.
Cypress,CA 90630

Peavey Electronics
711 A Street
Meridian, MS 39301

QSC Audio Products, Inc.
1926 Placentia Ave.
Costa Mesa, CA 92627

Rane Corporation
6510 216th SW
Mountlake Terrace, WA 98043

Soundcraftsmen
2200 So. Ritchey
Santa Ana, CA 92705

Substrate Engineering
PO Box 4312
Davis, CA 95617

UREI
8500 Balboa Blvd.
Northridge, CA 91329

New Products

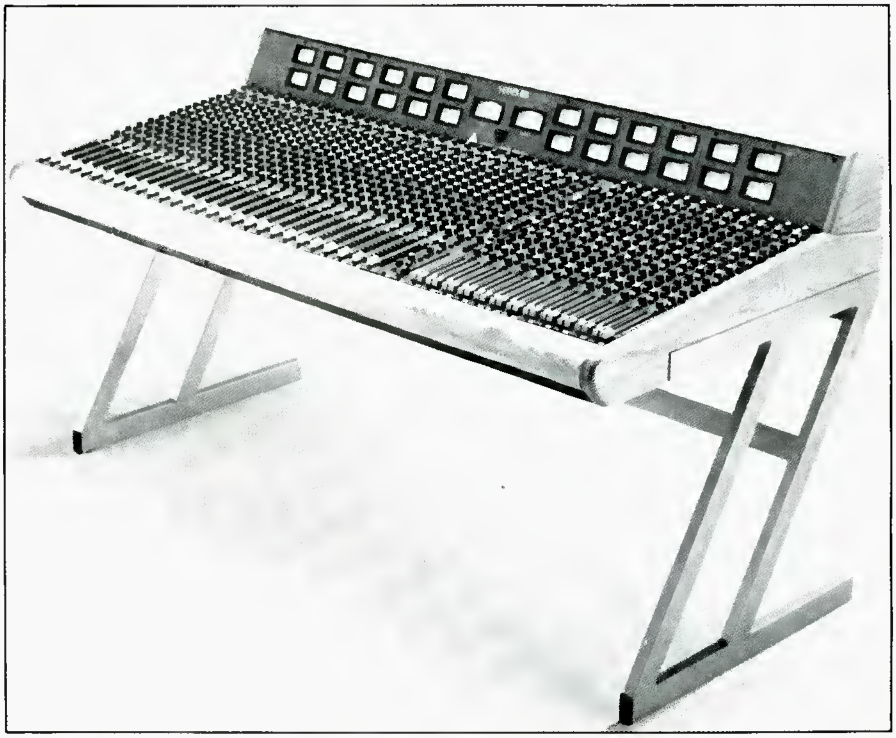
MULTITRACK CONSOLE

Trident Audio USA has announced the Trident 24 Multitrack Recording Console. This new version is available in 28 and 36 input frames with 24 discrete output buses and 24 track monitoring and metering. The console features separate mic and line inputs with phase reversal, 4 band eq with variable hi-pass filter, 8 auxillary sends, auto muting, solo in place, monitor fader reverse, direct outputs and four echo returns.

Mfr: Trident Audio USA

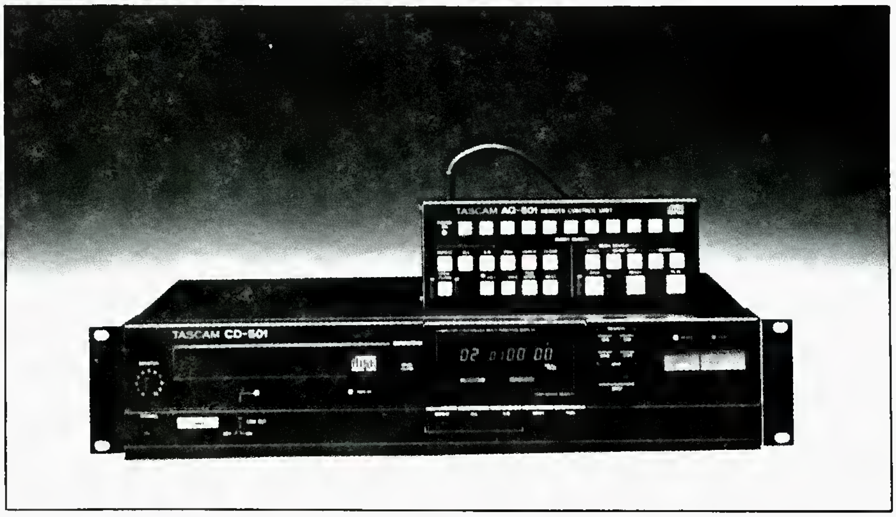
Price: $16,900.00 for 28 inputs and $19,900.00 for 36 inputs.

