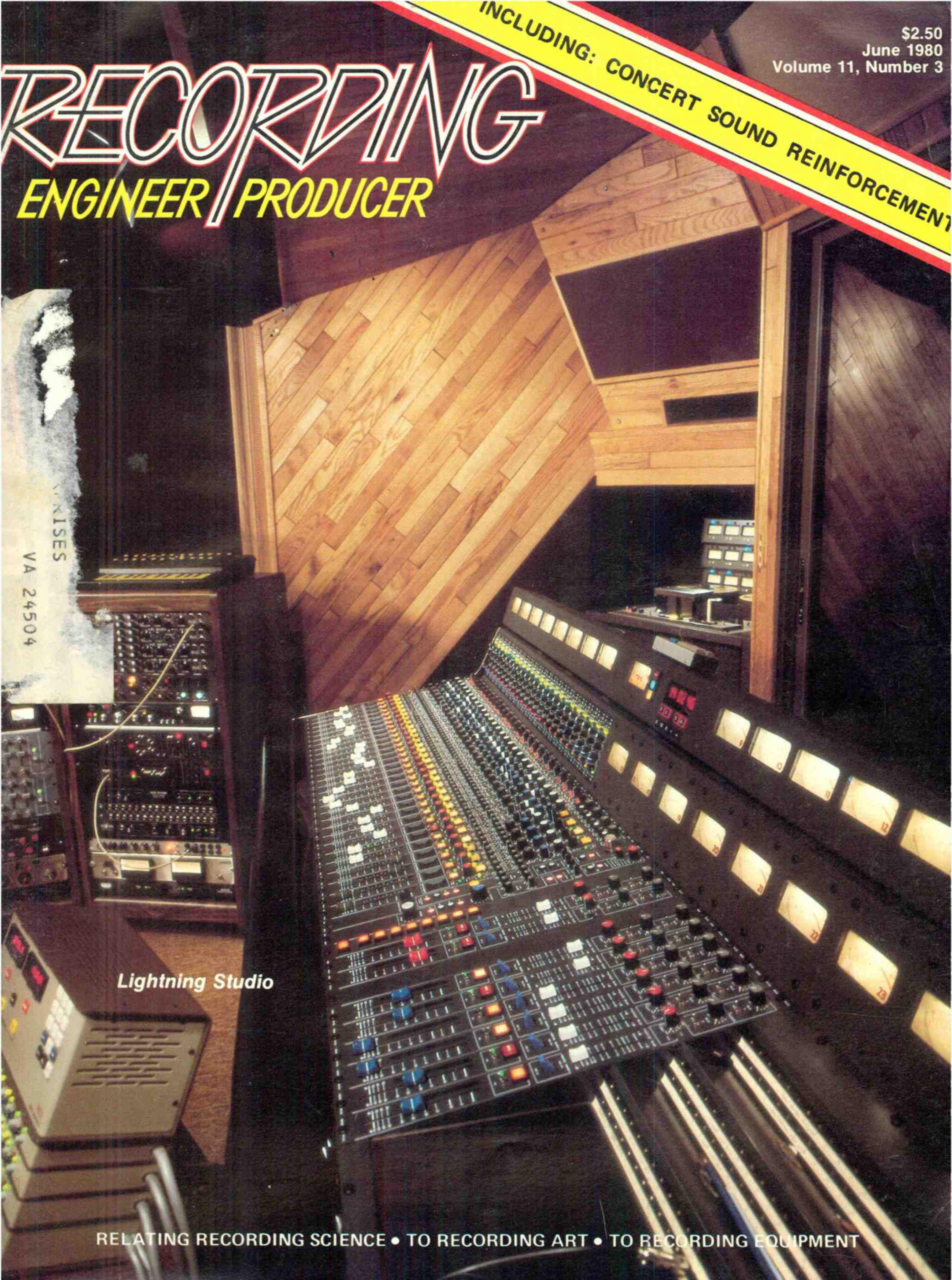
$2.50
June 1980
Volume 11, Number 3

RECORDING ENGINEER/PRODUCER

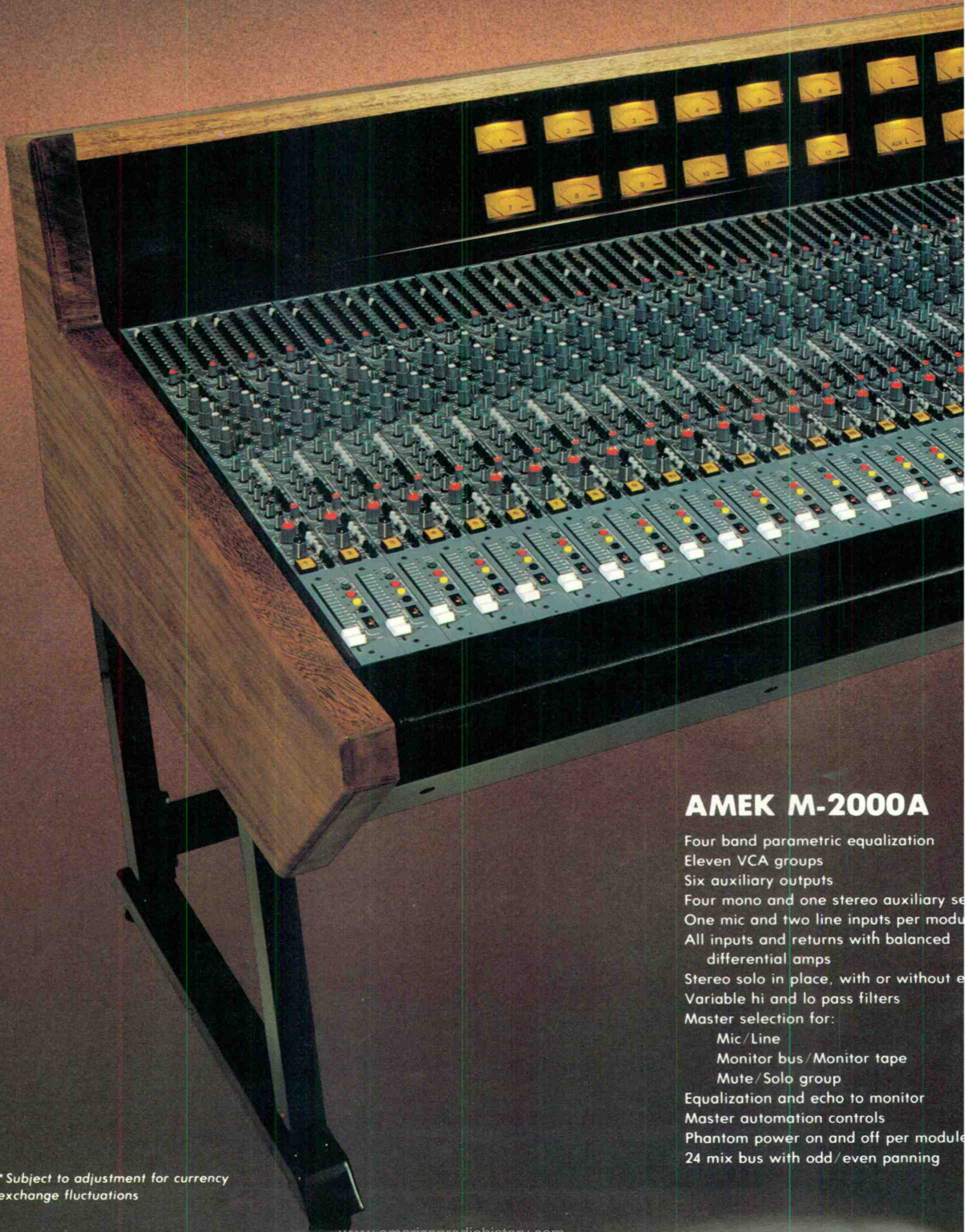
INCLUDING: CONCERT SOUND REINFORCEMENT



Lightning Studio

RELATING RECORDING SCIENCE • TO RECORDING ART • TO RECORDING EQUIPMENT

T H E P O W E R O



AMEK M-2000A

Four band parametric equalization
Eleven VCA groups
Six auxiliary outputs
Four mono and one stereo auxiliary se
One mic and two line inputs per modu
All inputs and returns with balanced
differential amps
Stereo solo in place, with or without e
Variable hi and lo pass filters
Master selection for:
Mic/Line
Monitor bus/Monitor tape
Mute/Solo group
Equalization and echo to monitor
Master automation controls
Phantom power on and off per module
24 mix bus with odd/even panning

Subject to adjustment for currency exchange fluctuations

TECHNOLOGY

patible with Auto-Pak (see
mation at right) or Tape-Pak,
's low-cost computer system
53.) Call or write for more
mation.

frame size: 36 in/24 out

M-2000 A

,000

RODUCTORY PRICE
June 1, 1980:

2,000*

for color brochure

$25,000

Auto-Pak

Auto-Pak is Amek's computer editing and mass storage system. It is the only console computer that can vocalize all commands when entered and all information as it appears on-screen. The master keyboard selectively lights the correct keys for each function group. The computer also performs as a tape locate and motion control unit, employing the SMPTE time code.

Auto-Pak's storage has three times the capacity with none of the problems inherent in floppy disks. Memory consists of 4k pages of high-speed Random Access Memory (RAM) and is used only when required by fader action. Mixes can be edited and stored on tape or in RAM.

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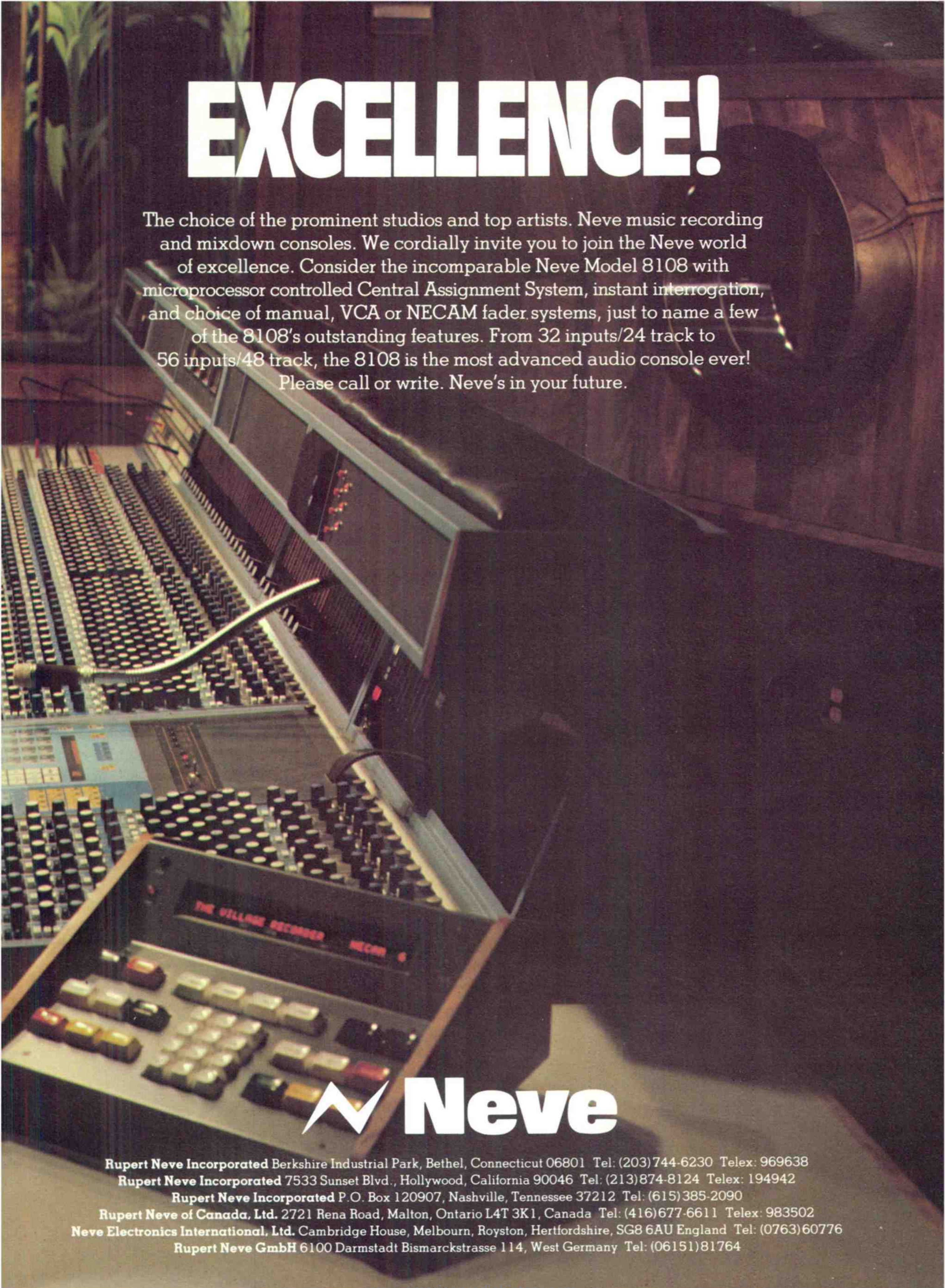
Picture shows Model 8108 console fitted NECAM Computer Mixing System at Studio B, The Village Recorder, Los Angeles. Their Studio D is equipped with a Neve Model 8078 and NECAM.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The choice of the prominent studios and top artists. Neve music recording and mixdown consoles. We cordially invite you to join the Neve world of excellence. Consider the incomparable Neve Model 8108 with microprocessor controlled Central Assignment System, instant interrogation, and choice of manual, VCA or NECAM fader systems, just to name a few of the 8108's outstanding features. From 32 inputs/24 track to 56 inputs/48 track, the 8108 is the most advanced audio console ever! Please call or write. Neve's in your future.
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June 1980
Volume 11 - Number 3

RECORDING Engineer/Producer

— the magazine to exclusively serve the **RECORDING STUDIO** and **CONCERT SOUND** markets . . . all those whose work involves the recording or producing of commercially marketable audio.

— the magazine produced to relate the audio **ART** . . . to the audio **SCIENCE** . . . to audio **EQUIPMENT.**

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- Contents -

the cover —

Lightning Studio is the culmination of a two-year project to provide something a little different.

It operates as a private live-in facility, available by invitation only. Complete guest quarters include such human necessities as private refrigerator, bath, steam bath, swimming pool, jacuzzi, etc. The guest quarters are linked to the studio with stereo audio and color video links.

The recording facilities are quite different in that they can be operated in a conventional manner, with engineer and group — or they can be completely managed by one person. In order to optimize the 'one-man' function, several special considerations had to be realized.

The control room is actually constructed backwards, with the console facing away from the studio, looking into a glass wall between the front monitors facing the woods outside. Above this window are color monitors (behind glass to attenuate HF audio garbage from monitors) with remote low light cameras in the studio.

In addition, between the color monitors is a parabolic mirror, allowing the engineer to view the main studio directly, through a second glass window behind him.

This backwards arrangement has several advantages; monitors are at ear level, and you get to look at something a lot prettier than a rock 'n' roller on his 6th take. It allows a single person to alternate between the studio and control room, without having to walk around to the other side of the console, and it allows one to read metering on the console from studio.

Cover Photography:
Stephen St. Croix

R-e/p's new logotype:
Designed and executed by
artist and graphic designer
Eric Wrobble

Fantasy

PRESENTS

studio d

DESIGNED BY

KENT DUNCAN, TOM HIDLEY & SIERRA AUDIO

A STUDIO MAGNIFICENT IN EVERY RESPECT

Fantasy's Studio D represents the zenith of today's recording technology. "D" offers an unprecedented combination of electronic, acoustic, and aesthetic features designed to meet virtually every conceivable recording need.

A fully computerized and automated 46-track studio, "D" is equipped with Studer mix machines and two A800 24-track machines which are interlocked by the Studer TLS 2000 (SMPTE Time Code Synchronizer), giving you 46 usable tracks; Sierra/Hidley Acoustics and Sierra SM III studio monitors; Crown Monitor Amplifiers; and a 56 x 48 console by Neve with NECAM computer-assisted mixing.

This mixing system offers amazing flexibility in arriving at the final mix. It gives the ability not only to store as many mixes as desired, but to combine (merge) sections of any number of mixes into the "final mix." Incredible!

The unusually spacious control room affords an unobstructed view of both the 30 x 50 main room and the adjoining 18 x 30 string room. In addition, the main studio area contains three isolation booths and a drum booth large enough for two drum sets. Dramatic modification of color and mood can be achieved by an extraordinary lighting system, while the design of the room permits total acoustic control. Sliding mirror wall panels, ceiling louvers remote-controlled from the console, Sierra Audio's use of marble to hardwood to carpeting all add to the variety of acoustics available. Hearing is believing!

Overall, Studio D provides the ultimate in a comfortable, visually pleasing, and technically complete recording environment. "D" is one part of the Fantasy Studios complex. This unique Northern California facility offers a full line of engineering services, including a superb new Sierra disc mastering room, a 16-track studio, and two 24-track studios.

Studio Dimensions
Studio A (24-track) — 30 x 50
Studio B (16-track) — 21 x 26
Studio C (24-track) — 24 x 37
Studio D (46-track) — 30 x 50 main room
18 x 30 string room

Equipped as follows:

Consoles
1 Neve 56 in 48 out
2 Custom DeMidio — 24 in 24 out
1 Quad 8 — 20 in 16 out

Monitor Amps
Crown 300's, 150's, and 75's
McIntosh 2100s

Monitor Speakers
Sierra SM III — Cetec Gauss and TAD
UREI 813
Altec 604E w/Masterlab crossover
JBL 4311, MDM, Auratone

Echo
EMT 250
EMT 140S
Ecoplate
and live chambers

Outboard Equipment
Eventide Harmonizer
Eventide DDL
Lexicon Prime Time
ADR Complex limiter
ADR Vocal Stresser
MXR Phaser
ADA Flanger
EMT PDM Compressor
Scamp Multi-Rack
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and more!

Dolby Noise Reduction — included
DBX Noise Reduction — extra

fantasy studios

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for additional information circle no. 1

views letters news

- M/S MICROPHONE TECHNIQUE -
from: **Doug Pomeroy**
Brooklyn, NY

Herewith a few thoughts prompted by Paul Rainey's article *(R-e/p, April 1980)* on his use of the M/S mike technique. I, too, have used a mixing desk in place of the special transformers and T-pads traditionally used for M/S matrixing. In the studio, each mike can feed two mixing busses, by double-punching, with the phase of the appropriate one reversed before feeding the stereo mixdown section. This avoids any problems which might result from use of a 'Y' cable, such as improper impedance seen by the mike outputs.

Also, while I agree it is always excellent practice to have a scope across the stereo output, a more accurate way to set the proper balance between the two anti-phase (S) signals (busses 3 and 4), is to solo them in mono summation: they should always cancel completely. If they don't, the center image created when they are added to the M signal will appear off to one side, and the mono (L + R) will be less than ideal. In the case of a mixer which has VCAs and grouping capability, the exact balance between the anti-phase signals, once set, can be easily maintained as the desired amount is added to the mix; alternatively, a high quality stereo fader could be used for this function. The point is that cancellation is easier to hear than to see!

Incidentally, as a varaition of the M/S technique, a figure-of-eight pattern (or an intermediate one such as hypercardioid) may be substituted for the forward-facing cardioid, thus supplying to the M component more room ambience and audience pick up.

- DRUM TUNING -
from: **Michael Rettinger**
Consultant on Acoustics
Encino, CA

The equation for the first resonance frequency of a drum head on page 128 of your April 1980 issue is incorrect in value and incorrect in the manner in which it is written.

It is incorrect in value because the density of the membrane (sigma) is stated in kilograms per meter — an obvious impossibility for a circular sheet. Correct is to state it in kilograms per square meter.

It is incorrectly written because the radical appears in the denominator, whereas it should occur in the numerator.

Correctly written, the equation should appear as:

$$f_1 = 0.0766\ (T/\sigma)^{1/2}/D$$
$$(\sigma = \text{sigma})$$

where T is the tension of the membrane in N/meter and D is the drum head diameter in meters.

from: **Bob Hodas**
Record Plant Studios
Sausalito, CA

With respect to the description of Figure 8 in my article, a line was left out which made it quite confusing. The description on page 136, column 2, paragraph 2, should have read:

". . . , it will have a maximum effect on the second partial with little effect on the fundamental (Figure 8B). Figure 8C shows a minimal effect on the fundamental with no effect on the second partial. By changing . . . "

Please print this correction as several people have already asked me about this.

- INDEPENDENT PRODUCTION -
from: **Gregory McKay, President**
Gem Productions, Inc.
Beverly Hills, CA

While reading your April 1980 issue, I noticed an article by Dave Pell, which gave me great concern.

I am the head of a record production company, and also two affiliated publishing companies, and while I am not a practising attorney, I am well versed in some of the legal ramifications regarding certain points in the article.

With regard to Mr. Pell's "Letter of Agreement:"

#3. Musician's services cannot be provided "on spec." They must be paid according to the A.F. of M. scale for demo's.

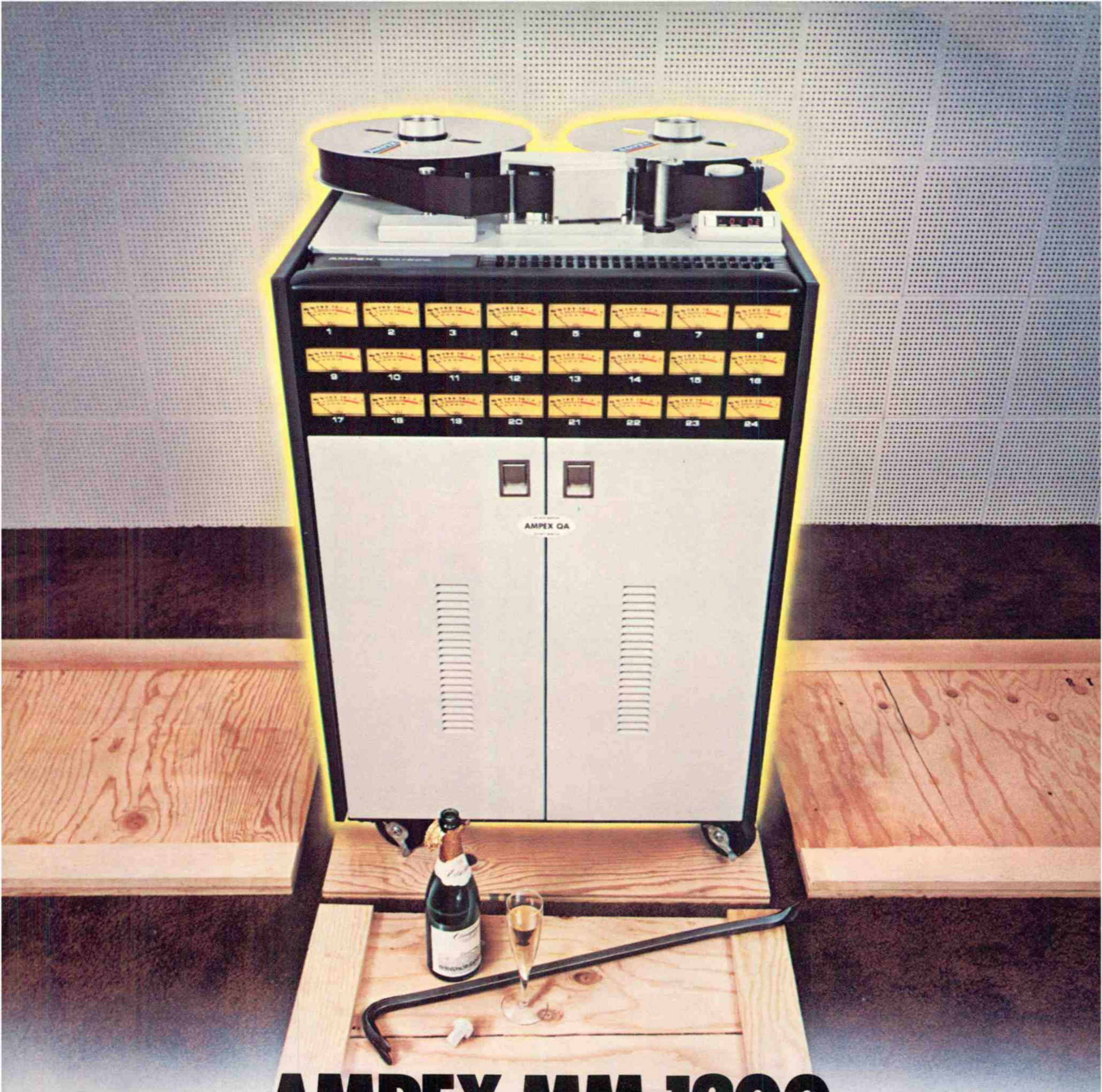
#6. Administration fee of 20% is exorbitant.

#7. How can he possibly be entitled to a "Management Fee" when he is already the "Production Company." This is considered "double-billing" and is improper. Further, 25% is considered exorbitant by everyone today. This management fee sounds like an "agent's fee" for getting the deal (in disguise), and since he is unlicensed as an agent, is illegal.

In the "Rules" chart, he states: "Personal Manager has right to assign agency to handle bookings." This again is untrue and illegal. According to the Labor Commissioner of the State of California, any act of solicitation of employment by an unlicensed agent is illegal, and will result in voiding of all agreements (in addition to refunding of all fees paid, and cancellation of all debt).

I would suggest some corrections and retractions be published in your magazine,

— continued overleaf . . .



AMPEX MM-1200. WHERE GREAT STUDIOS START.

Select your multitrack recorder as carefully as the other facets of your studio. Select the Ampex MM-1200. Because it's the one multitrack recorder that can go through every change your studio goes through on its way to greatness. And still be as economical and easy to operate as the first great day you got it.

Growth without growing pains. With the MM-1200, you'll seldom be faced with a situation you can't solve. Because the MM-1200 comes prewired to make upgrading from 8 to 16, or 16 to 24-track operation simple and swift. And if adding channels won't solve your problem, the MM-1200's versatility will. Mastering, live sound reinforcement, double system sound, video sweetening or film and TV production/post production are all jobs that the MM-1200 can handle. Built-in single point search-to-cue, elevated record level capability, 16" reel capacity and fast start times also help you grow.

Performance you can depend on. The MM-1200 has proven itself under some of the most adverse conditions. The massive, stable top plate comes aligned and stays aligned . . . through repeated sessions in the comforts of the studio, or on remote locations.

Ampex keeps your options open. The list of optional accessories for the MM-1200 is the longest in the business. You can add multi-point search-to-cue and store 20 cue locations. This time-saving tape handling accessory provides tape time readout, cue point readout, "on-the-fly" cueing and more. Other accessories include the PURC™ Record Insert Controller, Search-To-Cue Remote Control, and MSQ-100 Synchronizer for jobs that require more than 24 tracks. Contact your Ampex sales representative for complete details.

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for additional information circle no. 3

especially since Mr. Pell claims to be an expert (and a professor — to my amazement!). Otherwise, an ambitious new independent producer who forges ahead based on your advice may find himself in serious trouble, and seek remedies against Mr. Pell and your magazine.

reply from:
Dave Pell
Hollywood, CA

As an active producer, publisher, and teacher, I'm constantly barraged with questions regarding music publishing and administration. For the past several years, I've been working with Larry Shayne Enterprises, a very successful independent publishing concern. By independent, I'm referring to any publisher unaffiliated with a major corporation or record manufacturer. As a fixture in the publishing world for more than 30 years, Larry administrates copyrights for a number of firms, and individuals such as Marvin Hamlish, Henry Mancini, Livingston and Evans, and Pat Williams.

Administration of a firm entails a lot more than just doing bookkeeping on royalties that are received. Copyrights licensing, and day-to-day procedures make it generally advisable to have a professional publisher administrate the copyright.

Administration fees are negotiable, and consequently, vary with each situation. While fees range anywhere from 10 to 25%, the average charge is 15%. Even major publishing houses have this divergence in the fees they charge their clients. There are numerous publishers who administrate copyrights for record companies, for fees as high as 25%.

There are occassionally situations where a split publishing company is set up for the producer and artist/composer where no administration fee is charged. In this instance, administration rights acquired from the artist, are used by the producer as a negotiating tool to set deals and get advances.

Artist management fees are also negotiable, while these fees range between 10 - 25%, the average is 15%. A manager's role is that of guidance and counsel. Most artist managers handle a stable of only a few clients and are interested in the long range aspects of career development. Advice regarding image, style, artistic direction, photographs, and personal services, fall within the manager's purview.

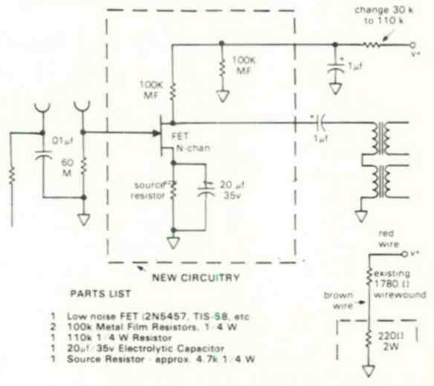
A talent agent seeks employment for his client. The fee is 10%, the exception being for one-nighters. There are many different kinds of agents. Some specialize in a certain style of music, some book strings of lounges, and some are packagers of talent. Agents who represent artists usually have a large stable to fill their varied needs.

There are many guilds and associations that have meetings featuring informative speakers. There are droves of books covering the many show business fields. Knowledge of the industry, it's component parts and how it all works, is available, and will prove helpful to those who are serious about a career in the music business.

- *U-47 RETROFIT* -

from: **Ethan Winner**
The Recording Center, Inc.
Norwalk, CT

Congratulations on the excellent series of articles about condenser mikes. The one on the U47 was of particular interest to me as the recent failure of the VF-14 tube in my unit prompted me to look into an alternate preamplifier.



Trying to find a replacement tube seems extremely difficult because there is little available data (to my knowledge) on the

— continued on page 16 . . .

THE ROLE OF THE RATE-CARD RELATIVE TO THE COST OF BUILDING A RECORDING STUDIO

"I would lke to build a recording studio. Can you supply me with the necessary cost and planning information I'll need?

By Jerry Milam, Milam Audio, Pekin, Illinois

This question is asked of most pro and semi-pro audio dealers dozens of times each month. Though the question is common, the answer is becoming more and more complex. Formulating cost and planning guidelines requires a study of "the current studio" and its market position.

Be they large or small, pro or semi-pro, multi or single room, the operations of most studio facilities share many common problems: fixed costs, the pressures of competition, the establishment of a position in the marketplace, the demands of advancing technology, etc. Without discrediting the importance of creativity in the studio, if the business end is neglected and failing, production output and originality will become extremely difficult under the strain of financial pressures. The successful studio operator must have a thorough understanding of the business aspects of the audio recording industry.

FIXED RATE CARDS

A current and significant business decision involves the need and use of studio rate cards. Many say that even though they would like to use them, the loss of business to other studios undercutting their prices makes fixed rate cards detrimental. If this, in fact, is the case, one must ask if the card prices being charged are enough to allow profit on investment in the face of stiff market competition. Necessary income rates must be based on all expenses, such as fixed overhead, maintenance and service, technological upgrading, studio image, potential loss, etc. Once established, these rates should not be compromised, even if it means the loss of some bookings.

SALES

Since running a studio is centered heavily around selling, basic sales rules should not be forsaken. The price charged must be profitable and able to fulfill all terms of the agreement. There is an old cliche in sales that states profoundly, "There is no such thing as a sale without obligation." Simply put, no matter how secure a deal may look or sound, there is always obligation of some kind to fulfill after the sale is completed.

An example of what could occur when business is slow is the studio down the street undercutting your price. A group comes in to your operation and wants to record, but makes it clear that it can do it elsewhere for less money. Pressured by that factor, even though you may have a more professional facility to offer, and by the thought that any business is better than no business at all, you make the sale at the lower price. No one will argue the fact that you needed the income and therefore sold at a lower rate than required to make a profit. Things were tight and so it goes, but what has happened is, in fact, a compromise of your basic values. In addition, the group that you struck this bargain with will come back in a few weeks or months, expecting an even better price. They now consider themselves steady clients, forcing you into compromising even further.

Word of mouth is often as significant as media advertising and it won't take long before you'll be selling time to everyone at prices lower than the basic rates needed to meet expenses. This will cause the studio to develop an image and sales pattern that is very hard to break out of once established. Rate cards can be a very valuable and necessary tool when quoting profitable rates to perspective clients. If the rates cannot be sold for the minimum number of hours required per month to profitably survive, cutting the price in special arrangements will only slow down the eventual death of the operation if there is no support from outside the studio.

— continued on page 16 . . .

I am extremely interested in your new Series 400 console.

Please send me the colour brochure which gives full details.

Name

Address

Post to:
Soundcraft Electronics Ltd., 5-8 Great Sutton St.
London EC1V 0BX, England
Telephone: 01-251 3631. Telex: 21198

or
Soundcraft Inc., PO Box 2023
Kalamazoo, Michigan 49003, USA
Telephone: (616) 382 6300. Telex: 22-4408 Soundcraft KMZ

Catch the new Sound with the French Infernal Machine

DHM 89 B2 stereo

The D.H.M. 8 9 B2 is a stereo audio computer, which allows :

Dual digital delay :
In quasi-stereo mode, two independante delays operate on the same sound from 1 ms to the maximum value.
In true stereo mode the two channels are completely separate.
The maximum value of delays depends upon selected bandwidth : 1200 ms for 5 KHz, 600 ms for 10 KHz, 300 ms for 20 KHz.
In option additive memories can upgrade delay up to 5000 ms» The D.H.M. 8 9 B2 offers an unique feature which allows a continuous variation of delay without doppler effect nor switching noises. So that it is possible to change delay during operation without audible effects.

Dual echo :
By association of dual delay and feedback.

Dual pitch shifting :
From – 2 to + 1 octave. Sophisticated micro-computer operates phase coincidence of joining points and eliminates «glitches». The serial delay is adjustable.

Dual automatic arpeggio :
is a pitch-increasing or decreasing echo.

Dual reversed sound :
Electronic equivalent of a magnetic tape running reversed. Reversed sound itself can be pitch-shifted.

Dual memory latch-mode :
memorised sound can repeat indefinitly. The two borders of repeating sound are continuously selectable, as wall as reading sense and pitch ratio.

Principal technical features :

- 16 bits flying comma A/D converter allowing a 95 dB dynamic range, without use of an analog compressor/expander technique.
- **Distorsion :** 0,1 % for delay mode, 0,2 % or pitch shifting mode.
- **Memory capacity :** 210.000 bits.
- **Sample rate :** 52,91 KHz (for 20 KHz bandwidth).
- **3 dB bandwidth :** Selectable, 5, 10 or 20 KHz.
- **Printed boards :** Easily removeable for maintenance.

Publison Audio Professional

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Put any sort of Sound in memory and Tame it!

KB 2000

KB 2000 is an external programmation unit for the audio computer DHM 8 9 B2. It includes a three octaves piano keyboard, and a control panel for setting the following functions :

Three voice chorus :
By adding to original sound two pitch shifted voices. The resulting chord is controlled by the keyfingers. A serial delay can be added, which is continuously adjustable.

Reverse synchronisation :
The reverse mode of DHM alone affects the original tempo in a random way. The «Reverse synchro» mode synchronises this effect with attacks of the sound, so that reversed sound has the same tempo as original. The added delay is adjustable.

Biphonic memory synthesizer :
Any existing sound, supplied by a tape recorded or by a microphone, is memorised in DHM by simply depressing the «memory latch» button. The length of memorised sound is 1,2 s for DHM 8 9, 5000 ms for option. Then the KB 2000 synchronises memory reading with attacks of the notes. «Attack point» sets the part of the sound at which each note begins, «End point» and «Return point» select the part of memorised sound which is repeating continuously when the note is sustaining. This sustain can be very clean, for example on a bell resonance or a human voice vowel, by means of the internal phase tracking computer of DHM. «Speed» sets the reading speed of memorised sound.

Two envelope generators :
Drive two VCA to control the envelope of the notes. They are adjustable in attack time, hold time and release time.

Dual evolving vibrato :
Three basic parameters : frequency, sharpness and depth are modulated function to time by a form generator initialised by attacks of the notes.

Glissando time :
From a note to the following note is adjustable.

Sustain mode :
Once a keyfinger is depressed, the note starts and continues, even when the key is released. The real duration of note depends upon settings of envelope generator.

Push-play mode :
The notes are heard only if the corresponding keyfingers are continuously depressed.

Trimmer :
Adjusts the fine tune of the whole keyboard. It is a general adjustment, the notes are always in a good ratio together by means of a digital generation.

Création

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- *U-47 RETROFIT* -

characteristics of the original tube and also, of the few tubes being manufactured today, none are intended to operate at the unusually low plate voltage within the U47. And besides, tubes are such "messy" things. So it was an easy decision to go solid state. Though some may disagree, it is my contention that the sound is governed mostly by the capsule which, by the way, is *not* the same as the one supplied with newer FET versions of the U47.

The circuit shown will provide good noise and overload performance provided the source resistor is optimized for the particular FET being used. Start with 4.7 K and adjust up or down for approximately 0.3 mA of drain current. Use a low noise N-channel FET such as the 2N5457 or TIS-58. In fact, it is advisable to purchase a half-dozen or so FETs and try each one selecting the one that gives the lowest noise. One very important note: with the tube removed from the circuit, the power supply output will rise to over 300 volts so a load must be installed where the tube's filament had been. Use a 220-ohm, 2-watt resistor between the existing wirewound dropping resistor nested in the curve of the case and ground. Since the wirewound resistor is still active, the mike case will warm up as it always did so you can make believe the tube is still there. (This is undoubtedly why the microphone has retained that "warm" sound that U47s have always been known for.)

- *DISK CUTTING* -

from: **Steven Cavanaugh**
WUOM
Ann Arbor, MI

I am concerned about side effects of the new "groove-nestling" method of disc cutting.

This is a technique for conserving groove space in which the groove computer compares the waveforms of consecutive grooves and advances the cutting stylus by only the required amount, once each revolution.

The problem with this method is that instead of advancing the cutting stylus slowly and continuously it is advanced in a series of rapid jerk-like motions. Under certain signal conditions this can be a substantial jerk, which causes observable tonearm instability and thus generates unwanted sub-audible signals, similar to record warp signals, with practical arm-cartridge combinations.

I recognize the advantages of this cutting system in terms of increasing the permissible level and/or playing time, and do not suggest that it never be used. However, I would not like to see it become the universal standard system in state-of-

— *continued overleaf* . . .

THE ROLE OF THE RATE-CARD RELATIVE TO THE COST OF BUILDING A RECORDING STUDIO

SPECIAL RATE SITUATIONS

There are always ways to offer lower priced time without negative effects on the normal rate card pricing. One is to offer demo rate hours with the time of day determined by the studio operator and based on cash only sales. If the tapes produced during these hours are eventually used as a master product, the studio could be reimbursed the difference between the two rates when that occurs.

The high to low average billing rate for 16- and 24-track time three years ago was \$75.00 per hour. The average rate today is still \$75.00 per hour or less. The cost of building, equipping, maintaining, and upgrading a present day studio has increased more than 45%, and in many areas, more than 60%. Why is this happening? There are a lot of opinions. Some say that we are reaching a point of too much supply for the demand. Semi-pro is taking more and more pro business; groups and clients do not have the budgets, cost of doing business is out of control, etc. The list goes on and on, but when you look at the successful professional operator, his rates have not only advanced with the times, he is never ending in his quest to do bigger and better things. More importantly, he has made a total commitment to the industry, both in time and money. He's keenly aware that he must continually incorporate fresh new ideas to produce positive results and rise up from the mainstream of rate cutting competition.

This is not to imply that everyone must have a rich relative and ten years of background in the industry before considering getting into it. Though it certainly won't get in the way if you do have. What must be realized is that in order to compete, one must really know where they will fit in, how much they can charge in theory and reality, and be willing to stick to a game plan. Each marketplace differs in make-up: the cost to get into business, the number of tracks necessary to compete, how high the rates are based, etc.

SINGLE vs. MULTI STUDIO OPERATIONS

A facility with two or more studios definitely holds the winning hand when it comes to marketplace competition. The more rate levels and hours that can be billed, the better the chance to succeed. While costing more to build and equip in the beginning, the long term return will far exceed the single studio's ability to earn income. It also relieves the pressure of head-on competition with rate cutting facilities, loss of new business because of lock-out booking time of a single studio, etc. An ideal arrangement of two or more studios would be two 24-track rooms; one very well equipped, and the other with little flair. Also, a smaller production facility containing 1 to 8 track, large enough to produce a six piece group. If a client desires effects gear, mikes, etc., that the lower priced rooms don't have, it can be rented to the client from the better equipped rooms.

CASH FLOW & CREDIT

When considering building a new studio or studios, it must be understood that in most cases a positive return on the investment may take as long as six months to a year. Being somewhat different from other types of business, the studio must first prove itself with sound as well as service, before the ball can get rolling. Accounts receivable from most recording dates lag from one to four months behind the work being completed. In some cases, as with major record labels and advertising companies, it has been known to take in excess of six months. This can create a severe cash flow hardship even with the largest of studio operations.

Business reputation is centered around clean credit, and even though receivables will be slow, studio bills must be paid on time. To do this, the business has to have reserve financial muscle to see it through. It doesn't take many slow pay accounts to mess up cash flow, and if your credit lines begin to falter, the next sound you may hear is the huffing and puffing of the wolf at your door. Doing business with other people's money, such as paying invoices after they are past due, and stretching credit lines to the limit, has been a heavily used, unjust method of getting by for many in the past. Business climates are rapidly changing, however, and more financial stability with cleaner credit policies will be an absolute must for the studio operator and businessman of the future.

THE BOTTOM LINE

In answering the original request, plan your studio with open eyes and realism. Bankers just do not cotton to "Pie In The Sky" in any form. They require hard facts and solid credit data. They also, place little value on recording equipment for collateral. Normally 40% of its list price is the average credit allowance with a bank. This is a tremendous joke, but many bankers do not care to understand the studio business. Leasehold improvement cost on building space not owned by you can also be difficult to finance. Banks consider it a high risk investment that is impossible to recapture if the studio should go under. If you can buy or build your own building, money is more available. In many cases, the owners of some buildings will do part or all of the remodeling, and figure it into a long term secured lease arrangement.

Do a very thorough study of "your" marketplace. See where and how you fit in. No matter what size project you do, have ample funding to start it off on the right foot. Working capital reserves should be equal to six months of the fixed overhead cost of operation, if available. Once the total cost of the project is determined, add 25% to that figure. This is the average overrun cost of most finished facilities.

In closing, and most important, if you do have a *very* rich uncle that thinks you're the greatest — *call me without delay!* □ □ □

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the-art cutting rooms, and wish to make engineers and producers aware of this drawback in present groove-nestling cutting systems.

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reply from:
Jack E. Hunt
Senior Engineer
JVC Cutting Center, Inc.
Hollywood, CA

It is felt that a reply to Mr. Cavanaugh's letter about "groove nestling" in disc cutting is in order. Mr. Cavanaugh's description of the system is, I am sorry to say, quite inaccurate. He states that the stylus is advanced only by the required amount once each revolution. That is not a description of the "groove nestling" technique, but is a description of "fixed pitch." Even the standard "off-the-shelf" version of the Neumann VMS-70 lathe changes the number of lines per inch, or "pitch," *four times* each revolution!

Now let's talk about these improved pitch and depth control systems. Neumann has recently introduced a new mastering lathe, the VMS-80. This lathe changes pitch and depth *sixteen times* each revolution, greatly improving the use of the space on the disc. *This* is an example of "groove nestling." There are also other systems on the market to bring older lathes up to this level of performance. One is by a company called Sontec and is quite elaborate. Another system, the one that we choose to use, is the Zuma disc computer. This system was created by a disc mastering engineer who was *aware* of the problems of too-rapid a change in pitch and how they would affect disc playback. This system also changes pitch and depth sixteen times each revolution. While we have been able to observe these changes in pitch on an oscilloscope when cutting at half-speed, they are at such a low level that they do not cause any problems when the disc is played back.