Circle 61 on Reader Service Card



BROADCAST CD PLAYER

Tascam introduces its new CD-501 compact disc player, designed to bring audiophile-quality digital play back to the broadcast industry. The rack-mounted CD offers balanced XLR +4dBm outputs, fixed and variable monitor outputs for com plete room compatiblity. CD-501 is engineered specifically for the broadcast environment. Two CD-501's can be operated from a single hard-wired remote control unit due to a special link connector. The CD-501 offers direct access to up to 99 tracks while the Track Skip function automatically locates the beginning of the next selection up or down the disk. The Repeat function is able to replay single selections, programmed sequences or entire discs. A 3-beam laser pickup ensures perfect alignment and instantaneous error correction. Double oversampling (88.2kHz), advanced design digital and 7th order analog filters are used.



Mfr: Tascam

Price: $1095.00.

Circle 62 on Reader Service Card

PATCH LIBRARY

MusicData U.S.A. is introducing a new Patch Library for the Yamaha FB-011 FM Sound Generator. The set features 128 new programs designed for professional applications. The Library is available on Commodore-64 computer disks and is compatible with Dr. T's and most other voicing software.

Mfr: MusicData U.S.A.

Price: $20.00

Circle 63 on Reader Service Card

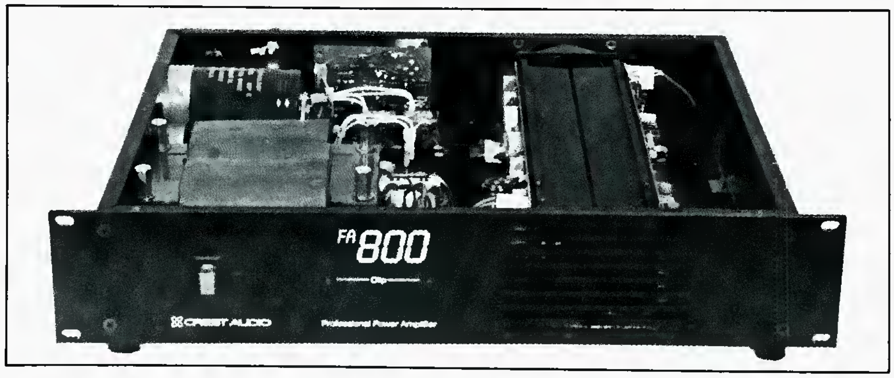
AUDIO POWER AMP

Crest Audio has introduced the Model FA800 Professional Power Amplifier. Rated at 240 watts/channel into 8 Ohms, 400 watts/channel into 4 Ohms and occupies 2 rack spaces. The FA800 also features back to front cooling, completely modular construction, full-loaded protection and active balanced inputs.

Mfr: Crest Audio

Price: $792.00

Circle 64 on Reader Service Card



SEQUENCING SYSTEM

The CMS Visual Sequencing (CMS-VSS) System includes a Roland compatible MIDI interface with 1 input and 4 outputs, tape in and out and metronome out. All this located on a 1 space 19-in. panel for front or rear rack mounting. The CMS-VSS is designed around a high performance 8mhz turbo mother board with 640K of RAM, Hercules compatible Hi-Res graphics card. TTL Amber monitor, printer port and 2 360K floppy drives all in a 4 space rack enclosure. Software with the system includes MS- DOS 3.1 and sequencer Mach-1 sequencing. All IBM compatible software will operate on the CMS-VSS. Options include 20-160 megabyte hard drives, modems, color monitors, printers, 80186 motherboards for a 5 times speed increase and 80286 mother boards for a 10 times speed increase are available.

Mfr: CMS Computer Music Systems

Price: $1299.00.

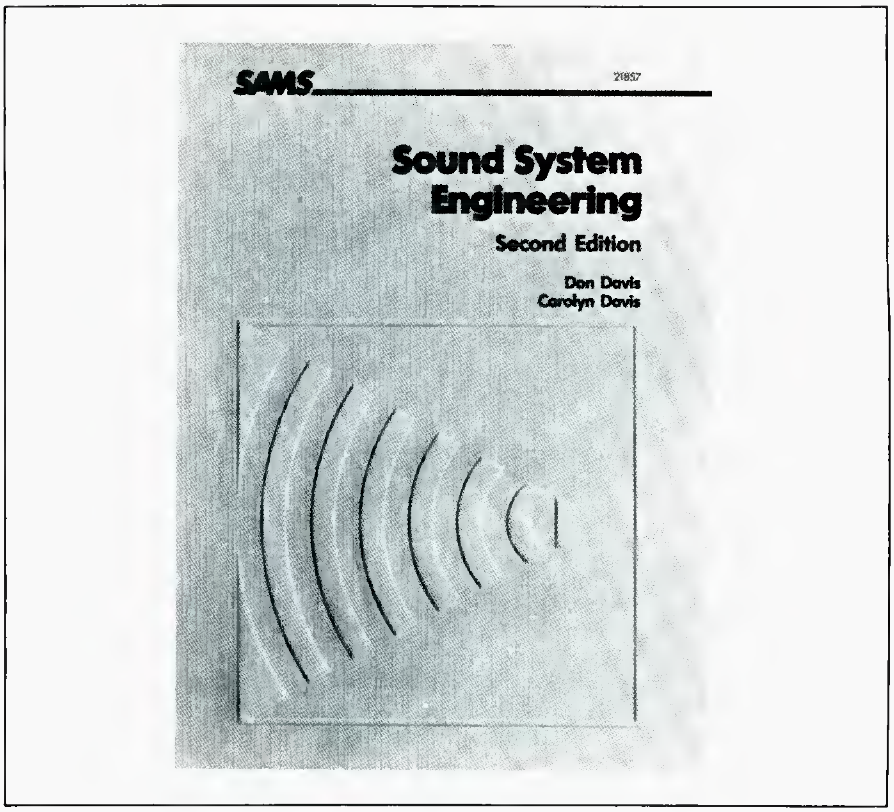
Circle 65 on Reader Service Card

SOUND SYSTEMS BOOK

Howard W. Sams recently published the second edition of Sound System Engineering by Don and Carolyn Davis. This comprehensive text carefully examines methods of accurately predicting such variables as acoustic gain, clarity of sound, and required electrical input power while plans are still on the drawing board. Discover problems that might occur in a sound system as it evolves through design, installation, equalization, operation and maintenance.

Mfr: Howard W. Sams & Company.
Price: $29.95.

Circle 66 on Reader Service Card



MIDI MIXER

The MPX-820 fully programmable MIDI-controlled audio mixer has a programmable front panel with the exception of the input gain and the headphone level control! The MPX-820 has 100 memories which can be recalled using the key pad or you can step up and down thru the memories with footswitches. You can also step up thru the memories by using the internal pulse generator to mark the spot on your multitrack recorder and when played back the MPX-820 will read these markers to step thru the memory automatically. The memories will be maintained by the MPX-820's backup battery for ten years. You can dump the memory bank to cassette via the Cassette Interface. Loading the MPX-820 allows you to store not only the multitrack master tape but also the MPX-820's mixing information. The master outputs have a switch which allows you to refrence O VU on the bargraph to O dB, -10dB or -20dB. The MIDI-out from one MPX-820 to the MIDI-in of the second unit allow you to chain up to eight MPX-820's.