What this all boils down to is a truism: *Any* new system or technique can cause problems if improperly handled, or, "it's only as good as the guy in the saddle." These systems are capable of spectacular results and *do* have a definite place in state-of-the-art disc mastering studios.

- TV AUDIO -

from: **S. A. Cisler**
General Manager
Sight and Sound
Louisville, KY

It seems to me high time for an article investigating the variation in level being used today in masking the main voices of actors or singers on records and especially TV sound.

Anyone who viewed the CBS-TV movie, "Scarecrow," with Gene Hackman and Al Pacino, had to endure one of the outstanding horribles of this type. Although Hackman spoke his mumbles with a chewed cigar in his mouth most of the time, the skittering levels of Al Pacino made the complete soundtrack a travesty. Hardly a sentence could be understood. Possibly no mike or mixer could have saved this butchery of an interesting story, but how did such an audio botch get on a national TV network?

As an old broadcast engineer myself, and an avid collector of big band records, I have more than a passing experience. Today most sound on records and TV lacks *presence* of the major voice. He or she is only a murmur in the uproar. Compared to the diction and articulation of an early Sinatra backed with the Dorsey band, or even an earlier Rudy Vallee, the present crop of singers, apart from their miserable sonic garbage they call "music," is a poor reflection on the new equipment and the fists controlling levels.

Mixers should be forced to hear their product through the bulk of reproducer types we peasants have at home. But even to pipe many of today's records or TV sound into a good home system is to lose the sense a good voice presence can bring to the average listener.

With all your mikes and equalizers try a little boosting of the main voice.

CORRECTION

In the December 1979 issue of R-e/p, (Volume 10, Number 6), *the distributor of Harrison consoles was mistakenly identified in the article about Automated Film Consoles, by Levi Storm. The Harrison distributor for all post-production, and recording consoles, for the Western Rockies to the Pacific is* Electro-Media Systems, Inc., 8230 Beverly Boulevard, Suite 27, Los Angeles, CA 90048. Telephone: (213) 653-4931.

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for additional information circle no. 7

What The
AUDIO/VIDEO fusion
Can Mean To The
RECORDING STUDIO INDUSTRY

— part 3 in the series —

. . . for those brave enough to try —
PRODUCTION VIDEO EQUIPMENT PACKAGES ——— by Steve Barnett

In previous discussion in these pages about possible expansion into video by the audio studio, various possibilities were explored with regard to the marketplace. A number of factors have to be taken into consideration, not the least of which, is the city in which the studio is located and other video competition in the area, possible clientele and the direction the studio may wish to take considering the rapid changes in audio technology.

Keeping these qualifiers very heavily in mind, let us consider some options.

A studio may wish to link up with an established video house by providing it with production sound recording and/or audio post-production services. Obviously, for the former service it is necessary to be a mobile audio operation. In either case, the cost of

— *continued overleaf . . .*

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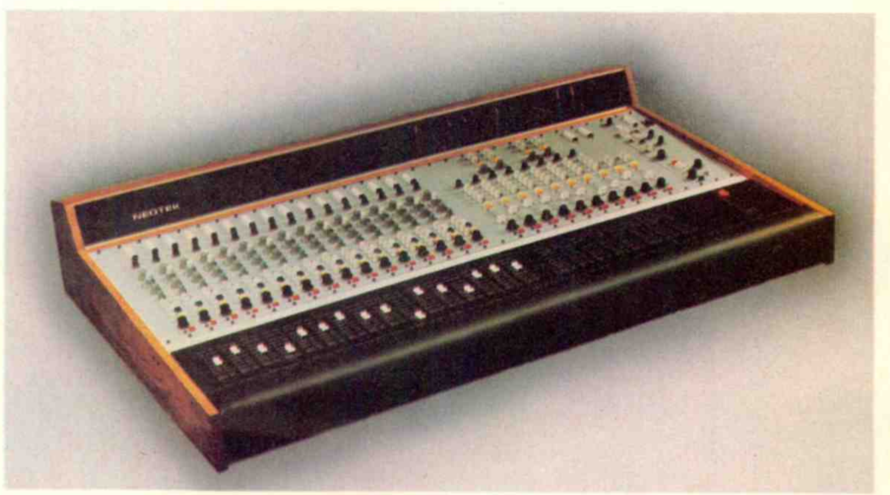

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What The
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. . . for those brave enough to try —
PRODUCTION VIDEO EQUIPMENT PACKAGES ________ by Steve Barnett

— continued

the additional necessary equipment is minimal when compared to the cost of outfitting a complete video studio. In many instances, as in the case of Houston Recording, in Cucamonga, California, a tie-in can be made to the video truck's own SMPTE time code generator, thereby providing sync with little cost to the audio studio. Though such an arrangement is quite basic and simple, it is a start nonetheless.

For those audio studios wishing to explore the areas of video production for themselves, the cost factors are becoming more and more manageable, though a thorough evaluation should be made of every possible aspect or pitfall of the investment. Some of these were mentioned in the two previous issues of this magazine. Technology, however, has made the jump a little more within the bounds of financial reason.

"Video is becoming less and less expensive in the areas of semi-pro and professional level video gear," observes Martin Polon, associate editor of Video magazine and a consulting editor for Recording Engineer/Producer.

"You're talking about names that were virtually unknown five years ago, like Ikagami and the ENG people, who are developing postures here. You're talking about companies like Hitachi, Sony, JVC, and Panasonic, who are showing color cameras that are very, very high quality and small in size that fall into a package-price range of about $25,000."

In terms of video cassette recorders, companies like JVC, Sony, and Panasonic are manufacturing ¾-inch machines which are becoming quite affordable. The same companies are also able to supply switchers (switchers are to the video man, what mixing consoles are to the audio man) or there are a number of other companies that specialize in making switchers only, who may be approached.. There is a fairly broad range of hardware available on the market that will do creditably, according to the specific definition of the job to be done.

Keeping this in mind, we've put together several representative video packages in this price range. In discussing these packages we took the position of a recording studio wishing to enter video tape production to provide its clients with video demos for both club circuits and the record companies. The ¾-inch color format was emphasized because of its professional quality video and its editing capabilities . . . at a price far below the 1" on-the-air systems, although technology is creating other acceptable alternative formats, including much improved ½" systems. Manufacturers of the equipment included in the sample packages were selected on an arbitrary basis to get a cross section of what's being offered in the field, and no particular product line was endorsed.

Another area of discussion with the company representatives in reference to their cameras was the equipment's ability to interface with a professional one-inch format should the need arise. However, a studio should be absolutely certain that it can deliver product that justifies the expense of the equipemnt if they should decide to move into this high-end pro-video field even on an occasional basis, using rented equipment. This requires no little amount of expertise and experience.

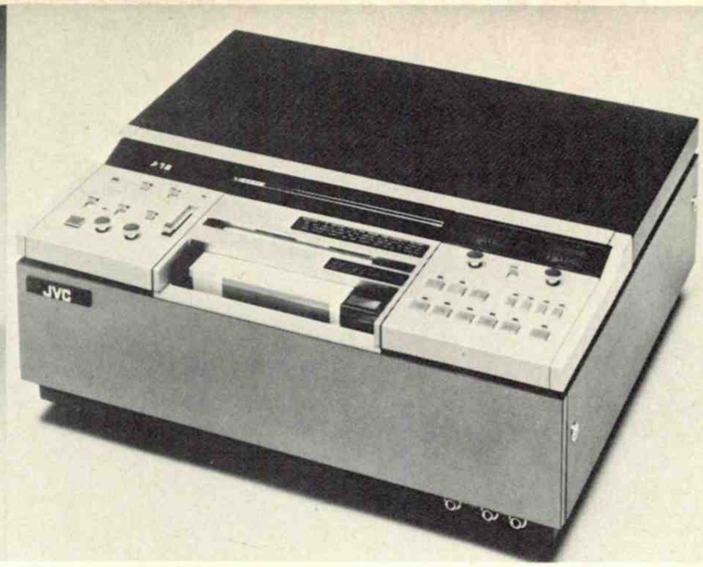
"If they want to do first level pro production, they can always rent the one-inch recorder, cameras and switchers, etc.," says Polon. "It may be worth spending the extra money to get better picture and camera quality but," as Polon points out, "one-inch availability from rental houses is usually found only in large market areas. One should also remember that competition is fierce among high quality video production houses and entering *their* market should not be taken lightly."

THE EQUIPMENT

Ron Suttle, National Video Products Manager for JVC's Pro Video Division, described his company's ¾" equipment as consisting of a system of two cameras each feeding, in isolation, two separate video recorders. The two tapes can then be edited

. . . the JVC RM-88U editing controller

. . . the JVC Model CR-8200U, ¾" recorder/playback/editing deck



together in post-production. The camera Suttle recommended was the JVC S-100.

"That's our newest camera," he explains, "and it will be available this fall. It uses a single Saticon striped tube and will be available for under $4,000."

Though acceptable camera performance in low light levels can be a matter of taste, with some New Wave artists actually preferring the gritty, off-color look achieved in club lighting situations, the S-100 will perform better with fewer foot-candles of light than its JVC predecessors.

3/4-Inch Video Recorders/Editors

A couple of combinations can be suggested for ¾-inch video cassette decks from JVC. Both include the JVC 8200. "The 8200 is a recorder-player and editor." says Suttle, "and you can use it in isolation (recording) and then again while editing. It lists for $5,300. Another deck is the JVC 6600, and it's a recorder-player, but not an editor, and costs less, $3,400."

A cassette deck with editing obviously has the capability of making video cuts and inserts, while the recorder/playback units can only do just what their name implies. A way of assembling a system here would be to have one camera feeding the JVC 8200 and the other feeding the 6600 deck. In post-production, then, the two tapes could be assembled by inserting material from the 6600 onto the 8200. To accomplish this, an editing controller is required.

"We make two of them." says Suttle, "One is the RM88, which is $3,200, and the RM82, which is $1,900. The 88 is a full-blown deck with everything that you really need in editing, and the 82 is more limited in its features and effects."

For quality control, Suttle suggests a test and generator package be incorporated into the production system. Such equipment can be obtained from companies such as Tektronics or Lenco for around $3,000.

—continued on page 135 . . .

STUDIO UPDATE

Teig Brosious

Northeast:

■ **ATLANTIC STUDIOS** (New York City), a division of Atlantic Recording Corporation, is undergoing the most extensive upgrading in its history, including the purchase of new equipment from Audiotechniques, of Stamford, Connecticut. Included in the package are five MCI JH-24 transformerless tape recorders, two MCI 500 56-input consoles, and one MCI 500 36-input board. A second Neumann VMS-70 cutting room is also being added in the studio's remodeling and reconstruction, which is being handled by acoustician **ALAN FIERSTEIN,** architect **MAURICE WASSERMAN,** and interior decorator **DON HOPKINS.** The completion date for the project is early in 1981. **DAVE TIEG** is the studio's general manager. *1841 Broadway, New York, NY.*

■ **EASTERN ARTISTS RECORDING STUDIO, INC.** (East Orange, New Jersey) has announced the appointment of **NEAL STEINGART** to the position of chief engineer. Steingart is a graduate of the RCA Institute, was a staff engineer at the Record Plant, and built and operated Fly Studios, in Brooklyn, prior to joining the E.A.R.S. staff. Along with E.A.R.S. president **BILL GALANTY,** Steingart recently completed installation of an APSI 32-channel console in the East Orange facility. *36 Meadow Street, East Orange, NJ 07017. (201) 673-5680.*

■ **SAXONY RECORDING STUDIOS** (Rouses Point, New York) announces the opening of its new 24-track facilities on the shore of Lake Champlain. Acoustic design for the room was handled by **JOHN STORYK,** of Sugarloaf View, with a console shipped in from Criteria's Studio C in Miami. Owner **MARC CHAPMAN** also mentions the capability of live 24-track recording from the night club three floors below Saxony. **ALAN ROWOTH** is staff engineer. *228 Lake Street, Rouses Point, NY 12979. (518) 297-6543.*

■ **VIRTUE RECORDING STUDIOS** (Philadelphia) has taken delivery of two MCI 2-track machines for mastering and mixdown. Engineering at the studios is handled by **FRANK** and **MARYANN VIRTUE.** *1618 N. Broad Street, Philadelphia, PA 19121. (215) 769-9479.*

■ **SECRET SOUND STUDIO, INC.** (New York City) announces the addition of ¾-inch video recording and editing facilities, making the studio now capable of handling audio on its own video taping in or out of the studio. **JACK MALKEN** is the head engineer. *147 West 24th Street, New York, NY 10011. (212) 691-7674.*

■ **TURTLE BEACH RECORDING** (York, Pennsylvania) reports the addition of **DAVE LONG** to the staff in the position of director of technical services. Turtle Beach is owned by **RAY SMITH.** The announcement was made by **LAUREN HALE.** *1912 Alcott Road, York, PA 17402. (717) 757-6344.*

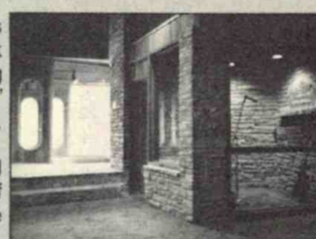
■ **VANGUARD RECORDING SOCIETY** (New York City) has opened its newly remodeled 24-track facility, which was redesigned by Vanguard engineer **JONATHAN THAYER,** incorporating a new MCI JH-636 computerized console with parametric EQ. The studio dimensions are 40' by 80' with a 20' ceiling, and includes two Steinway concert grand pianos, a Hammond organ, and a collection of guitar and bass amps. Digital mastering is also available. Chief engineer at Vanguard is **JEFF ZARAYA.** *71 West 23rd Street, New York, NY 10010. (212) 255-7732.*

■ **ELECTRIC LADY STUDIOS** (New York City) has appointed **DORY LANIER,** formerly of Full Tilt Studios, to the post of studio manager. The technical staff is now headed by **MICHAEL FRONDELLI** in the position of director of operations. ELS has also recently opened its new Studio C, which features a Neve 806A computerized console with Necam automation and a Westlake four-way monitoring system. The announcements were made by ELS president **ALAN SELBY.** *52 West 8th Street, New York, NY. (212) 677-4700.*

South Central:

■ **JACK CLEMENT RECORDING STUDIOS** (Nashville) has changed its name to **SOUND EMPORIUM,** according to studio manager **JIM WILLIAMSON.** The two-studio, 24-track facility has been known as Jack Clement Recording Studios since Clement opened the operation in 1969. In 1975, he sold the studio to producer **LARRY BUTLER** and financial consultant **AL C. MIFFLIN.** Williamson has managed the studio since 1974. *3102 Belmont Boulevard, Nashville, TN 37212. (615) 383-1982.*

■ **INDIAN CREEK RECORDING** (Uvalde, Texas) is located on a 4,000 acre ranch 80 miles west of San Antonio, and is equipped with a Neve console feeding Ampex 1200 24-track recorders, UREI Time Aligned™ monitors, and a full complement of outboard gear. Living accommodations are available on the ranch as well. **JOHN ROLLO,** late of The Kinks' London studio, Konk, is the chief engineer. *P.O. Box 487, 911 Cherry Street, Uvalde, TX 78801. (513) 278-5802.*



Indian Creek

■ **RUFF CEDAR SOUND STUDIO** (Austin, Texas) has upgraded to a fully automated 24-track facility with the addition of an MCI JH-536 console and the complete remodeling of the studio, which is located on nine acres of wooded park land. Plans are underway for the addition of a new room as well. The announcement was made by **RUSSELL WHITAKER.** *5012 Brighton Road, Austin, TX 78745.*

■ **THE MUSIC PLACE** (Birmingham, Alabama) has appointed **ANN HOLLOWAY** to the position of chief engineer and director of A&R and production. Holloway leaves T.K. Productions, where she had been a staff engineer and producer. **TOM HOLLOWAY** has also been named to the engineering staff following work with T.K., Quadradial, and other Miami area studios. *1817 Oxmoor Road, Birmingham, AL 35209. (205) 871-4221.*

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1

STUDIO UPDATE

Southeast:

■ **STRAWBERRY JAM STUDIOS** (West Columbia, South Carolina) has taken delivery of a new NEOTEK Series III console with installation handled by The Valley People, Inc., of Nashville. A Lexicon 224 digital reverb unit and a Lexicon 93 Prime Time DDL are on order. *3964 Apian Way, West Columbia, SC 29169. (803) 356-4540.*

■ **dgp STUDIOS** (North Miami, Florida) has opened its new 24-track Studio B, according to owner **DAVE GRAVELINE.** Acoustical and technical direction came from Studio Supply, with construction by Victor Corporation. Equipment is by MCI, Sound Workshop, DeltaLabs, and Klark-Teknik, with microphones including AKG, Electro-Voice, Beyer, Shure, and Sennheiser. Video capabilitites have also been installed. *1975 N.E. 149th Street, North Miami, FL 33181. (305) 940-6999.*

■ **ALPHA AUDIO** (Richmond, Virginia) announces the completion of its new Studio IV featuring a live-end, dead-end control room, and a studio utilizing a front wall of featherstone to break up standing waves; a side wall with an integral bass trap and high frequency reflector, and a dead rear wall of anechoic foam. The room is wired for 32 inputs, and can accommodate multi-channel recording upon request. *2049 W. Broad Street, Richmond, VA 23220. (804) 358-3852.*


Alpha Audio

■ **OF RECORDING STUDIO** (Fort Lauderdale, Florida) announces its grand opening with the facility featuring an MCI 16-track recorder and a Stevenson mixing console. A grand piano is included in the selection of keyboards and instruments, with in-house production handled by **GARY OHLSON** and **DAVE FEDER. DAVE GREENE** and **MIKE BRANDO** are the staff assistant engineers. *4602 N.E. 6th Avenue, Fort Lauderdale, FL 33334. (305) 753-7431.*

■ **TRACK RECORDING** (Summerville, South Carolina) has gone 16-track with the purchase of an MCI recorder, fed by a Tangent 3216 console. Remodeling has also been completed in the control room and drum chamber. *P.O. Box 857, 208 S. Cedar Street, Summerville, SC 29483. (803) 873-0607.*

■ **AUDIO IMAGE** (Pompano Beach, Florida) has upgraded with the addition of a 16-track recorder, an URSA Major Space Station, a Symetrix Signal Gate, Symetrix compression, dbx, and a complete line of mikes by Audio-Technica and Sennheiser. Owner/manager **ROB HENION** and engineer **MIKE HOFFMAN** also report plans to go 32-track. *P.O. Box 5681, Lighthouse Point, FL 33064. (305) 943-5590.*

Midwest:

■ **SOLID SOUND, INC.** (Ann Arbor, Michigan) reports the addition of a new Lexicon 224 digital reverb unit as well as a Lexicon Prime Time digital delay. In addition, **DANA GROSS** has been appointed business coordinator for the studio. The announcements were made by Solid Sound president **R. G. MARTENS.** *P.O. Box 7611, 1286 Dixboro Road, Ann Arbor, MI 48017. (313) 662-0667.*

■ **MASTERPIECE SOUND STUDIOS** (Detroit, Michigan) is operating with a Loft Series 440 16 x 8 console feeding a modified 3M 8-track recorder. Outboard gear includes Lexicon Prime Time, URSA Major Space Station, and two Loft flangers. *1611 Webb Street, Detroit, MI 48206. (313) 867-7874.*

■ **DAYSTAR RECORDING COMPANY, INC.** (Joliet, Illinois) is planning installation of a UREI 813 Time Aligned™ monitoring system along with an upgrade to 24-track and an expansion of the main studio. These changes would be in addition to the 32-channel custom console and the array of outboards, which include harmonizers, parametrics, live echo, Ecoplate, and delay lines. A full drum kit, a grand piano, and a Hammond B-3 organ are among the instruments available. **JACK BARBRE** is the president and chief engineer. *509 W. Jefferson Street, Joliet, IL 60435. (815) 723-8488.*

■ **FIFTH FLOOR RECORDING STUDIOS** (Cincinnati, Ohio) has installed UREI 813 Time Aligned™ monitors in its operation. *517 W. Third Street, Cincinnati, OH 45202. (513) 651-1871.*

■ **COUNTERPART CREATIVE STUDIOS** (Cincinnati, Ohio) has added **DAVID SHEWARD** to the staff in the position of chief engineer and studio manager. The announcement was made by Counterpart president **SHAD O'SHEA.** *3744 Applegate Avenue, Cincinnati, OH 45211. (513) 661-8810.*

■ **STUDIOMEDIA ENTERPRISES, INC.** (Evanston, Illinois) has opened its new recording complex which features a studio plan by **JERRY MILAM,** of Milam Audio. The 24-track studio is equipped with MCI tape machines, a NEOTEK console, Doly A noise reduction, and a UREI monitoring system. Instruments include a Baldwin concert grand piano, full drum kit, a Rhodes Stereo 88, and a number of synthesizers. *1030 Davis Street, Evanston, IL 60201. (312) 864-4460.*

■ **ALPHA RECORDING COMPANY** (Lombard, Illinois) has added to its equipment an Ecoplate, an Ampex 440-C 2-track recorder, Dolby A, Ashly limiters, an aural exciter, and a number of Telefunken mikes. The studio is a full 24-track operation featuring an MCI 24-track recorder, a NEOTEK console, dbx noise reduction, EMT and AKG echo, and an assortment of keyboards, drums, and amps. *515 W. Harrison, Lombard, IL 60148. (312) 495-2241.*

■ **PYRAMID AUDIO, INC.** (South Holland, Illinois) is currently planning expansion of its facility with plans for opening by the end of the year. The new room is being designed by **BOB VUKELICH,** of Pyramid, and will feature a new Otari MTR-90 and possibly a Soundcraft Series 3B. Pyramid is currently an MCI equipped 24-track studio. *16240 Prince Drive, South Holland, IL 60473. (312) 339-8014.*