Mfr: AKAI Professional
Price: $2495.95.

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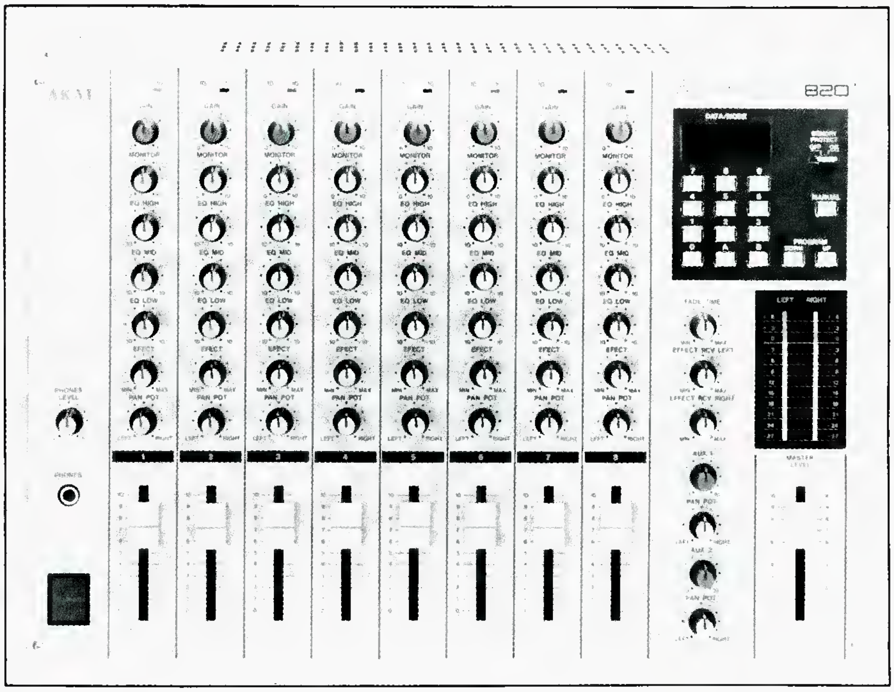

EQUIPMENT RACKS

The Pro Rack is an innovative EIA rack system that brings an expensive high-tech look to a low-cost cabinet. The Pro Rack can be expanded by replacing the side panels. Pro Rack is available in 7, 13 and 20 spaces. It is constructed of 3/4-in. multi-ply hard wood with a scratch resistant finish. Standard features include medium duty handles, steel rack rail with 18-in. rack-able depth. Accessories include casters, rear rack rail, mounting hardware and a Rhino (ATA) flight case.

Mfr: JAN-AL

Price: 7 sp.-$129.00, 13 sp.-$165.00 and 20 sp.-$215.00.

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AUDIO TEST SYSTEM

Audio Precision has introduced the "A version" of its System Audio test system. System One-A features improved stereo system measurement capability, improved acoustic measurement capability and flexible filtering. System One measures harmonic, intermodulation and quantization distortion, phase, frequency, wow and flutter, and wideband or selective amplitude and noise. A version of System One can make simultaneous amplitude measurements on two channels, permitting stereo response measurements at twice the speed. Both channels can be displayed simultaneously as analog bar graphs or the dB difference between the channels can be displayed. Additional filter capability is added by a fifth internal option socket plus provisions for external filters. The tunable band-pass filter has also been improved to ANSI Class II 1/3 Octave standards. More specific specifications upon request.

Mfr: Audio Precision.

Price: $6650.00.

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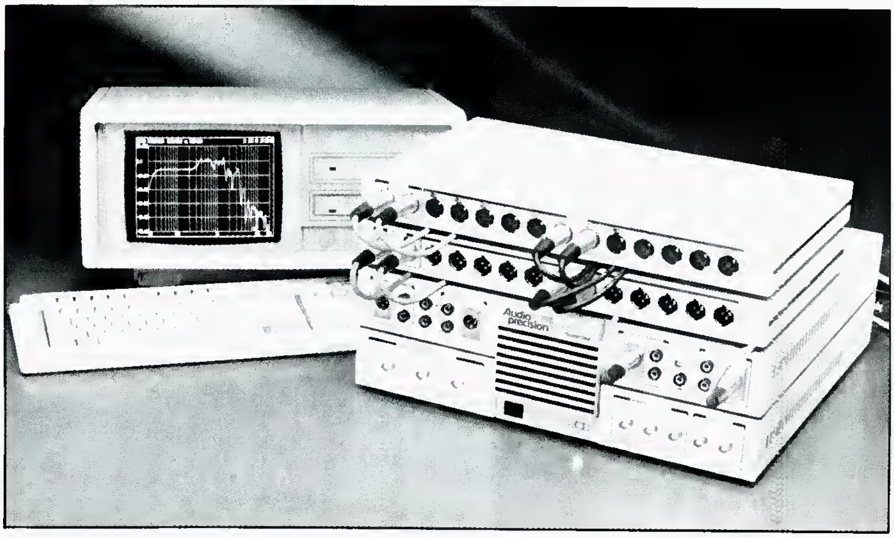

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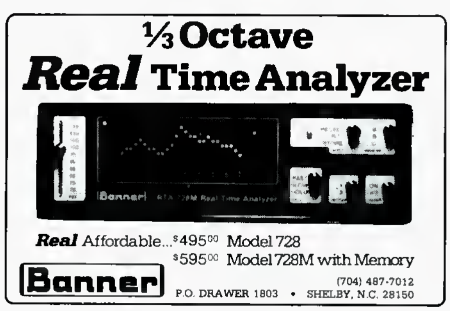

EMPLOYMENT

EXPERIENCED AUDIO TECHNICIAN for dealership. To fix current and vintage equipment. Send resume to: City Sound, Box 9830, Berkeley, CA 94708.

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People, Places...& Happenings

Soundwave Inc. of Washington, D.C. announces a major remodeling and the addition of a new Chips Davis, LEDE Design, studio and control room. The addition of a fifth control room, which will be equipped for video post and audio work, will make Soundwave one of the most comprehensive and complete audio production centers.

Live Oak Studios of Berkeley, California, is proud to announce the opening of **The Attic,** a fully-equipped, pre-production and electronic music facility. The Attic has sequencing and programming of electronic music projects. The latest electronic music gear include a Kurzweil 250 Music System with 50kHz sampling option, the Yamaha TX816 rack system with 8 DX7 modules with Opcode Voice Editor.

Bhaskar Menon, Chairman and Chief Executive, **EMI Music Worldwide,** announced the commissioning of Capitol/EMI's first Compact Disc manufacturing plant in the United States located in Jacksonville, Illinois. The Jacksonville plant has 180,000 square feet dedicated to compact disc manufacturing and cassette tape replication with an annual capacity of 7 million compact discs and 25 million cassettes. A separate 97,000 square foot distribution center is also located at the site.

Protolog Inc. is proud to announce the opening of its new recording studio in St Petersburg, Florida. The facility is geared towards SMPTE-MIDI controlled synthesis, digital sampling and special effects for records, jingles and videos.

The Andre Perry Group has recently installed an additional music recording/mixing suite, outfitted with the most recent generation Synclavier. The new and totally digital audio suite is situated between the Le Studio audio and video departments. Sound effects, voice recording and full scale music scoring can be made in superb digital quality.

Record Plant Scoring, Inc. has completed a second facility dedicated to the most sophisticated synthesizer recording. The facility is completely remodeled to house a Trident 65 mixing console, Synclavier and fairlight digital synthesizers as well as a host of processors, analog synth systems, sequencers and MIDI-related devices.