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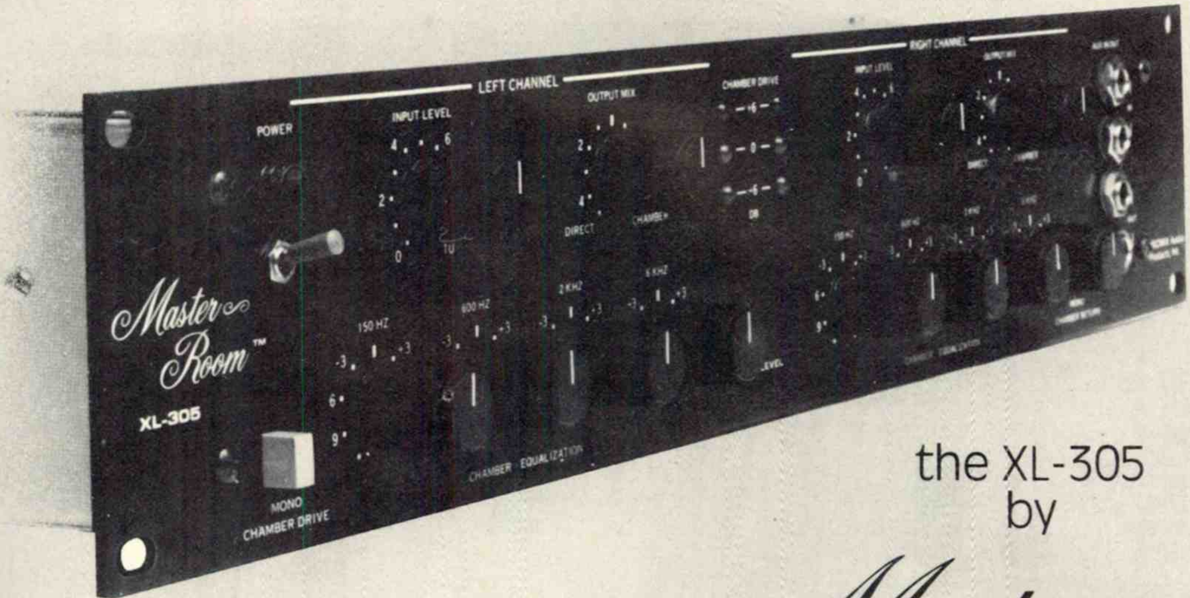

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for additional information circle no. 11

STUDIO UPDATE

Rocky Mountain:

■ **ASPEN RECORDING STUDIOS** (Aspen, Colorado) has installed a UREI Time Aligned™ monitoring system with TAD drivers, and White 1/6-octave equalization. *P.O. Box 1915, Aspen, CO 81611. (303) 925-8414.*

Southwest:

■ **JOHN WAGNER PRODUCTIONS, INC.** (Albuquerque, New Mexico) is now operating two studios to serve in-house and outside recording needs. Studio A is equipped with an MCI 24-track recorder, a JWP custom 26-channel console, dbx, and UREI monitors. Studio B features an MCI 16-track, a JWP 20-channel board, dbx, and JBL monitors. *202 Wisconsin N.E., Albuquerque, NM 87108. (505) 265-3441.*

Southern California:

■ **FIFTY-FOUR EAST SOUND RECORDERS, LTD.** (Pasadena, California) features a 30' x 34' studio equipped with an APR 44 x 40 console with 8 VCA sub-groups and automation linked to Ampex recorders. Monitors are by JBL, Augsburger, and Auratone, and outboards include an Eventide Harmonizer, an Aphex Aural Exciter, DeltaLabs DDL, UREI limiters, and dbx noise reduction. Mikes are by Neumann, AKG, Sennheiser, and Pearl, while a Yamaha C-7 piano and a Hammond organ with Leslie are among the instruments offered. Film scoring capabilitites are also featured. *54 E. Colorado Boulevard, Pasadena, CA 91105. (213) 796-5630.*

■ **MAD DOG STUDIO** (Venice, California) announces the opening of their new facility featuring a Scully 16-track recorder and other machines by Ampex, Techniques, and Tascam. Outboards include tube and solid state limiters, various digital delay lines, and a number of flangers. Keyboards available are by Oberheim, among others. *1715 Lincoln Boulevard, Venice, CA 90291. (213) 306-0950.*

■ **MONTEREY RECORDING STUDIOS** (Glendale, California) welcomes **LES BROCKMANN** to their engineering staff following his Downbeat Studio Recording Award for live performance recording. Brockmann is out of the University of Miami recording program. *230 S. Orange Street, Glendale, CA 91204. (213) 240-9046.*

■ **CAPITOL STUDIOS** (Hollywood, California) has appointed **CHARLES COMELLI** to the post of recording production manager and **JOHN HANLON** to the position of night recording supervisor. Capitol Records' director of studio administration, **RICHARD BLINN,** explains that Comelli previously was production manager of the company's film and video department, while Hanlon carries a background in mixing, editing, and maintenance at several Hollywood studios prior to joining Capitol. Both will report to studio manager **JOHN KRAUS.** *1750 N. Vine Street, Hollywood, CA 90028. (213) 462-6252.*

■ **CIRCLE SOUND STUDIOS** (San Diego, California) has upgraded its operation to 24-track with the installation of an MCI 440 transformerless console along with a UREI Time Aligned™ 813 monitoring system and Lexicon 224 digital reverberation. Other equipment at the studio is by Eventide, Studer, dbx, and JBL, with the instrument selection boasting Baldwin grand and upright pianos and Ludwig drums. *3465 El Cajon Boulevard, San Diego, CA 92104. (714) 280-7310.*

■ **K-DISC MASTERING** (Hollywood, California) announces the availability of disc mastering utilizing a new Sony PCM-1600 digital audio mastering system. **BILL LIGHTNER,** of K-Disc, adds that the new equipment will work in conjunction with the operation's two Neumann VMS-80 disc mastering lathes. Lightner's experience includes prior work with the PCM-1600 as well as with direct-to-disc recording for Century Records. *6550 Sunset Boulevard, Hollywood, CA 90028. (213) 466-1323.*

■ **SALTY DOG RECORDING STUDIOS** (Van Nuys, California) has upgraded with the addition of two 3M-79 2-track machines. All the main recorders are now 3M, with two additional Ampex 2-track machines for copies and tape delay. A vocal stressor and a vintage Altec 1591A compressor have also been added to the rack. *14511 Delano, Van Nuys, CA 91411. (213) 994-9973.*

Mars

■ **THE MARS STUDIOS** (Hollywood, California) has completed their new 24-track facility with design by **JOHN EDWARDS** and construction by **STAN BRUCE.** The studio is 45' x 35' with a 13' to 22' sloped ceiling and four acoustically varied open ended isolation booths. The control room is centered around a Trident TSM Series 40 x 32 console feeding an MCI JH-16 24-track recorder with AutoLocator III. Other 2-track machines are by Studer and MCI, with the monitoring system consisting of JBL 4343 bi-amped units, 4313s, and Auratones. Sideboard gear includes full dbx, two Ecoplates, Lexicon 224 digital reverb, and a full collection of compressor/limiters, gates, and DDL. Mikes are by Sennheiser, Shure, Neumann, and AKG. *665 N. Berendo Street, Hollywood, CA 90004. (213) 660-6334.*

Northern California:

■ **WHITE RABBIT, INC.** (Sausalito, California) has opened its doors for public booking with addition of an Ashly parametric equalizer, tunable notch filters, a DeltaLabs DL2 digital delay, and Neumann 87 microphones. **CRAIG TALMY** and **FERNANDO KRAL** have been added to the engineering staff to help handle the new business. *301 Harbor Drive, Sausalito, CA 94965. (415) 332-4852.*

■ **THE AUTOMATT** (San Francisco, California) is utilizing its new 3M 32-track digital recording equipment on the first multi-track album recorded in this format in Northern California. The Columbia LP, entitled "Swing of Delight," features **CARLOS SANTANA** and **HERBIE HANCOCK,** and is being produced by Automatt owner **DAVID RUBINSON.** *827 Folsom Street, San Francisco, CA 94107. (415) 777-2930.*

Rubinson Hancock Santana

Northwest:

■ **THUNDER OAK AUDIO** (Bothell, Washington) celebrated its second anniversary with the addition of an MCI JH-16 with AutoLocator III. This is the latest in a series of upgradings which have included the installation of a Neve console, UREI monitors, an Ampex ATR-102, and an ADR Scamp Rack System. Staff engineers are **VIC COUPEZ** and **LARRY NEFZGER.** *23717 Bothell Way South East, Bothell, WA 98011. (206) 487-2177.*

■ **LEW'S RECORDING PLACE** (Seattle, Washington) has promoted **CARY WAKELEY** to the post of studio manager. The operation has also added dbx noise reduction, a second editing room, and location recording facilities. *1219 Westlake North, Seattle, WA 98109. (206) 285-7550.*

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for additional information circle no. 12

STUDIO UPDATE

Hawaii:

■ **SEA-WEST RECORDING STUDIOS** (Honolulu, Hawaii) have installed a new MCI JH-24-24 recorder with AutoLocator IV. The new machine is totally transformerless with differential inputs, outputs, and head coupling. *P.O. Box 30186, Honolulu, HI 96820. (808) 293-1800.*

Paradise

Australia:

■ **PARADISE RECORDERS** (Sydney, Australia) is now operational under the new management and ownership of **BILL FIELD.** The 24-track studio is of an Eastlake **TOM HIDLEY** design and features a Harrison 36 x 24 console with Allison automation, MCI recorders, an EMT 140 echo plate, a harmonizer 949, and a Yamaha concert grand piano. **RICHARD LUSH,** late of Abbey Road Studios, in London, is the chief engineer. *70 Judge Street, Woolloomooloo, Sydney, NSW 2011, Australia. Telephone: (02) 357-1599.*

■ **ALLAN EATON SOUND SERVICES** (Melbourne, Australia) is now capable of recording music to 35 millimeter picture through the use of an MCI AutoLock interfaced by Sontrom with a computer to allow the MCI 24-track recorder to follow the projector. Further modifications by Sontron have updated the projector, which is now servo-controlled with full logic switching and rewind, and fast forward at twice the normal speed. *80 Inkerman Street, Saint Kilda, Melbourne, Australia.*

■ **LEEDS MUSIC FACTORY** (Sydney, Australia) has purchased an Otari MTR-90 24-track recorder to go with their Soundcraft Series III console. The latter now has a full compliment of inputs, featuring 24-channels of 4-band parametric sweepable EQ. Studio monitors are by JBL. *4343 Oxford Street, Darlinghurst, Sydney, Australia. Telephone: 31-3314.*

■ **LEO RECORDERS** (Sydney, Australia) formerly **UNITED SOUND STUDIOS,** have taken delivery of a new Lyrec TR-532 24-track recorder with remote, two Tannoy Super Red monitors, a Lexicon Prime Time, a Lexicon 224 digital delay unit, and an EMT 258 noise filter. This equipment is in conjunction with the studio's recent remodeling. *21 Pier Street, Haymarket 2000, Sydney, Australia. Telephone: 26-6816.*

Spencer Lee — Leo

Africa:

■ **GOLDEN BAOBOB RECORDING STUDIO** (Dakar, Senegal, Africa) has taken delivery of an MCI JH-600-36 recording console along with MCI JH-114 and 110 recorders, Big Red monitors, and a 24-track Dolby noise reduction unit. The facility is owned by **FRANCIS SENGHOR,** the son of Senegal's president. Audiotechniques, Inc., of Stamford, Connecticut, supplied the equipment, which was shipped to Senegal by Air Express International Corporation, also of Stamford.

MCI for Baobob

■ **PHONODISK (NIG.) LTD.** (Lagos, Nigeria) has opened its 24-track studio, record pressing, and packaging plant 100 kilometers outside of Lagos. The operation offers an MCI 428 console, an MCI JH-114 24-track recorder, two MCI JH-110 2-track machines, Dolby noise reduction, and microphones by Neumann, Shure, Sennheiser, and AKG. A full complement of outboards is also featured. The disc cutting room utilizes a Scully lathe with a Westrex 3D11 cutter head. Equipment was provided by Audiotechniques and installed by Ransteele Audio, of New York, with **CHARLES GERBER,** Phonodisk's chief recording and cutting engineer. *40 Ikorodu Road, Igbobi, Lagos, Nigeria.*

United Kingdom:

■ **ROUNDHOUSE RECORDING STUDIOS** (London, England) is offering 3M digital mastering to the United Kingdom along with its 3M 32-track digital recording system in its multi-track studio. Roundhouse has installed a 3M editing system and a 3M 4-track master recorder with digital lathe preview unit.

■ **HORIZON RECORDING STUDIOS** (Coventry, England) has taken delivery of a new Trident Series 80 32 x 24 x 24 console. *Horizon House, Warwick Road, Coventry, England CV3 6Q5. Telephone: (0203) 21000.*

■ **THE POINT RECORDING STUDIO** (London, England) is designed for laying down basic tracks for an outside mixdown, according to its co-director, **ALAN O'DUFFY.** The facility features an Allen & Heath 28 x 24 Syncon board feeding an Otari MTR-90 multi-track recorder and Quad powered monitors by KEF, Tannoy, and Radford. The recording area measures 30 feet by 25 feet, and keyboards include a concert grand and electric pianos by Yamaha, Hammond, and RMI.

■ **ODYSSEY STUDIOS** (London, England) is owned by State Records and is comprised of two studios, one with an 18 foot ceiling capable of accommodating 50 musicians, with the other smaller room used for solo instruments and vocals. Both control rooms feature identical custom built MCI JH-500 56 x 32 consoles with automation feeding MCI tape machines with full Dolby noise reduction and sync capability to 46-tracks. Monitoring is by BGW powered Audicons, and other gear includes an Aphex Aural Exciter, Audio & Design compressor/limiters, EMT echo plates, and a wide selection of Neumann and AKG microphones. Acoustic designer and studio director of Odyssey is **KEITH SLAUGHTER.**

have you?
• increased track capacity — gone 24, 16, 8 •
• added key people • won awards •
• moved or expanded • added important equipment •
these are some of the interesting news items that can be announced in the next available issue. Write:
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YAMAHA

for additional information circle no. 13

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□ 15" diameter cone
□ 102dB SPL/1 meter/1 watt sensitivity
□ 30Hz to 4kHz Frequency Range (maximum recommended crossover, 800Hz)
□ 8 ohm [JA3882] and 16 ohm [JA3882(B)]

The new JA3881 is an excellent choice for use in stage monitors, keyboard speaker systems, or for 2-way systems. Its flexible suspension and lightweight edgewound aluminum coil give the JA3881 a usable response up to 5kHz.

□ 15" diameter cone
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□ 40Hz to 5kHz Frequency Range (maximum recommended crossover, 3kHz)
□ 8 ohm [JA3881] and 16 ohm [JA3881(B)]

High frequency compression driver: JA6681B With high sensitivity and high frequency handling capacity, the JA6681B high frequency driver makes an excellent mid-to-high frequency reproducer for use in 2- or 3-way full-range, high-level sound reinforcement or monitor speaker systems. Its unique S-shaped beryllium/copper suspension system and quality construction ensure long life.

□ 145dB SPL reference sensitivity, 1 watt, using 1 inch plane wave tube
□ 108dB SPL at 1 meter, 1 watt (using Yamaha H1230 horn)
□ Nominal 800Hz-12kHz, usable down to 500Hz
□ 16 ohms nominal impedance

Combination high frequency horn & driver: JA4280B/H1400 This high frequency reproducer's versatility enables it to be used as the mid and high frequency reproducer in a full-range stage monitor, keyboard monitor, or general sound reinforcement system. Or as the upper midrange reproducer in an ultra-wide-range system.

□ 90° H x 40° V dispersion
□ 106dB SPL at 1 meter, 1 watt
□ Aluminum horn with damping
□ 16 ohms nominal impedance
□ Nominal 1,500Hz-16kHz, usable down to 800Hz.

Compression tweeter: JA4281B This new tweeter is a high-sensitivity, integral horn/driver unit designed to handle the uppermost portion of the frequency spectrum. It is an excellent super-tweeter for use in 3-way or 4-way full-range, high-level sound reinforcement systems when used in conjunction with our JA6681B high frequency compression driver. Its superb on/off axis response and absence of diaphragm resonances also make it a fine choice for studio monitor systems.

□ 120° dispersion pattern at 10kHz
□ 108dB SPL at 1 meter, 1 watt
□ Nominal 7kHz-20kHz
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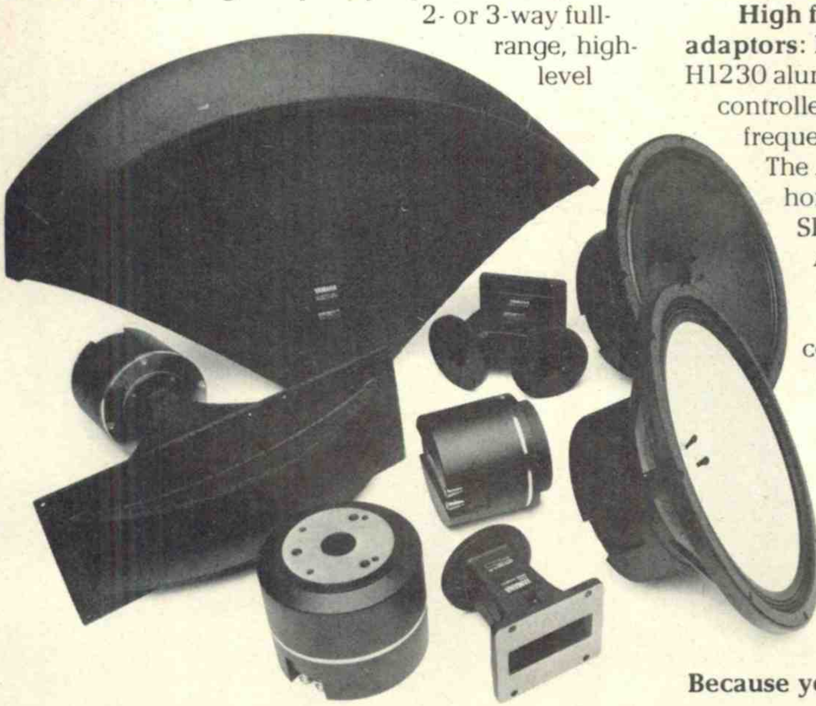
High frequency radial horn and throat adaptors: H1230, AD3500 & AD3502 The new H1230 aluminum radial horn is designed to provide controlled dispersion (90° H x 40° V) of high frequencies in high-level, wide-range systems. The AD3500 throat adaptor is used to couple the horn to the JA6681B driver to produce 108dB SPL at 1 meter with 1 watt input. Use the AD3502 throat adaptor to connect two drivers for greater output.



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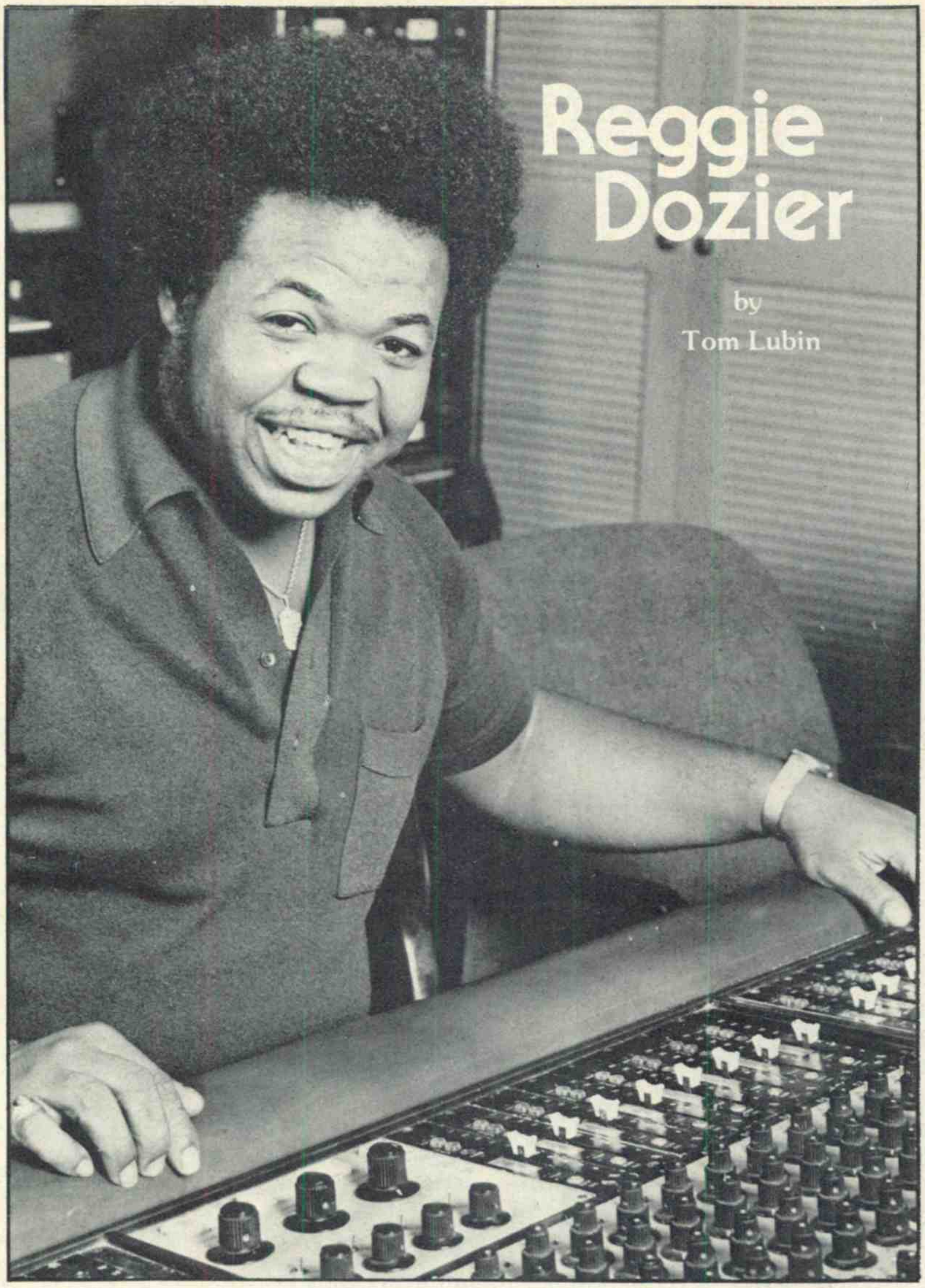


— UPWARD MOBILITY —

When he talks about Motown . . . Reggie Dozier literally means starting out on a production line job at Ford. Currently chief engineer at L.A.'s Scott-Sunstorm Studios, his credits include well over 150 albums with artists such as Nancy Wilson, Ronnie Laws, Aretha Franklin, Gabor Szabo, Wayne Henderson, Marilyn McCoo & Billy Davis, Jr., The Biblical Gospel Singers, Lamont Dozier . . .