Composer **Craig Huxley's** new state-of-the-art recording studio was recently completed at **The Enterprise,** Burbank, CA. Huxley has installed the first set of Quested Control room monitors in the U.S. in its studios A&B. The cone-type speakers are currently popular in many top London recording studios and recently drew raves when demonstrated at the AES show. The Enterprise will be the only U.S. studio where Quested Monitors will be demonstrated.

Micro Plant, a new computerized MIDI-synth room, has opened its doors at the **LA Record Plant** recording complex. The cost efficient independent operation was launched by synthesizer maven **Steve Deutsch.** Micro Plant houses a TAC Scorpion 32 x 8 console, JBL and Yamaha monitors, with racks of synthesizers, sequencers and effects processors. Recorders include Fostex 16-track, Technics 2-track and digital Sony PCM 501-ES with SMPTE video lockup for scoring.

The University of Miami's School of Music has introduced a Master's degree in Music Engineering Technology, the first degree program of its kind in the United States. The new two-year curriculum emphasizes technical studies including electrical engineering, digital audio, video and psychoacoustics. A Master of Science from audio programs in music and electrical engineering are eligible to apply.

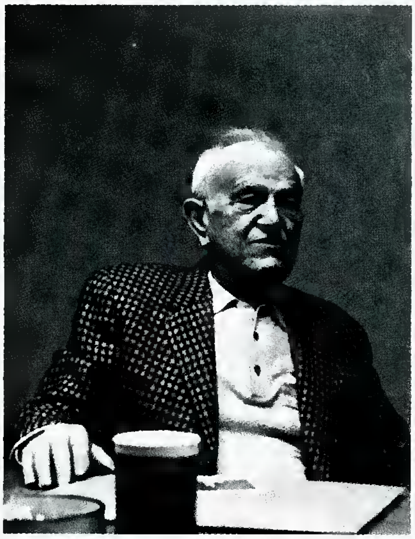
Shelley A. Herman is pleased to announce the formation of the **Shellex Company.** Its purpose is to assist small manufacturing concerns in establising a manufacturers' representative and dealer network. For assistance in marketing your product write the Shelex Company, P.O. Box 3752, Hollywood, CA 90078, or call (213) 849-4136.

A letter arrived several days ago, early February, with a terribly sad tone to it. It was from the daughter of audio pioneer Joel Tall, informing me that her father had died in December. This was a profound shock, for Joel was a personal friend for many years. At the time of his death, we were, and still are, in possession of his life story, which is in the early editing stages of transformation into a series of magazine articles, and possibly a book.

Joel would have been 82 in April 1987. In tribute to this man who invented and perfected tape editing, I would like to excerpt from some introductory remarks he made in his letter when sending the manuscript to me. Apparently, a turning point in his life took place when he was about 17 years old. "I must explain that ,according to routine, I was transferred to Boston Latin School at the end of my sixth year of grammar school because I had the highest marks in my class. Thus began a training in Latin,French, English, German, ancient history and mathematics that was the curriculum. But I was not destined to graduate and go onto Harvard, even though I had passed the entrance exams to that school.

"My home room teacher, a Mr. Stone, said the words 'dirty Jew' in a low voice, I heard him, though, and told him that I was a Jew but not dirty, and asked him to apologize for that remark. He refused to do so. Since the headmaster would not force him to do so, I walked out of Boston Latin. I have never been sorry about what I did, but any further formal education was out of the question. I just went ahead with educating myself, learning the things I was interested in learning.

"Another turning point was in 1942. I was the radio expert at a shop in Scarsdale (NY),when I heard that our armed forces needed radio experts. So, I re-



signed from my job and went down to the U.S. Navy enlistment center on lower Broadway in New York City. There I was shunted through one physical exam after another and was finally turned down because I had lost four teeth in my lower jaw. A little bone had been chipped away, the dentist said. Besides, I was 37 years old, too old, I guess.

"Of course, I needed a job right away. I phoned NBC and they told me to report for work the following Monday. So, I also phoned CBS, and they told me to report right away.

"I started with CBS at a salary of $50 per week. At CBS, I learned much abut magnetic recording (just becoming available at that point), becoming their first tape editor, then invented the Editall block, became the first expert witness in cases involving alterations in recorded sound, and wrote five articles on The Art of Tape Recording for the AES Journal, and began to write my book Techniques of Magnetic Recording, which was not finished for another seven years(partly delayed by a severe heart attack in 1956), to be finally published in 1958 by MacMillan in 1958.

"A few months after I had written the AES articles, a gentleman calling himself Yamada phoned me to ask if he and the president of the Tokyo Telecommunications Engineering Company could visit with me. I agreed, and a few evenings later Masaru Ibuka came to our apartment in the Bronx bringing along his associate. We talked every evening for six days.

"A while later I heard that Masaru Ibuka and his associate Akio Morita had begun the Sony Corporation. Right now, Masaru Ibuka is the Honorary Chairman of the Sony Corporation, and I had a letter from him about a month ago.

"A major turning point in my life came also in 1963. Ed Murrow, with whom I had worked for more than 12 years, had become very sick and had left CBS and moved to Washington D. C. to become head of the USIA. I felt lost without him, and was also becoming more annoyed by all the talk about running the CBS network by computers. I did not have faith in the ability of computers to out think the human brain. So I resigned from CBS.

"I recovered my rights to the Editall block, and began to manufacture them as the Tall Co. In time this became Joel Tall, Inc. and stayed that way until 1980 when I decided to sell the Corporation to Claude Karczmer, who operates it now as part of his XEDIT Company."

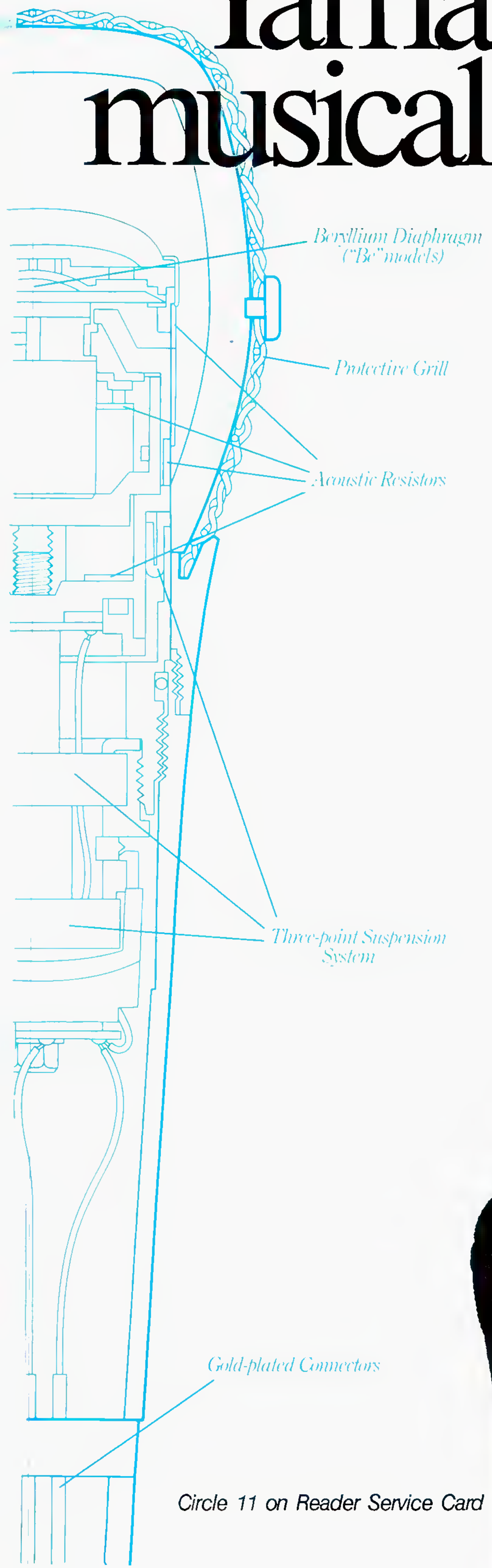
So concluded Joel Tall's July 18th, 1986 letter to me, accompanying his manuscript.