R-e/p (Tom Lubin): The last year or so has seen a wide interest and increased sales in Gospel albums. You've been recording them for some time.
Reggie Dozier: Yes, at one time ABC had a Gospel label. They liked the way I worked with Gospel acts.

R-e/p (Tom Lubin): It seems to me the approach on Gospel recordings would have to be much different.
Reggie Dozier: It is in the sense that it's done very fast. They don't have the budgets that pop has. They might have seven thousand for the whole album. When they go in they often do the vocals at the same time they do the tracks. When they get through it's just a matter of mixing it.

You're right though, Gospel music has recently become much more popular, helped partly by the Gospel segments on the recent Grammy Show. It's a style of music. Gospel is more pop, rock, and R&B these days. The tunes are relating to everyday life . . . they are Gospel . . . but they aren't. It's not totally Praise the Lord, etc., it's more about people loving one another and working together.

The style has also changed. I remember working with the biblical gospel singers when they were just using mostly rhythm tracks. About two years after that, I noticed they were starting to bring in strings. Now they are using more instruments in general; more percussion, more of everything.

R-e/p (Tom Lubin): As an engineer, how does your approach to recording Gospel vary?
Reggie Dozier: It doesn't change a great deal. R&B is very close to Gospel. Some groups have a lot of people who sing the chorus parts, others have one or two, for instance. Aretha Franklin, who's into Gospel. If we have a big choir coming in, we round up and use every pair of phones we own. We also have a lot of single earphones.

R-e/p (Tom Lubin): What do you think your most difficult sessions are?
Reggie Dozier: I'd have to say rhythm. It's more involved, very intense. I'm dealing with a lot of microphones and sometimes a lot of people. The basics are the jumping point of multi-track production.

R-e/p: How long do you usually have to get together a rhythm sound?
RD: It's never long enough in a sense. If I play a take from the first take I recorded to the last, you'll hear a hell of a difference. I'm not saying that the first things are bad, they're not. But I make improvements all the way to the end of the session.

R-e/p: When you begin a session, how do you set up your tracking date?
RD: For a set of drums with four toms, I'll mike each tom individually. If they are using left and right cymbals I will normally put one mike to each side overhead. I also mike the high-hat, snare, and sometimes I'll place two or three mikes on the kick drum, and select which one I want when we start the tune. If the tune is a ballad and we need a bigger sound on the kick, I'll probably use a FET U47 or RE-20 to make it fat, and a Sennheiser 421 if I want it harder. The RE-20 is the heaviest of the three.

I never leave the sound alone. I'm always trying to make it better. I'll listen to the playback and evaluate what I've got and make changes as I go. I also go into the room as much as possible.

R-e/p: What do you use on the toms?
RD: Sometimes I'll use Shure 546s, and I've recently begun to use the Shure SM-81, which is a very big-sounding mike. I also like AKG 452s or Neumann U87s on toms, but it depends on the drum set. If the person has a lot of drums I won't use 87s, because I'll get too much phasing. Their cardioid pattern is just too wide. I might use the 87s on the floor toms, but never on the top. I spend more time getting a drum sound than

— continued overleaf . . .

Freddie started backup singing in his New Jersey junior high school. He earned a Bachelor of Music Degree from Howard University, and taught in Washington, D.C., while moonlighting as a producer. In 1969, his first Motown production, "I Want You Back" by the Jackson Five, went platinum. Since then, he has collected close to 30 gold or platinum records. Freddie now owns his own studio in L.A. and has recently produced disco hits for Yvonne Elliman, Tavares, David Naughton, Gloria Gaynor, and Peaches and Herb.

ON CREATIVE EXPRESSION

"I'm thinking charts. I'm thinking commercial. And I'm thinking hit, as opposed to creative expression. Because that's usually what I'm hired for. I mean, I hear the standard rap that I would get from a company person or a manager is that 'this group, live, is a knockout. I mean, they're killers. All they need is that hit record. When they get that hit record, man, you're gonna see the baddest group that ever existed in the history of recorded music.' So they want the charts. And that's why I approach it like that."

ON HEARING

"I only go by the ears, and I do hear very well. Musically and technically. I hear stuff all over the place. The guitar player—if he accidentally hits an open A string while he's fingering a chord, we could have thirty pieces on tape and I'll hear that and solo it out and bust him—say, 'Hey, could you keep that string quiet?' He says, 'You mean you actually heard that?' So my ears are really my fortune. That's where everything lies. Right in my ears."

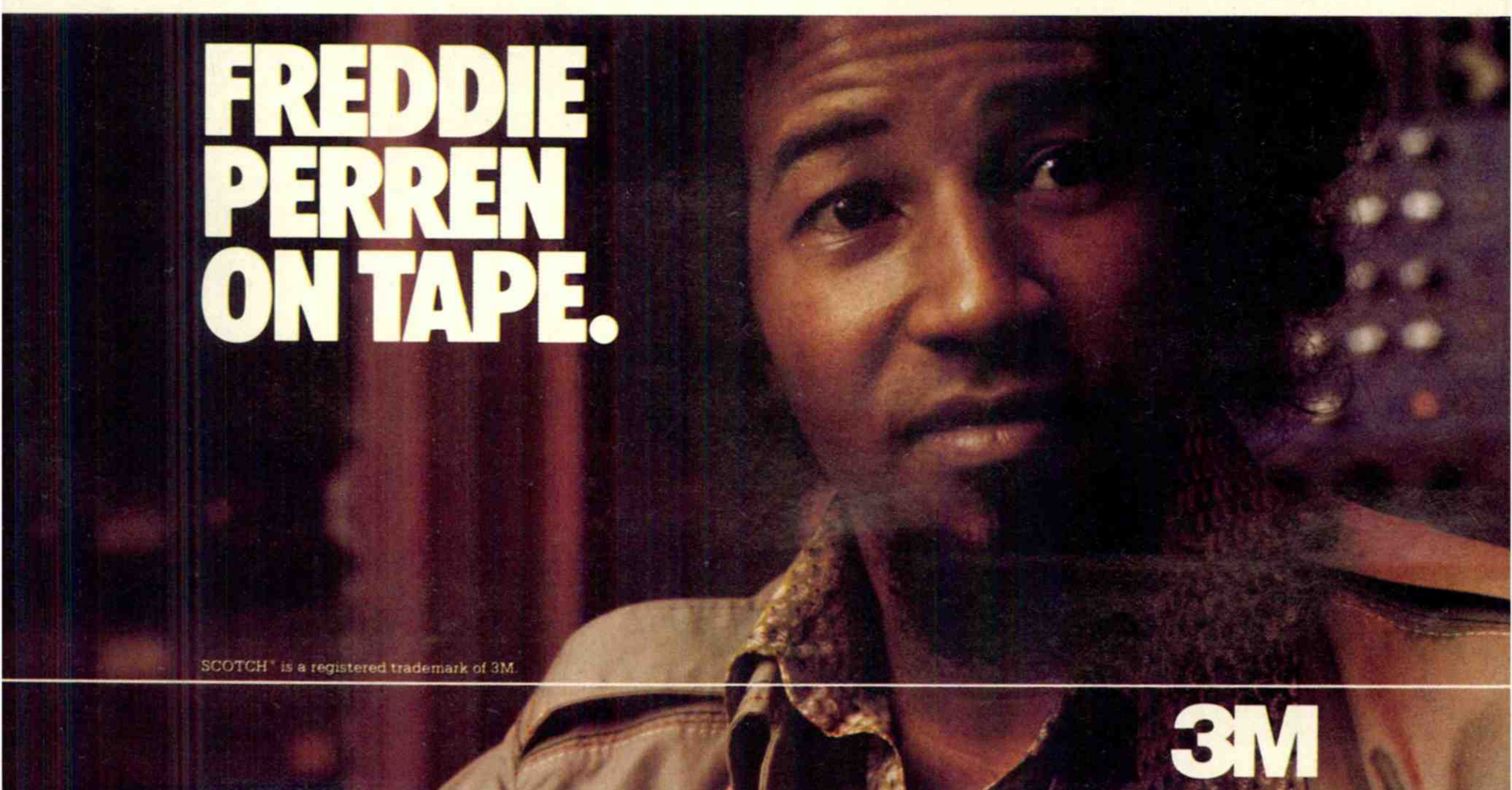
ON RHYTHM SESSIONS

"I do my basic rundown on the rhythm date. The guys are really cookin' and the groove is there and everything. I come in and take a listen to what kinds of sounds I have. But if that sound is not there, then I don't record until the sound is right. There may be some other producers who would just go with the flow. 'If it's groovin', hey, you know, we'll save it in the mix.' But I've attempted to save things in the mix. It doesn't happen. It has to be on tape."

ON TAPE

"I do not know much about the characteristics, physically, of what tape is made of. I'm not too much into that—the chemistry involved. However, after spending six years at Motown—they had many, many rules and regulations. Now, one was that we always use Scotch Tape. When I ventured off into the world of independent producing, out of habit, and not wanting to change a good thing, I went right back to the same tape, which was 250. And I was then approached by other engineers telling me that if you switched, you could increase your performances here—you know, the bottom end, so forth and so on. And I did stray away and I did try cutting other projects on different types of tape. And the bottom line is that I came back to Scotch. I can't say that I noticed the difference of, you know, 3 dB and the low end with Scotch, and the other only gave me a dB-and-a-half. I can't say that. I only go with my ears, which tell me that my home is with Scotch Tape."

SCOTCH 250
WHEN YOU LISTEN FOR A LIVING.



anything. I'm very picky; I'll go back and forth until I have what I want. It takes awhile, but once it's done everything else falls into place. On overheads I'll use AKG 452's . . . 414's . . . or sometimes Neumann KM-84's. I get to within a foot or so. If you get the condenser microphone too close it tears them up and the sound won't really be true.

On the snare there's a 546, but I have used a 414 a couple of times. If possible I try to have the drummer come in a little early to get him set up. If that doesn't happen, what I try to do is have the second engineer go out and beat on the drums so I can get the sound in the ball park. I'll also put a blanket and a weight inside the kick drum shell and lay it a little against the head, just enough to tighten the skin. Then I'll put a blanket over the kick and its mikes. If that's just not quiet enough isolation I'll use a Keypex on the kick just to get rid of the leakage.

I try to get the drummer to tune his drums, but sometimes have to resort to processing. Though I'd rather get it right in the studio and have it sound the same in the control room. Most of the time I'm more inclined to change the mike than reach for the EQ. If I start out with one mike and I need to EQ it by 4 dB one way or the other, then I go ahead and change the mike. I like it natural. For the same reason I seldom use a limiter.

Because I use so many microphones I'm constantly checking the oscilloscope to make sure I don't have a phasing problem. If I do I'll move the mikes until their combined outputs look and sound correct.

R-e/p: Do you mike the bass?

RD: Sometimes. For a jazz thing, I would mike the bass. For R & B I wouldn't mike it. With jazz I'd use a Sennheiser 421. It's always given me a better sound on the bass amp. For most other things I use a Carl Countryman direct box.

R-e/p: Do you put up go-bo's between all the instruments?

RD: In front of the drums, around the piano, in front of the Rhodes.

It depends on the sound the client wants. HDH did a "Rock 'n Roll" tune . . . and didn't use as many baffles, so there was a lot of leakage . . . that was the sound back then and that's what they wanted. On the other hand, Wayne Henderson wants it all baffled with everything nice and tight. I try to keep it a little open so that some of the room ambience is retained, but we do separate the instruments so they don't leak directly into each other without a whole lot of baffling.

R-e/p: Do you usually cover up the piano with a quilt?

RD: We have a specially designed cover. It's made so that it enclosed the entire piano with the lid open. It zips up, and there's zippered holes where you can stick the microphones. It's made out of heavy quilt material and was specially made for us by Alpha Acoustic Equipment, in Los Angeles.

R-e/p: What's your favorite piano mike?

RD: I'm back on the AKG 414's. The sound is cleaner and sharper, more tone. I used to use 87's, but they were a little dull to my ears, and just not bright enough.

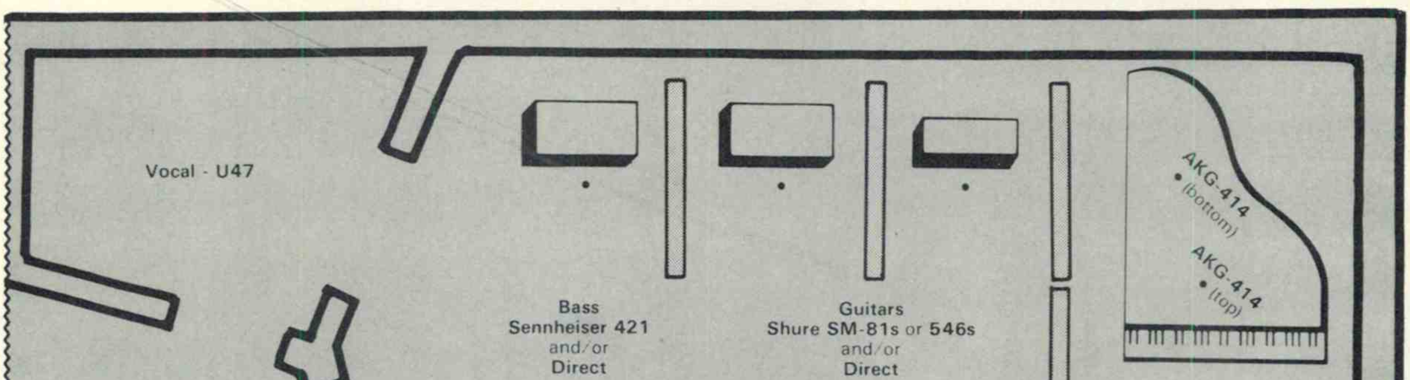
R-e/p: Are there any mikes you've recently "discovered?"

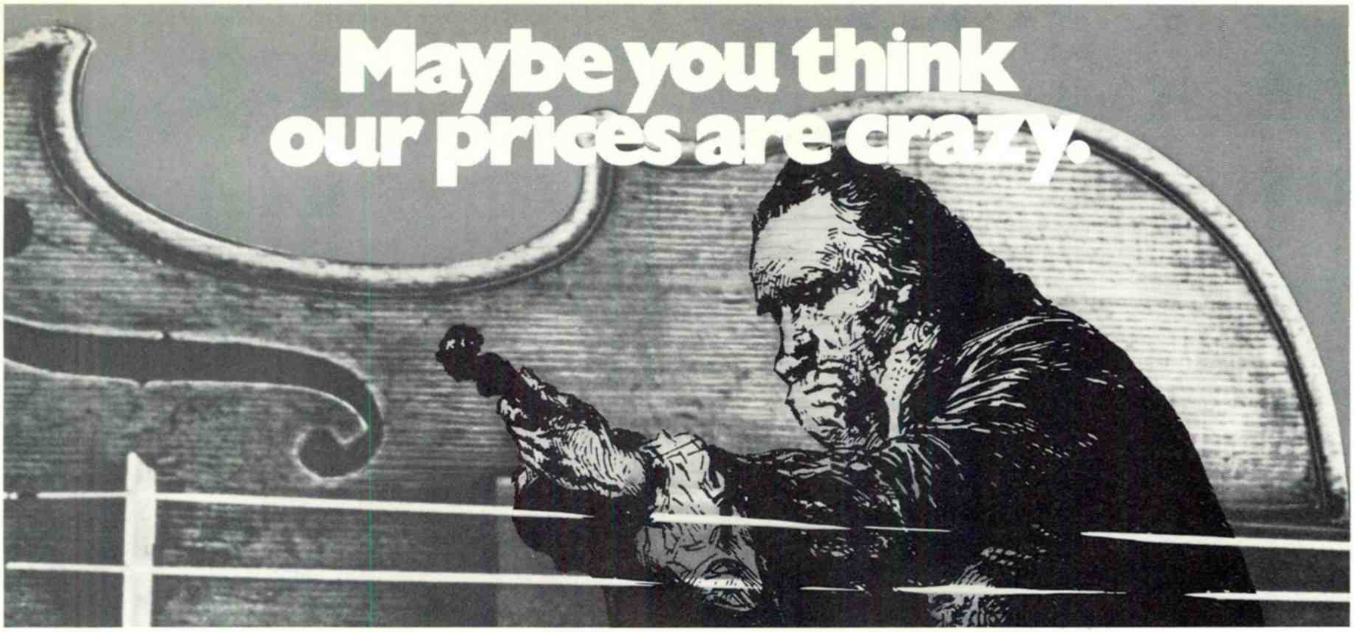
RD: Yes, the Shure SM-81. I use them as much as possible. I used two today on tom-toms and another on electric guitar. It's great for cutting tracks when I like to use condensers that have narrower directional patterns. If I'm doing an overdub I prefer to use an 87 that's pulled back a ways, and get some room sound. But the SM-81 has a beautiful sound and is really better doing tracks.

R-e/p: What do you use for vocals?

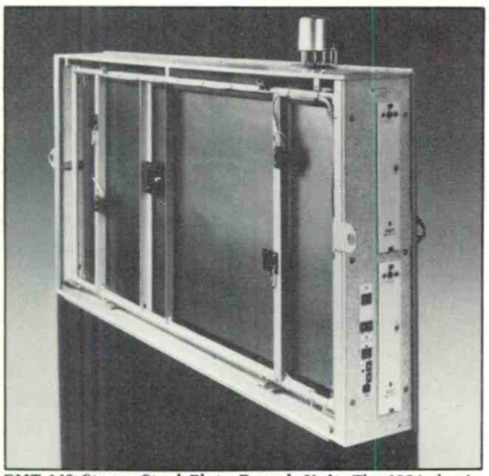
RD: We just bought an old tube-type

Typical Tracking/Rhythm Session Set-Up, Studio B, Scott-Sunstorm Studios, Los Angeles





But Antonio Stradivari would have understood.



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Right now, the *oldest* reverb unit in our line– the EMT 140– is priced higher than anyone else's *newest.* And our later models are even costlier.

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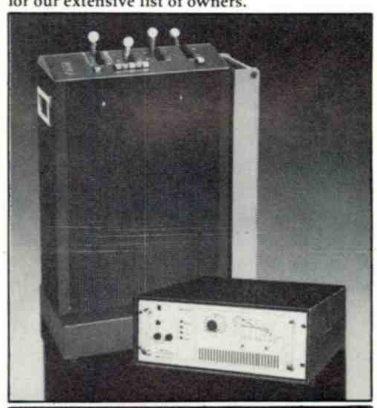

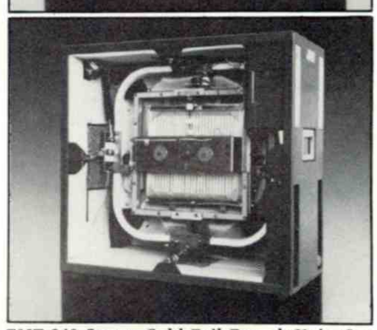

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GA-4

microphone. An old Neumann 48, which is just like a U47 except its pattern selection is different. The 47 can be either omni or cardioid, while the 48 is omni or bi-directional. We're using a lot of the tube-type microphones now. If I don't use that one I'll use a FET U47 microphone.

R-e/p: What do you think tube equipment in the audio industry seems to be making a comeback?
RD: It's the character of the sound. It's bigger and warmer.

R-e/p: Why do you think that?
RD: It's not a matter of thinking . . . it's a matter of hearing. I worked with Switch and was using an 87 and 67. The difference was like night and day. They refused to come into the studio without a 67 once we first used one.

R-e/p: Do you feel interest in tube gear is more widespread than just microphones?
RD: Yes, I use the La-2A. Again they have a warmer quality. It sounds better, and there's less to do.

R-e/p: When you're cutting basics, how do you lay out your track assignment?
RD: I have a standard way, as far as drums, bass and Rhodes. I try to do the Rhodes in stereo. I'll do a left and right drum, snare, kick, high-hat and bass. Left and right on the piano, left and right on the Rhodes, mike and direct on the guitar. If they want to ping-pong anything together later on, they can. All the tom-tom mikes are mixed in with the overhead to get the left and right field. Sometimes a producer will want the cymbals to be on separate left and right tracks.

R-e/p: Are you careful about working on edge tracks?
RD: As far as 1 and 24? Only on higher instruments like violins. I think the major machine companies and tape have worked out most of those edge tracking problems. Our 3M and Ampex's are good, and we take good care of them. But still, if the violins happen to be on the edge track, I might be a little concerned. I might put the cello rather than a violin on that end track. I wouldn't put a high-hat or cymbals on the edge either, but it's all from old habits.

R-e/p: What types of tape are you using?
RD: 3M and Ampex. 456 and 250, depending on what the client wants.

R-e/p: Does the client usually specify what sort of tape he wants?
RD: Some producers are aware of types of tape.

R-e/p: Have you found much difference between the two?
RD: In the sense of oxide. I remember when Ampex tape had a lot of oxide rubbing off. They had to fix it. Then 3M started to rub off. Now they are pretty much together. I do hear a difference due to the bias. It's much higher with the 250, and the sound seems to be cleaner as far as the top end is concerned. If it's okay with the producer, I like to have a choice between 250 and 456 depending on the tunes being recorded. If it's a jazzy tune I'll use the 456 because it's a rounder sound. If it's a really poppin' song I like the 250 for crispy cymbals . . . it's brighter.

R-e/p: Do you have trouble when you are overdubbing and moving from tape-to-tape?
RD: I'm usually adjusting and changing the machine myself. It doesn't bother me to do a little extra work for the sound. I usually have enough time between set ups to do that, and it only takes a couple of seconds. I only do it on the tracks I'm recording. If you had to re-align all 24-tracks it would be a problem, but I usually do it with only one track. I usually try to leave enough tape on either end of the reel so I can do a test record for bias.

R-e/p: Do you use any noise reduction?
RD: No, not any. To me noise reduction takes away the natural sound. Dolby seems to take the edge off the sound. It sounds mushy. It's not natural enough for me. I don't even like to think about dbx. It's not a natural sound either; it's too silent. I prefer to just record at 30 ips. Our machines have been gone through so there is no low end or top end deficiency.

'Piano Bag' manufactured by Alpha Acoustic Equipment, Los Angeles

R-e/p: Do you think it was just a matter of the noise reduction not being properly aligned?
RD: No, it's just the way they sound. I've used them when the console at the studio I was using didn't have enough headroom to put out the level I normally feed to the machine. If I'm working on a console that can handle the output, I'll just give the machine more level. I don't need any noise reduction. As far as sound is concerned, I feel anything you do to it, is taking something away from it.

I prefer to shock the tape and hit it really hard. I might take it 3 to 6dB above standard level on some things. I'll push the tape to its limit to get the sound on good and I will pin the meters at times.

R-e/p: You mentioned earlier that you seldom used a limiter/compressor. When do you like one?
RD: When I'm mixing down I'll use one on the bass, sometimes on the kick to get the punch, and on vocals if needed. When I'm doing vocals, if the singer doesn't have good mike technique, I'll use a dbx 160. It's very smooth, you can't even tell it's in the circuit.

R-e/p (Tom Lubin): What sort of speakers do you prefer?
Reggie Dozier: I start with Auratones. I'll listen to big speakers in some studios, but if they have Auratones, I'll go back to them to see where I'm at. Then I'll go play the big ones and go by feel. For big speakers I like the UREI Time-Aligned™, and the Augspurger speaker systems. But after working twelve-to-fourteen hours with the Augspurger system it gets fatiguing. It wears you down in a sense. I think that's because you can get them so loud. I can stay on the Time-Aligned™ much longer, because they only go to a certain loudness anyway.

R-e/p: Board and EQ. Scott/Sunstorm studios has an elaborate board and a simple one, both are excellent. How much do you think has to be included in a board?
RD: The DeMedio is simple and straight ahead. I would prefer to work a board like that because there's enough there to make you feel that you're doing some work. It's not quite as boring as just sitting there for a long time. You have more to do. With a fully automated board I get sort of lazy. You feel like the console is doing it all for you. Frank built this board in the early 70's, and it is a very musical console. We recently updated some of its functions. The amplifiers, VCAs, and transformers were replaced with faster circuits. But essentially, it's the same good "clean" sounding board it's always been.

You have one Harrison also?
RD: Yes. We have it set up for a computer. But personally, I feel it more when I do it myself. I don't want the computer to feel it for me, it just doesn't feel natural. I like to

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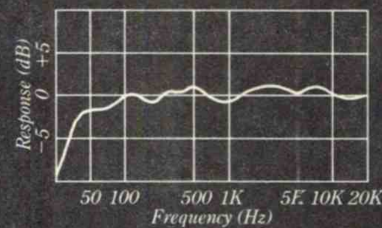

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FIRST WITH THE PROS.

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close my eyes and feel where I'm at with the music. However, I will use the memory for automating the muting, but not for controlling levels up and down.

Now they have just about everything you can think of on them. It's nice to have all those conveniences, but it's not necessary for all projects. For me, having outboard gear is enough.

R-e/p: What's your favorite outboard gear?
RD: The dbx 160. It keeps a nice even level. The 1176's are okay, but they take longer to set up. I like to work fast. With the LA-3A's we've modified ours by taking the transformer out and doing a few other things. I like the Lexicon 224, EMT 250, and a few different types of echo and definitely Kepex's.

R-e/p: How do you structure your echo?
RD: It depends on what I'm mixing. If it's a ballad I'll mix the strings with live echo. Our live chambers are pretty good. You see, these studios were built quite some time after the office building which faces the street. Where the studios are now located was a back yard with a swimming pool. When the studios were constructed they were built as an extension of the front building's second floor. The pool was drained but not removed, they just built above it, and it's become a great echo chamber. We also have three more very good chambers.

Vocals and some parts of the tracks I'll put through an EMT plate. If I have the 224 I might use it to enhance the strings or horns. And I often use a Cooper Time Cube.

MODIFYING THE LA-3A

by **David Baskin**

When the LA-3A was designed, use in older traditional facilities was the original intended purpose and it was therefore set up for 600-ohm termination standards. While this made installation easier in older systems, it created problems for users in the newer bridging systems and forced its designers to make trade-offs in its sound quality.

The primary trade-off is the input transformer because it exhibits considerable transient distortion.

A secondary trade-off is in use of C14 and R44 in the feedback network of the booster amplifier. These components are used to boost the high end of the system to compensate for high frequency loss in the input transformer. This practice, however, causes additional transient distortion in the unit.

In addition to the removal of the input transformer, C14 and R44, the reader may also wish the convenience of having the Compress/Limit switch on the front panel. The procedure for all of the above is covered below.

Improving LA-3A Performance

A - Conversion to Unbalanced Bridging Input:

1 - Remove T1 (input transformer).

2 - Remove R41 (510 ohms), R42 (510 ohms), and R43 (130 ohms) on rear gain select switch.

3 - Connect "IN+/-" to point on circuit board where screen wire of T1 was connected via red or white wire of a 2-conductor shielded cable.

4 - Connect "IN C" to point on circuit board where T1's blue wire was connected via black wire of the 2-conductor shielded cable.

5 - Connect shield to "CHASSIS" terminal.

B - Feedback Network Correction:

1 - Remove C14 (220 pf) and R44 (15 Kohm) on circuit board.

2 - Add 10 pf ceramic capacitor in parallel with R14 (220 Kohm) on circuit board.

C - Moving Compress/Limit Switch to Front Panel:

1 - Defeat power switch (S1) by disconnecting leads from switch and connecting them together (with appropriate insulation).

2 - Replace cable connected from "COMPRESS/LIMIT" switch to points D, E, and L on circuit board with longer 2-conductor shielded line that will reach from D, E, and L to S1.

3 - Re-wire S1 to this cable.

Finally, the LA-3A needs to work into 600 ohms in order to not have transient overshoot. If it feeds a bridging input it should be terminated with a 600-ohm resistor. But if it sees a 600 ohm input the resistor should be removed since too low of a load resistance will result in an output with a rolled-off top end. □□□

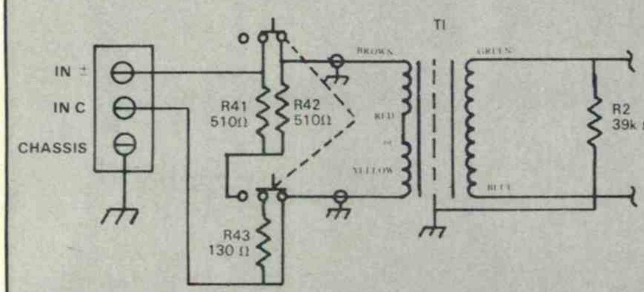


Figure 1: ORIGINAL INPUT CIRCUIT

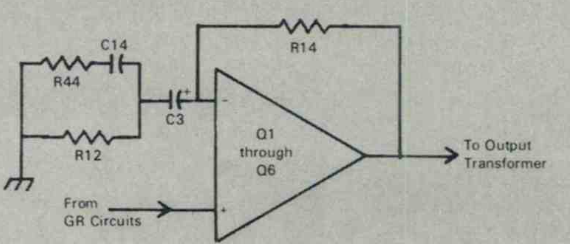


FIGURE 4: FEED BACK ELEMENTS
(Original Circuit)

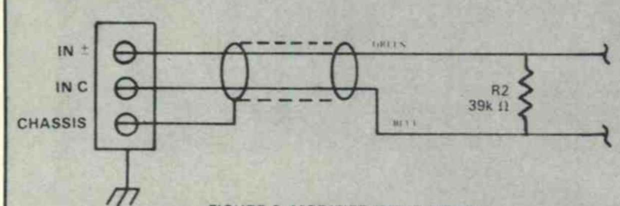


FIGURE 2: MODIFIED INPUT CIRCUIT
(Unbalanced)

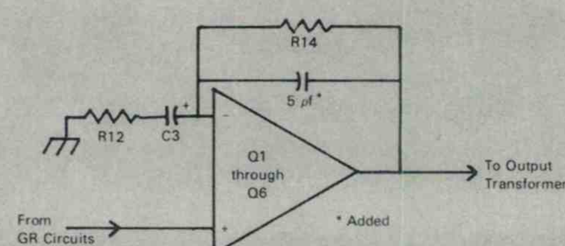


FIGURE 5: MODIFIED FEEDBACK ELEMENTS

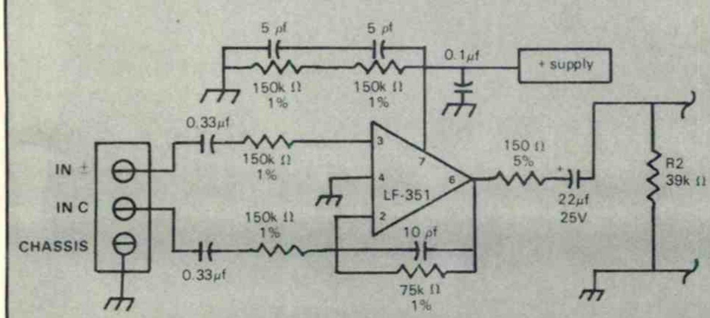


FIGURE 3:
BALANCED TRANSFORMERLESS INPUT CIRCUIT
(Optional)

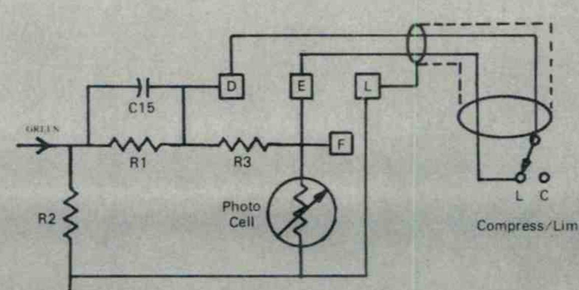
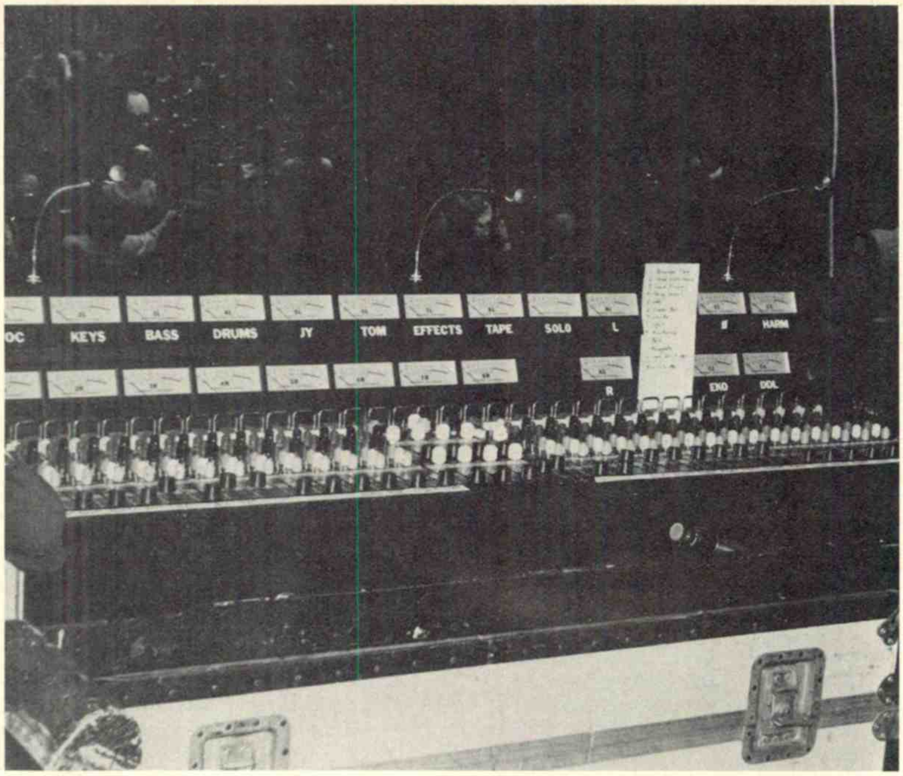
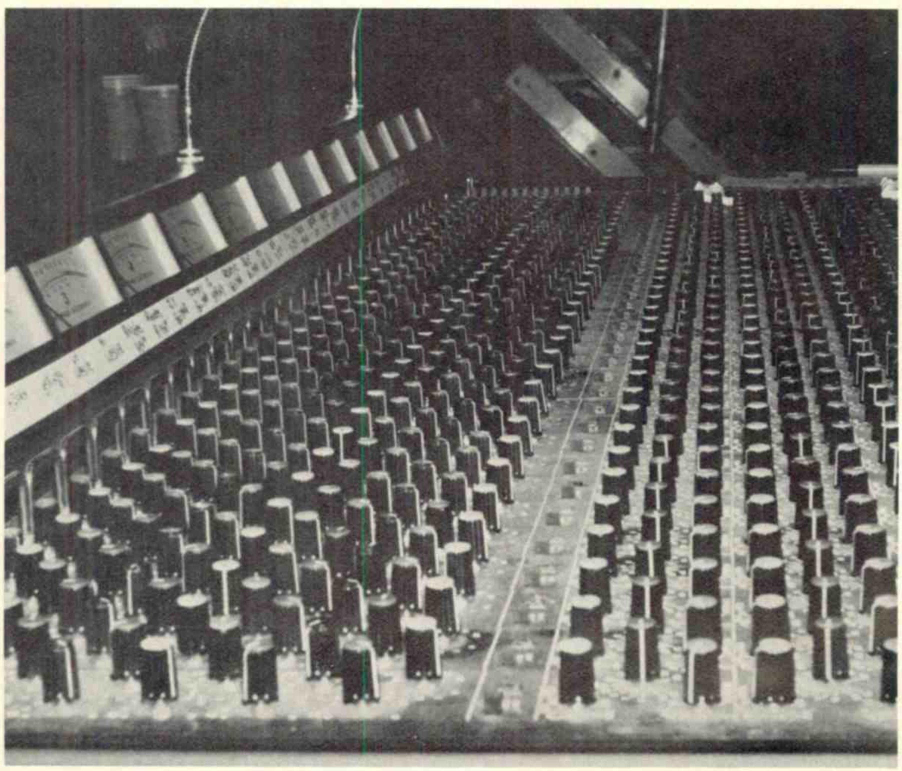


FIGURE 6: COMPRESS/LIMIT SWITCH



AMERICAN CONCERT SOUND DOES IT WITH INTERFACE



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ROCK CONCERT HOUSE MIXERS

Model 308-32X8S-32J/NS

"House Mixers" control 32 (or more) microphones and make up to eight stereo group submixes corresponding to up to eight performer groups, each with its own submaster and observable on VU meters. The NS module then makes a stereo house mix with a constant sum house panpot and a stereo slider house master, plus an operator's mix which is pushbutton selected from one or more of the submixes or any input solo. Inputs can even be soloed when off, to check before bringing them in. The type J module is standard and the type B module with parametric equalizers is optional. Either is also available with VCA, for VCA grouping. LED bargraph VU meters on every input are another option.

These simple rugged heavy duty mixers stand up well under difficult operating conditions to give long life on the road. Modular construction and plug-in IC's permit easy servicing. Large illuminated VU meters with 30000 hour lamps are visible under all conditions. Foam lined Anvil trunks are optional.

STAGE MONITOR MIXERS

Model 104-32X4A-32L/NS
312-32X12A32L/NS

Stage Monitor mixers make a number of mixes for feed back to a number of performer groups on stage, each of which wants a different mix. This requires a direct matrix and cannot be done with a standard pushbutton mixer. The 104L module has input pad, gain set switch, three equalizers, four position rolloff, LED overload danger indicator and solo, similar to the 104J, but the slider attenuator is replaced by eight color coded send pots, each with an eq in/out switch. The mixer thus makes a 32 X 8 pot matrix from inputs to outputs. Pot masters are standard, sliders optional. The NS operator's monitor permits the operator to listen to any mix or to any input solo.

These rugged, reliable Stage Monitors have become the standard of the industry and are used by many professional sound companies.

The 312L Stage Monitor is similar but adds three parametric equalizers, 12 send pots rather than 8, and a four inch slider attenuator to the input module, and is in a 308 frame. Slider Masters are an optional extra. The 312L system makes 12 output mixes.

for additional information circle no. 19

Reggie Dozier

They're pretty rare these days, but we have four of them. Cooper's Time Cube is an acoustic delay which is very good, but was only manufactured for a year or so. When digital delays first came on the scene the Cube's popularity quickly declined, but it continues to be a useful device to give a little separation and delay to guitars, strings, and sometimes vocals.

R-e/p: How do you treat echo for up-tempo tunes?

RD: I might use more of the plate EMT, the transistorized version. It has a little bit sharper echo . . . and it cuts through a bit cleaner, though I'll still EQ it some to make it a little broader. To enhance the whole thing more I'll run the signal into a Cooper before it goes into the EMT.

R-e/p: Do you find a great deal of difference between the transistorized and the tube EMT's?

RD: Yes, the transistorized ones are somewhat sharper in sound. The tube ones are a little bit warmer. I tend to prefer them on strings.

If I'm doing an album and the tunes are being mixed straight down the line, I'll still normal the board before starting the next song. I like for each mix to feel as though it were done at a different place and time.

R-e/p: Do you think that digital recording is really around the corner?

RD: I've seen it done, I've studied it. I think the editing process is not perfected enough having to use two machines. The sound is not quite as snappy. There is virtually no hiss, but I think it's too expensive to be around the corner. For one thing, the record companies are really squeezing, they want to produce product as cheaply as possible. Using standard recording machines is the least expensive. Personally, I like the way an analog recording sounds.

R-e/p: When you're dealing with a client's gear and he has a buzz problem, or you have a ground loop through a direct box, what will you try?

RD: If I have a 120-120 volt isolation transformer I'll try connecting the amp's AC through that. If it remains I'll isolate the AC ground of the amp with a U-ground plug. If that doesn't work, I tell him he has a bad instrument and to get another. Of course, I try everything I can before asking the person to do that. I've also discovered that the guitar cords with plastic sleeved connectors seem to generate less hum than the more durable metal ones.

R-e/p: What do you think the maintenance man of the future will have to be?

RD: Damn near genius. (Laughs)

R-e/p: You entered this business through maintenance.

R D: Yes, I started in 1960 just before I got out of high school. I was interested in electronics, and was working on TV's in my basement. I got into it without a formal education. I just picked it up. I forgot to mention that way back in 1957 to 1958 I used to go to Anna Records, which existed when Motown got started. I used to hang out there all the time. That was my first introduction to recording. I liked it, but my prime interest was in electronics.

When I went into the service in 1961, I was going to take electronics, but at the time I decided to take drafting which I did for two years. Unfortunately there wasn't time for me to get an education. I was in a special group that was constantly in training for combat defense, so I dropped out of school. In 1965 when I got discharged, I went to Ford Motor Company and they gave me a job on the crankshaft line. For three months I did that. The last month the company started watching me, so I was checking my work real close to make sure I was doing everything right. They decided to put me into a school pre-apprenticeship program for tool and die since I had already done a lot of drafting and machine shop work.

I was supposed to go back to Ford after pre-apprenticeship school, but Excello Corporation called me to come and work for them as an electrical discharge machining operator where I stayed for three years. During that same period I went to night school at the Electronics Institute of Technology, in Detroit. I worked from 7 a.m. to 4 p.m. Then be at school from 5 p.m. to 10 p.m. five nights a week. I did that for four years. A year before I got out of school I went to RCA because I wanted to get into television. I worked in TV for a year-and-a-half when my brother said, "Why don't you come to work for Holland Dozier Holland Studios and do maintenance?" HDH was formed after they left Motown. So I went there.

R-e/p: How big a studio was that?

RD: It was one studio with a mastering facility. They had two 16's that could be interlocked. We used an RCA Unilok system to lock them together. We would fill up the first fifteen tracks, then move that tape onto the slave machine, playback only. Then we would have 15 more tracks to record on for a total of 30.

R-e/p: Did the system use some sort of time code?

RD: Yes, the Unilok controlled the motors and kept them running together, with a 60 cycle tone. You would have to physically cue ad start them together. After the initial "locking in" they were okay. Occasionally when the sync was lost, you would have to go through the whole process again.

I did maintenance for about two years at HDH, and at the same time I was doing a few sessions with Lamont, a little recording in the evening. Maintenance during the day, and at night I'd be on the console doing some mixing or something. It was there, but it wasn't what I was totally interested in getting into. Then along came this guy named Richard Wiley, we called him "Popcorn," who told me the rough mixes I did were better than some master mixes he had from someone else. He asked me why I didn't just go into mixing? I told him I had never really thought about it and that I wasn't sure that I wanted to. It seemed like there wasn't enough involvement for me. With a maintenance job there was always something to do. I'm the type of person who's always gotta keep going. But in 1972 I decided to give mixing a real try.