I know the entire pro-audio world mourns the passing of Joel Tall.

I certainly do. He will be missed.

Larry Zide

Yamaha's newest musical instruments.



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For complete information, write Yamaha International Corporation, Professional Audio Division, P.O. Box 6600, Buena Park, CA 90622.

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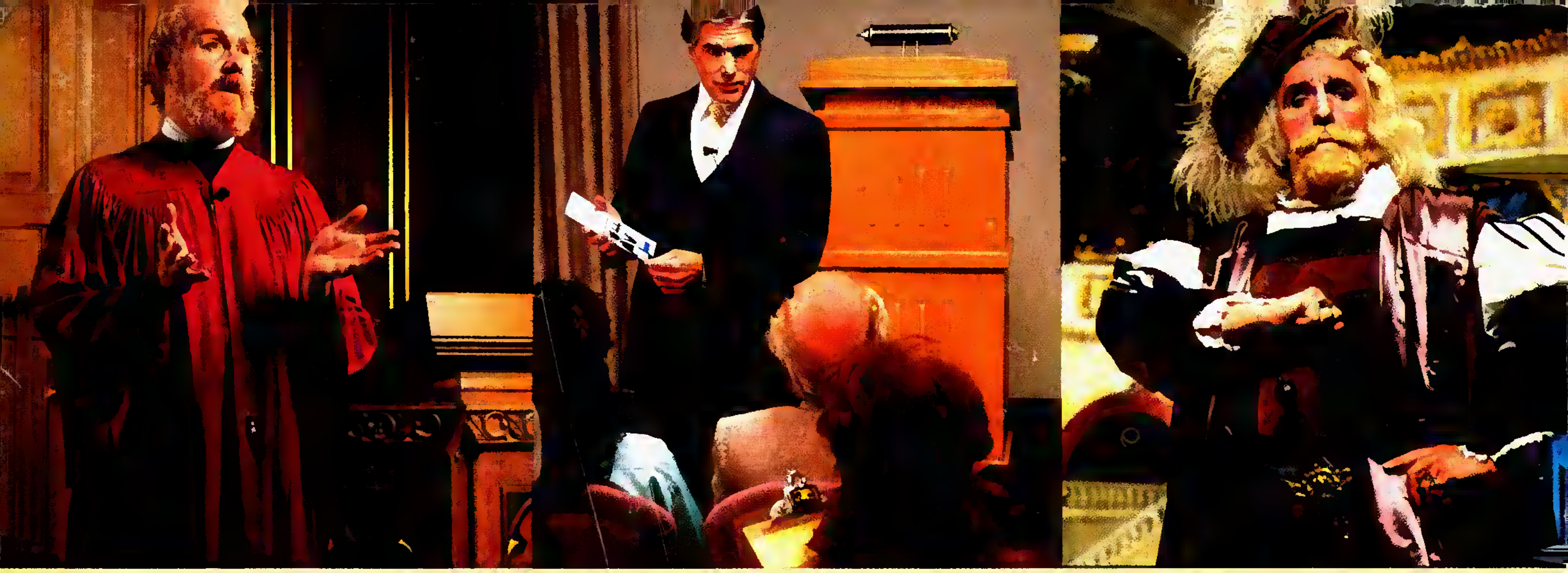

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Choose from either W25DR Diversiphase or W20R Single-Antenna Receiver with compact W10BT Transmitter. Either Shure system can be used with the specially designed WL83 Electret Condenser Lavalier or a variety of other Shure mics. For information, call 1-800-257-4873 (In Illinois 1-800-624-8522) or write Shure Brothers Inc., 222 Hartrey Ave., Evanston, IL 60202-3696. G.S.A. approved.

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