About six months after that, Lamont decided to leave HDH, and ABC picked him up as a solo act. He got a deal that specified that he could bring his own staff, which included me. We came to Hollywood in the latter part of 1972; early 1973. I worked as an independent at first, but when we finished his album, ABC Studios hired me as a mixer.

R-e/p: So you weren't doing maintenance any longer?

RD: Not then. When I first came out to Los Angeles ABC only had the one studio. When they put in more studios, I began to help do maintenance and wiring to the console. Frank DeMedio handled the consoles. I helped with everything that had to do with getting to and from the console. Since the other two rooms started going I've done some occasional maintenance if something happens.

When MCA bought it in 1979, ABC gave me the opportunity to stay with them or go to MCA. After a month I decided to go out on my own because I found out I could make more money. I worked a lot with Fantasy Records, Motown, and Marvin Gaye. I did two albums with Switch, and was at Caribou for two months. I worked with Martha Reeves, Lamont did another album, Wayne Henderson . . . all as an independent. All-in-all over the course of about six months, after leaving ABC, I worked on about fifteen albums.

— continued on page 45 . . .

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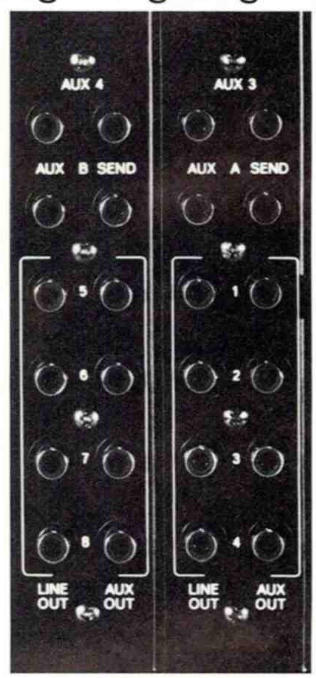

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". . . I feel that I'm only as strong as the maintenance we have . . ."

R-e/p: That's really banging them out!
RD: Don't ask me how I did it, I was flying from one studio to the next, work eight hours at one place and then on to the next.

R-e/p: When this building became Scott/Sunstorm they got in touch with you?
RD: When the Scott family bought the facility I was contacted since I knew more about the rooms at the time than anyone else. We went through the place and updated. Studio A was completely redone with TAD speakers and a new Harrison console. B has also been acoustically remodeled with TADs, Studer amp, and UREI 39 Room EQ. The DeMedio console is now fitted with Jenson transformers and some circuitry updates.

R-e/p: When you were taking classes did you take anything besides electronics?
RD: Yes, I had two years of business and two of accounting. When I got out of electronics school Lamont told me he wanted to start his own business and that he wanted me to be involved in it. So I went back to school and took business and accounting at the same time I was working. What I learned in those years has really come in handy as far as business deals go. I think if you have plans of being in a management position, a few courses in business are very useful. My mother always told me that knowledge will never hurt you, and to get as much as you can, no matter what you're doing. And I'd say in the studio it's 50% business and 50% of knowing what you're doing musically. One end needs the other to really make it.

R-e/p: Do you think a four year degree in electronics will be a requirement?
RD: Possibly, both are necessary but many maintenance engineers base their knowledge on experience. They are very good, yet their educational background is not that impressive. With a lot of people it's practical more so than academic.

R-e/p: What sort of qualifications are you looking for when you hire second engineers?
RD: If I see a resume that has a practical electronics background I'll go for that person faster than I would a person who is straight out of a school and mostly interested in mixing. I came up the hard way. I went inside the console and came out of it. Such a person would have a better understanding of how it's done.

R-e/p: If a band is coming in to do a demo and you have a chance to talk to them before hand, what do you ask them?
RD: First, I find out what kind of music they are doing. I find out what type of sound they want, how many instruments, and what kind. I'll also ask if they want a limited, tight sound, or live sound so I can mike accordingly when I first set up.

There's usually a lot of hums so I'll take care of that with isolators or transformers. If there's a problem with tubes or batteries we replace them, but it would be good if they did that before coming to the studio. A lot of musicians seem to feel batteries and tubes are thrown in free with the session, and I guess they are since they never get charged for them.

R-e/p: Why do you think most major labels have gotten out of being studio operators?
RD: Probably because a lot of the groups preferred to be somewhere else. They didn't like having someone constantly checking on their project. They didn't feel as free to be creative. When ABC closed their studios they were busy, but with acts on other labels.

R-e/p: You mentioned creative environments. How do you think a studio should feel?
RD: It should look like a living room or den. All the comforts of home so they can relax. It shouldn't feel like you are walking into an office building. It shouldn't feel like going to work. There can be wine or something to eat . . . big, comfortable chairs.

R-e/p: Do you think there's a problem with making it too cushy?
RD: If they choose to sit around . . . it's their money.

R-e/p: How do you go about buying a new piece of gear?
RD: I look in magazines, go to the shows and see what they are using in other studios. I make some calls, try to get a loaner, bring it in and try it out. All the guys here go through it to see what their opinions are. I want to know what our engineers will feel comfortable with, and then we arrive at a concensus.

R-e/p: Will you respond when a client says they want this or that?
RD: No. Some of the things are just not feasible to buy. We might use it two or three times, so we'll rent it.

R-e/p: You recently visited the Los Angeles AES Convention. What did you see that you thought was of particular interest?
RD: There was a foam material called Sonex Sound Control that has been designed for acoustically treating rooms. It surprised me; the thickness and the way it was designed. The shape was interesting. It looks like miniature anechoic wedges. It made the hotel room very dead. It comes in three different thicknesses: 2", 3", and 4", and in 4' x 4' sheets or 1' x 1' blocks. For their thickness they are surprisingly absorptive even at lower frequencies.

There's always a problem with available space when you're trying to acoustically modify a room. You always need more trapping. With this you can use a lot less space to achieve results when are comparable to thicker absorption systems.

R-e/p: I saw it too. It seemed to be fire resistant as well.
RD: They said they spray it with a protective coating. Recently I've seen it used in LEDE rooms to achieve the dead end.

At the show I also got a cassette from Agfa. I used it, and the sound was great. I've had problems in the past with that tape, the oxide rubbing off, the back coating rubbing off, and it's slightly abrasive. But I really loved the sound of it. They seemed to have solved their previous problems.

I was also impressed with the ATR-124. It's a beautiful machine. I liked everything about it. It looks good, I like the meter lights going into red when you punch into record. This machine has no drift at all when punching. They've got it down to the microsecond. They also have a programmable simultaneous re-biasing for any one or all tracks, which is sold as an option. Something like that will really save a lot of time re-adjusting the machine. In less than a minute all the tracks are re-biased to whatever you tell it.

R-e/p: Is there anything you would like to add in closing?
RD: I feel that I'm only as strong as the maintenance we have. I've tried to make this a family thing. I want everybody to feel comfortable so we can all work together at Scott/Sunstorm Recording Studios.□ □ □

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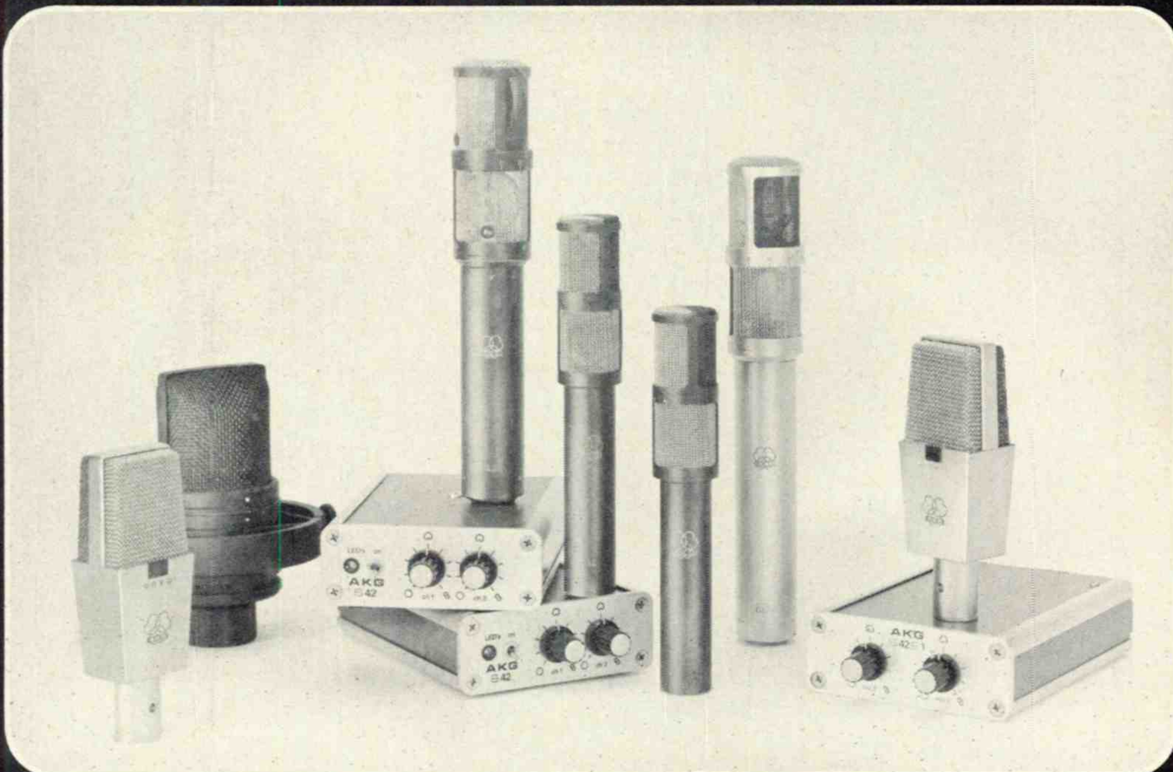

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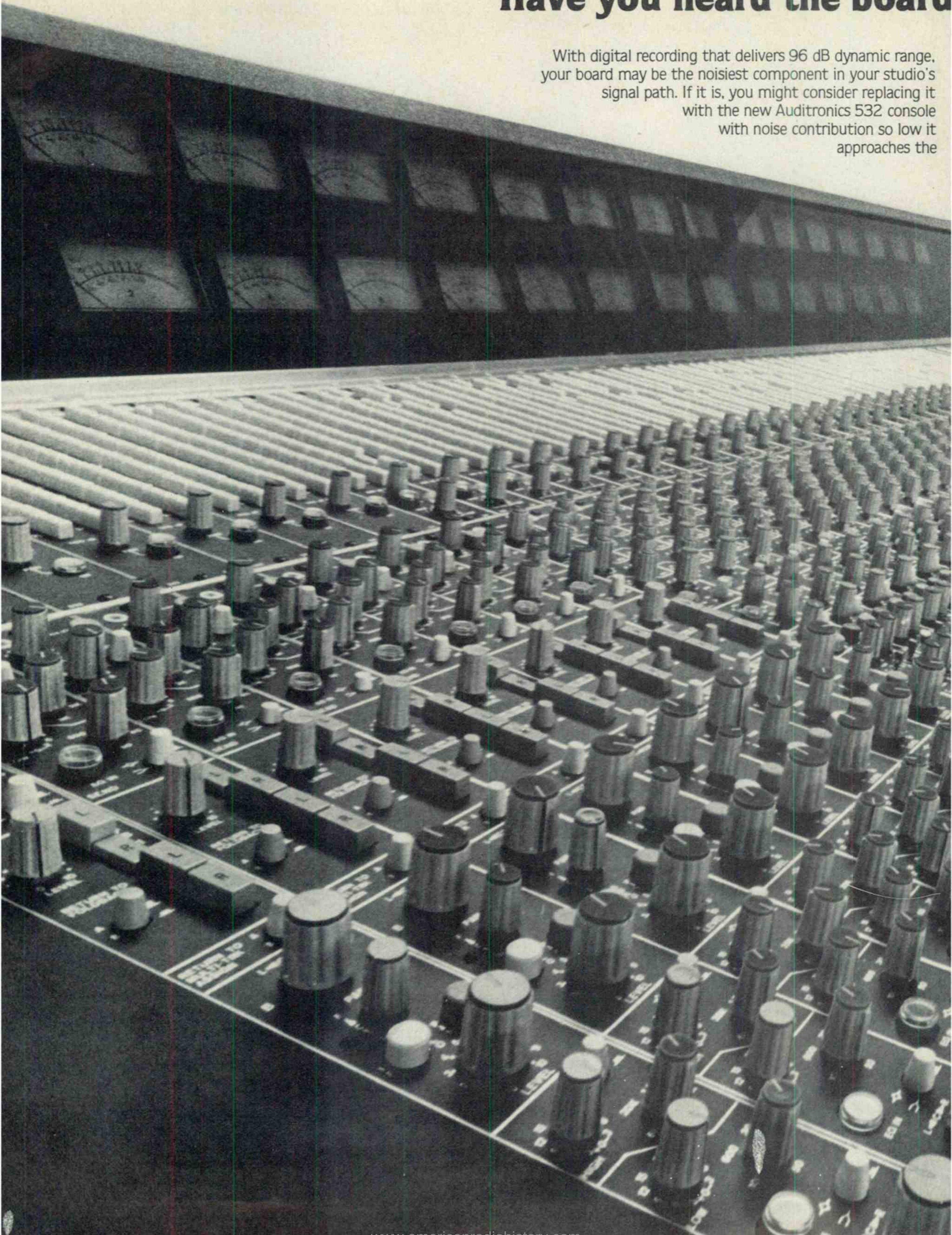

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Quiet Air for the Studio

by
F. Alton Everest

The background noise levels in recording studios, control rooms, and listening rooms must be kept under control if these rooms are to be of maximum use in their intended way. Hums, buzzes, rumbles, aircraft noises, auto horns tooting, dogs barking, or typewriter sounds are most incongruous if audible during a lull in the program. Such sounds might not be noticed outside the studio when they are a natural part of the situation, but during a pause or a quiet musical or speech passage they stand out like the proverbial sore pollex.

In a studio interfering sounds can come from control room monitors operating at high level or from equipment in adjacent areas. Control rooms have their own noise problems, some intruding from the outside, some generated by recorders, reverberation plates, equipment cooling fans, etc. There is one source of noise, however, that is common to all sound sensitive rooms and that is the noise coming from the air conditioning diffusers or grilles, the subject of this article. A certain feeling of helplessness in approaching air conditioning noise problems is widespread and quite understandable. Let's try to dispel some of the mystery surrounding the subject.

The control of air conditioning noise can be expensive. A noise specification in an air conditioning contract for a new structure can escalate the price. Alterations of an existing air conditioning system to correct high noise levels can also be expensive. It is highly desirable for studio designers to have a basic understanding of potential noise problems in air conditioning systems so that adequate control can be exercised during planning stages and supervision during installation. This applies equally to the most ambitious studio and to the budget job.

Selection of Noise Criterion

The single most important decision having to do with background noise in studios is the selection of the noise level goal to shoot for. The almost universal approach to this is embodied in the family of Noise Criteria (NC) contours[1] of Figure l. The selection of one of these contours establishes the goal of maximum allowable sound pressure level in each octave band. Putting the noise goal in this form makes it easily checkable by measurements. The downward slope of these contours reflects both the lower sensitivity of the human ear at low frequencies and the fact that most noises having distributed energy drop off in a similar way with frequency. To determine whether the noise in a given room meets the contour goal selected, the sound pressure level readings are made in each octave and plotted on the graph of Figure 1. The black dots represent such a set of measurements made with a sound level meter equipped with octave filters. The level at 1 kHz determines that a convenient single number NC-20 rating applies to this particular noise. If the NC-20 contour had been

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Photo of SONEX in recording studio courtesy of Alpha Audio.

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QUIET AIR FOR THE STUDIO

by
F. Alton Everest

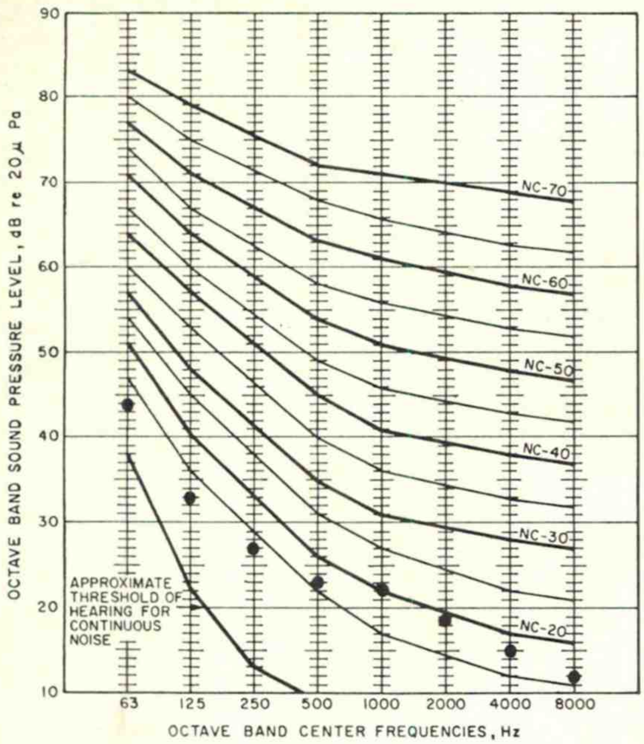


Figure 1: Noise Criterion (NC) Curves to be used in specifying the maximum permissible sound pressure level in each octave band. Measurements shown by the black spots would indicate that NC-20 contour would apply in rating that sound. *(Beranek, Reference 1)*

specified as the highest permissable sound pressure levels in an air conditioning contract, the above installation would just barely be acceptable.

Which contour should be selected as a limit of allowable background noise in a recording studio? This depends on the general studio quality level to be maintained, on the use of the studio, and other factors. There is little point in demanding NC-15 from the air conditioning system when intrusion of traffic or other noise is higher than this. In general, NC-20 should be the highest contour that should be considered for a recording studio or listening room and NC-15 is suggested as a practical and attainable design goal for the average studio. NC-10 would be excellent and it probably would take special effort and expense to reduce all noise to this level. If one feels the urge to calibrate the senses as to just how an NC-15 or NC-20 noise really sounds, the following is suggested. Beg, borrow, buy, or rent a sound level meter with built in octave filters. Measure the sound levels in several studios which you suspect have high noise or studios you consider quite acceptable. By the time you have measured four or five studios with A/C turned on and off, you will have little NC numbers in your head for ready recall in future discussions and you, too, will have become an expert.

ASHRAE

Perhaps it sounds like an Egyptian goddess, but our new word for today is ASHRAE (as in ashtray). It stands for the American Society of Heating, Refrigerating, and Air-conditioning Engineers. Although its primary purpose is to enlighten and standardize its own engineers, this highly respectable organization is a prolific source of helps for the studio designer who is a novice in A/C. In this brief article it is impractical to go deeply into specific design techniques, but ASHRAE Handbook and Product Directory - 1976 - SYSTEMS[2] has its entire chapter 35 devoted to step by step procedures in sound and vibration control. For those a bit shaky in acoustical fundamentals (e.g., how do you convert sound power to sound pressure?) a companion volume ASHRAE Handbook and Product Directory - 1977 - FUNDAMENTALS[3], is a must. For those with some engineering training and a modicum of determination, these two volumes can provide a background for intelligent dealing with the highest caliber of air conditioning contractors. For the lower caliber type this background is indispensible for avoiding big and expensive mistakes.

Machinery Noise

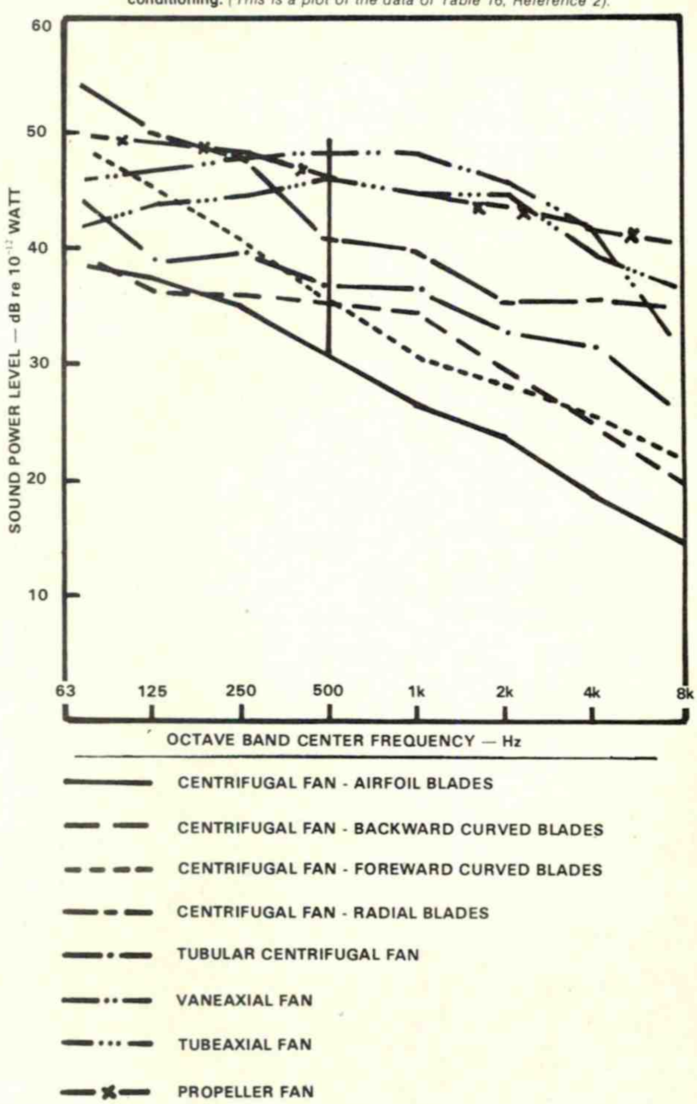
The first step in the reduction of A/C noise in the studio is wisely locating the A/C machinery. If this is left to chance, Murphy's Law will result in the equipment room being adjacent to or on the roof directly above the studio. The wall or roof panel vibrating like a giant diaphragm, is remarkably efficient in radiating airborne noise into the studio. So, step number one is to locate the equipment as far removed from the sound sensitive areas as possible, the next county if it can be arranged.

The next step would be the consideration of some form of isolation against structure borne vibration. If the equipment is to rest on a concrete slab shared with the studio, and plans are being drawn up, the machinery room slab should be isolated from the main floor slab. Compressed and treated glass fiber strips are available which are suitable to separate slabs during pouring. Other precautions would include proper vibration isolating mounts, designed accurately or they will be useless or downright damaging to the situation. Flexible joints in pipes and ducts as they leave the machinery room may be advisable.

Fan Noise

The fan is a chief, but by no means the only, contributor to A/C noise in the studio. The sound power output of the fan is largely fixed by the air volume and pressure required in the installation, but there are certainly variations between types of fan. In Figure 2 the sound power output vs. frequency is shown for eight different types of fans. The centrifugal fan with airfoil blades produces the lowest noise, but the octave centered on

Figure 2: Noise sound power output of different types of fans commonly used for air conditioning. *(This is a plot of the data of Table 16, Reference 2).*



by
F. Alton Everest

500 Hz has an exceptionally high level due, no doubt, to a strong single frequency component. Fans with less than 15 blades tend to generate pure tones which may dominate their output. The fundamental frequency of this tone is (rpm/60)(number of blades) and harmonics should be expected. Incidently, the graphs of Figure 2 are a plot of the data of Table 16 of Reference 2.

Most of the other fans of Figure 2 also have single frequency tonal components but they seem to be less dominant. Fan noise, aside from the fan tones, is distributed in nature, but falling off somewhat as frequency is increased. Noise is only one point to be considered in fan selection, but perusing the manufacturers' noise specifications should receive its rightful attention along with other factors. The 20 dB spread in sound power level emphasizes this point. In general, centrifugal fans produce less noise than axial flow fans.

Air Velocity

In air distribution systems the velocity of air flow is a very important factor in keeping A/C noise at a satisfactorily low level. Noise generated by air flow varies approximately as the 6th power of the velocity. As air velocity is doubled, the sound level will increase about 16 dB at the room outlet. Some authorities say that air flow noise varies as the 8th power of the velocity and give 20 dB as the figure associated with doubling or halving the air velocity.

A basic design parameter is the quantity of air the system is to deliver. There is a direct relationship between quantity of air, air velocity, and size of duct. The velocity of the air depends upon the cross sectional area of the duct. For example, if a system delivers 500 cu. ft./minute and the duct has 1 sq. ft. of cross sectional area, the velocity is 500 ft./minute. If the area is 2 sq. ft. the velocity is reduced to 250 ft./minute; if 0.5 sq. ft. velocity is increased to 1,000 ft./minute. Rettinger suggests[4] an air velocity of 500 ft./minute for broadcast studios and this value seems reasonable as a maximum for top flight recording studios. Specifying a low velocity eliminates many headaches later.

High pressure, high velocity, small duct systems are generally less expensive than low velocity systems. True budget systems almost guarantee noise problems in studios because of high air velocity and its resulting high noise. Compromise may be made by flaring out the ducts before the grille is reached. The increasing cross sectional area of the flare results in air velocity at the grile considerably lower than in the duct feeding it.

Effect of Terminal Devices

Even if fan and machinery noise are sufficiently attenuated by the time the air reaches the sound sensitive room, air turbulance associated with nearby 90° bends, dampers, grilles, and diffusers can be serious noise producers as suggested by Figure 3. Air flow in a duct can set duct walls to vibrating. Noise radiated from the duct walls can be a serious problem with exposed ducts running through a studio. Lagging the external surface with

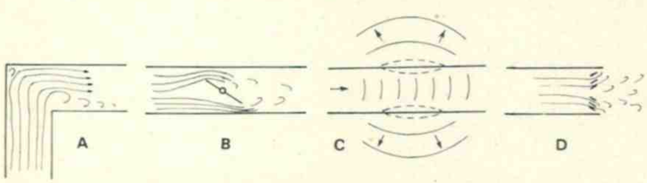


Figure 3: Air turbulance caused by discontinuities in the flow path can be a serious producer of noise in (A) 90° miter bend, (B) damper used to control quantity of air, (C) sound radiated from duct walls set into vibration by turbulance or noise inside the duct, and (D) grilles and diffusers.

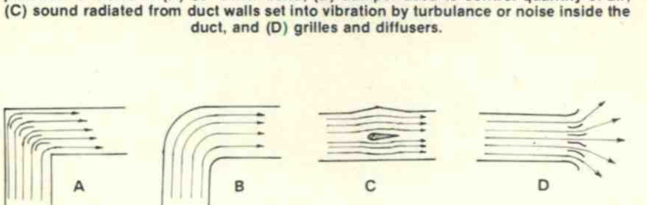


Figure 4: Air turbulance noise can be materially reduced by (A) use of deflectors, (B) radius bends, (C) airfoils, and (D) carefully shaped grilles and diffusers.

thermal type of material dampens such vibrations. Air turbulance noise can be reduced by carefuly designed vanes, deflectors, and airfoils as shown in Figure 4.

To emphasize the importance of such streamlining devices, here are a few numbers. Closing an ordinary damper to the 50% closed position increases the noise level approximately 15 dB. Figure 5 reveals the dramatic effect, both of air velocity and

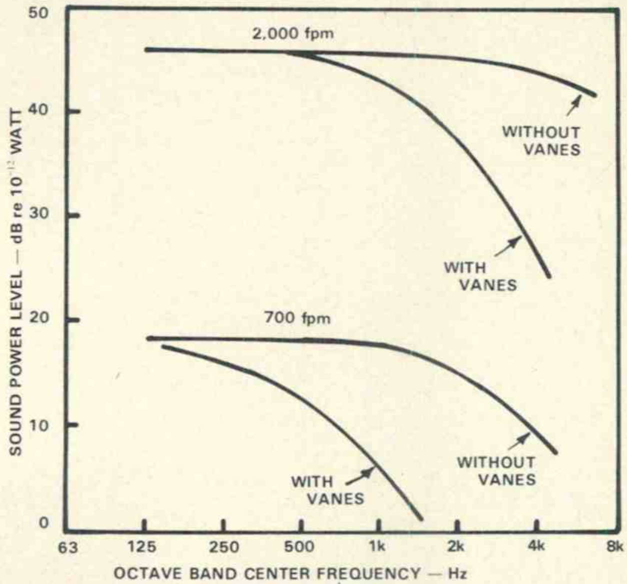


Figure 5: Noise produced bya 12" x 12" square cross section 90° miter elbow with and without deflector vanes and at 2,000 and 700 fpm air velocities. These curves have been calculated following the procedures in Reference 2.

shaped deflectors, in this 90° miter elbow. A 28 dB reduction of noise results from reducing air velocity from 2,000 to 700 fpm. The high frequency noise is greatly reduced at both air velocities by the use of shaped deflectors to guide the air around the right angle bend with a minimum of turbulance. These curves were calculated by following ASHRAE procedures in Reference 2.

"Natural" Attenuation

Care must be exercised to avoid overdesign of an air distribution system by neglecting certain attenuation effects built into the system. There is an attenuation resulting from the fact that ducts terminate flush with wall or ceiling. The duct acts

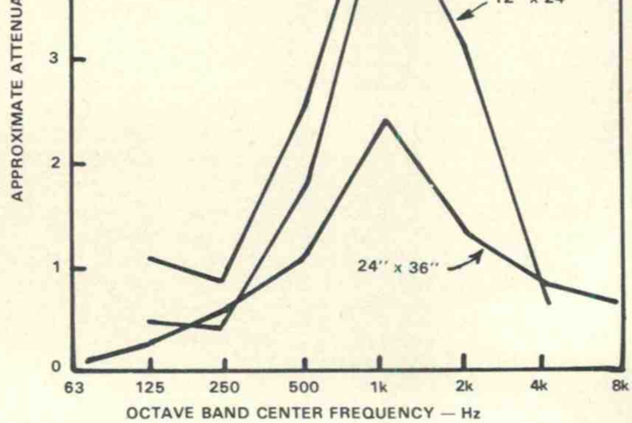


Figure 6: Attenuation offered by 1" duct liner on all four sides of a rectangular duct. Dimensions shown are the free area inside the duct, no air flow. This is a plot of data selected from Table 18, Reference 2.

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much like a radio frequency transmission line by reflecting energy back toward the source when a discontinuity in impedance is encountered. The higher air pressure in the duct encounters the lower air pressure of the room and acoustic energy is reflected back toward the source. This effect is greatest at low frequencies and for small ducts. At the 63 Hz octave, this loss amounts to 17 dB for a 5" duct according to Table 11 of Reference 2. A similar loss results at every branch or takeoff (Table 12). There is also an attenuation of noise in bare, rectangular, sheet metal ducts due to wall flexure amounting to 0.1 or 0.2 dB per foot at low frequencies (Table 13). Round elbows introduce an attenuation, especially at the higher frequencies (Table 14). All of these losses are built into the air handling system and serve to attenuate fan and other noise coming down the duct. Its there, its free, so take it into consideration to avoid overdesign.

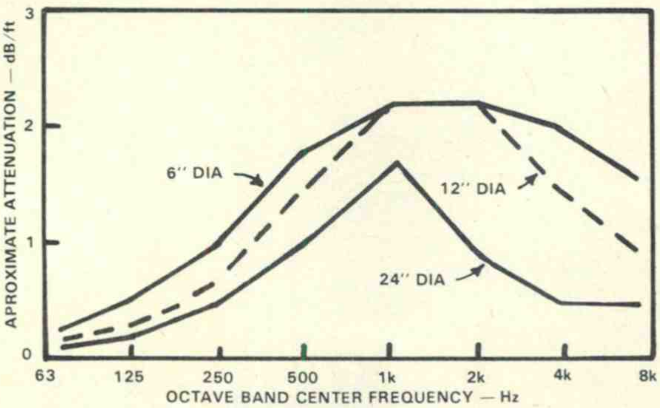

Figure 7: Attenuation offered by spiral wound round ducts with perforated spiral wound steel liner. Dimensions shown are the free area inside the duct, no air flow. This is a plot of data selected from Table 19, Reference 2.

Duct Lining

The application of sound absorbing materials to the inside surfaces of ducts is a standard method of reducing noise levels. Such lining comes in the form of rigid boards and blankets and in thicknesses of from ½" to 2". Such acoustical lining also serves as thermal insulation when it is required. The approximate attenuation offered by 1" duct lining in typical rectangular ducts depends on duct size as shown in Figure 6. The approximate attenuation of round ducts is given in Figure 7. Duct attenuation is much lower in the round ducts than in the lined rectangular ducts for comparable cross sectional areas.

Plenum Silencers

A sound absorbing plenum is an economical device for achieving significant attenuation. Figure 8 shows a modest sized plenum chamber which, if lined with 2" thickness of 3 lb./cu. ft. density glass fiber, will yield a maximum of about 21 dB attenuation. The attenuation characteristics of this plenum, calculated by Equation 3, page 35.13, Reference 2, are shown in Figure 9 for two thicknesses of lining. With a lining of 4" glass fiber board of the same density quite uniform absorption is obtained across the audible band. With 2" glass fiber board attenuation falls off below 500 Hz. It is apparent that the attenuation performance of a given plenum is determined primarily upon the lining.

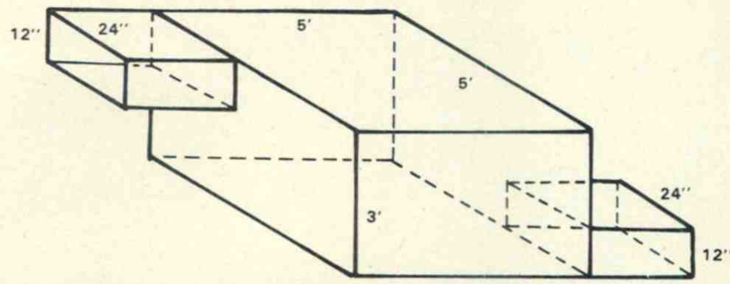

Figure 8: Dimensions of a plenum which will yield about 21 dB attenuation throughout much of the audible spectrum.

Figure 10 gives actual measurements on a practical lined plenum muffler approximately the same horizontal dimensions as that of Figure 8, but only half the height and with baffles within. Attenuation of 20 dB or more was realized in this practical case above the 250 Hz octave and it solved an otherwise intolerable noise problem.

Plenum performance may be increased by increasing the ratio of cross sectional area of the plenum to the cross sectional area of the entrance and exit ducts, and by increasing the amount or thickness of absorbent lining. A plenum located at the fan discharge can be an effective and economical way to decrease noise entering the duct system.

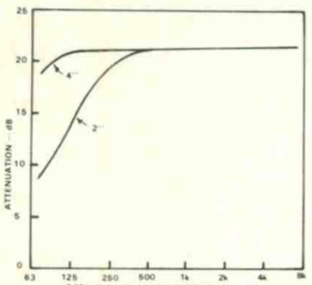

Figure 9: Calculated attenuation characteristics of plenum of Figure 8 lined with 2" and 4" glass fiber of 3 lb./cu. ft. density. Calculations followed ASHRAE procedures.

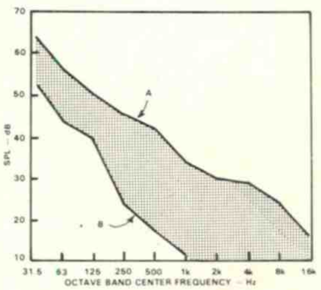

Figure 10: Measured noise sound pressure levels (A) at plenum input and (B) in room 5 ft. from supply grille. This plenum is approximately the size of that of Figure 8 but only half the height but it contained baffles.

Packaged Attenuators

Numerous proprietary packaged noise attenuators are available. Cross sections of several types are shown in Figure 11 with their performance plotted below. For comparison, the attenuation of our old friend the lined duct is given in curve A. Some of the other attenuators have no line of sight through them, i.e., the sound must undergo reflections from the absorbing material to traverse the unit and hence have somewhat greater attenuation. The absorbing material is usually

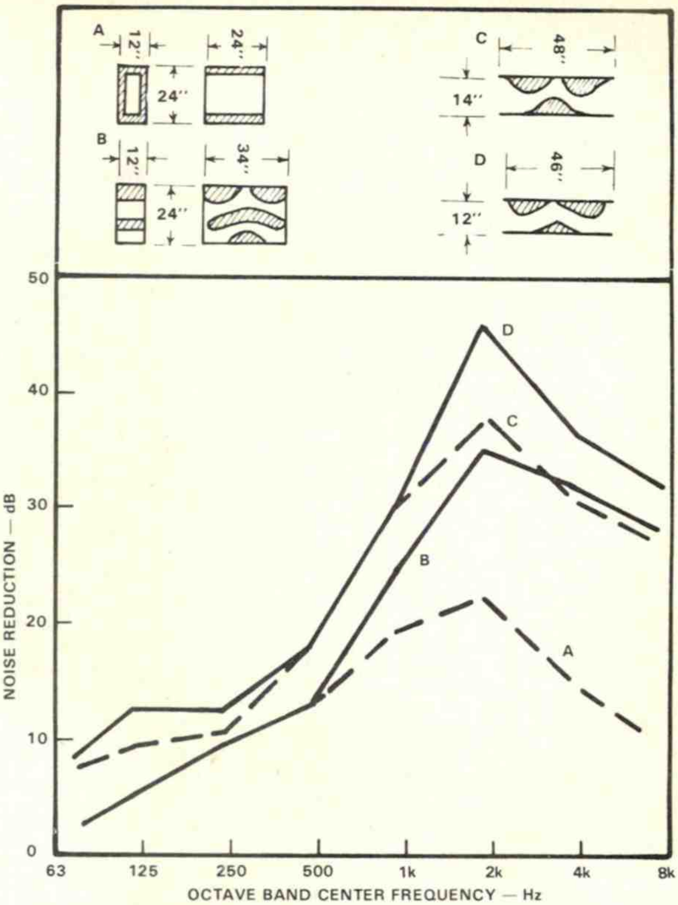

Figure 11: Attenuation characteristics of three packaged silencers compared to that of the lined duct, curve A. *(Adapted from Doelling, Reference 5.)*

protected by perforated metal sheets in these packaged silencers. The attenuation of such units is very high at midband speech frequencies, but not as good at low frequencies.

Reactive Silencers

Several passive, absorptive silencers have been considered which rely for their effectiveness on the changing of sound energy into heat in the interstices of fine glass fibers. Another effective principle used in silencers is that of the expansion chamber as shown in Figure 12. This type performs by reflecting sound energy back toward the source, cancelling some of the oncoming sound energy. As there is both an entrance and exit discontinuity, sound is reflected from two points. Of course, the destructive interference (attenuation peaks) alternates with constructive interference (attenuation nulls) down through the frequency band, the attenuation peaks becoming lower as frequency is increased. These peaks are not harmonically related, therefore they would not offer high attenuation to a noise fundamental and all its harmonics, but rather attenuate slices of the spectrum. However, by tuning, the major peak can knock out the fundamental while most of the harmonics, of much lower amplitude, would receive some attenuation. By putting two reactive silencers of this type in series and tuning one to fill in the nulls of the other, continuous attenuation can be realized throughout a wide frequency range. No acoustical material is required in this type of silencer which operates much like an automobile muffler.

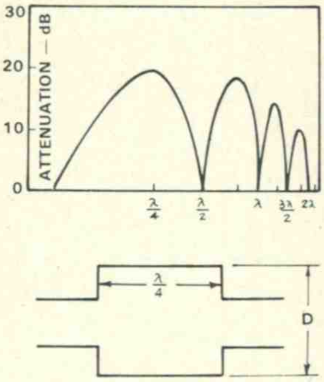

Figure 12: Attenuation characteristics of reactive type of silencer, the expansion chamber. Sound is attenuated by virtue of the energy reflected back toward the source, cancelling some of the oncoming sound. *(Adapted from Sanders, Reference 6.)*

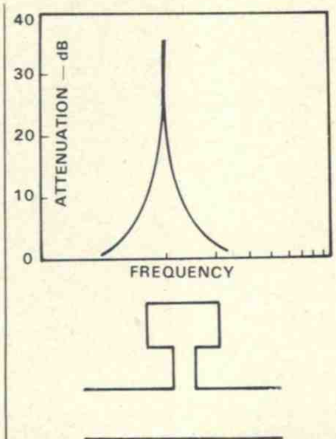

Figure 13: Attenuation characteristics of the tuned stub silencer. *(Adapted from Sanders, Reference 6.)*

Resonator Silencers

The resonator silencer, illustrated in Figure 13, is a tuned stub which provides high attenuation at a single frequency or narrow band of frequencies. Even a small unit of this type can produce

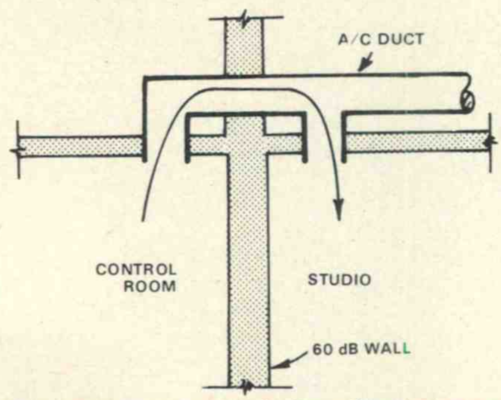

Figure 14: An expensive, high transmission loss wall can be by-passed by sound travelling over a short duct path from grille to grille.

40 to 60 dB attenuation. This type of silencer offers little constriction to air flow which may be a problem with other types.

Duct Location

Why build a 60 dB wall between studio and control room, for example, and then serve both rooms with the same supply and exhaust ducts with grilles closely spaced as in Figure 14? This is a tactical error which results in a short path speaking tube from one room to the other quite effectively nullifying the 60 dB wall. With an untrained air conditioning contractor doing the work without a supervisor sensitive to the acoustical problem, such errors can happen and do happen. To obtain as much as 60 dB attenuation in the duct system to match the construction of the wall requires application of many of the principles discussed above. Figure 15 suggests two approaches to the problem, (1) to separate grilles as far as possible if they are fed by the same duct, or, better yet, (2) to serve the two rooms with separate supply and exhaust ducts.

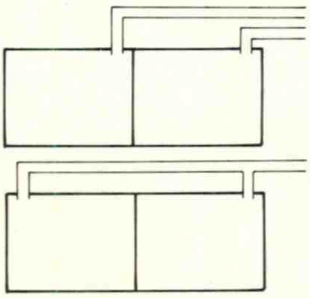

Figure 15: Two possible solutions to the problem of Figure 14, (1) separate grilles as far as possible, or (2) feed the room with separate ducts. Both approaches increase duct path length and thus duct attenuation.

Suggestions

1 - The most effective way of controlling flow noise is to size the ducts to avoid high velocities. The economy of the smaller ducts, however, may be more than enough to pay for silencers to bring the higher noise to tolerable levels.

2 - Right angle bends, dampers, etc. create noise due to air turbulance. Locating such fittings 5 to 10 diameters upstream from the outlet allows the turbulance to smooth out.

3 - Noise and turbulance inside a duct cause the duct walls to vibrate and radiate noise into surrounding areas. Rectangular ducts offend in this way more than round ones. Such noise increases with air velocity and duct size but may be controlled with an external treatment of thermal material.

4 - Acoustical ceilings are not good sound barriers and in a sound sensitive area the space above a lay-in ceiling should not be used for high velocity terminal units.

5 - The ear can detect sounds far below the prevailing NC contour noise (see Figure 1). The goal should be to reduce noise in the studio to a level at which it cannot be heard on a playback of a tape recorded at normal level and without noise reduction.

6 - Plenums are effective and straightforward devices adaptable to studio quieting programs and they offer attenuation throughout the audible spectrum. They are especially effective at the fan output.

7 - Some of the noise energy is concentrated in the highs, some in the lows. There must be an overall balance in the application of silencers so that the resulting studio noise follows roughly the proper NC contour. Otherwise overdesign can result. □ □ □

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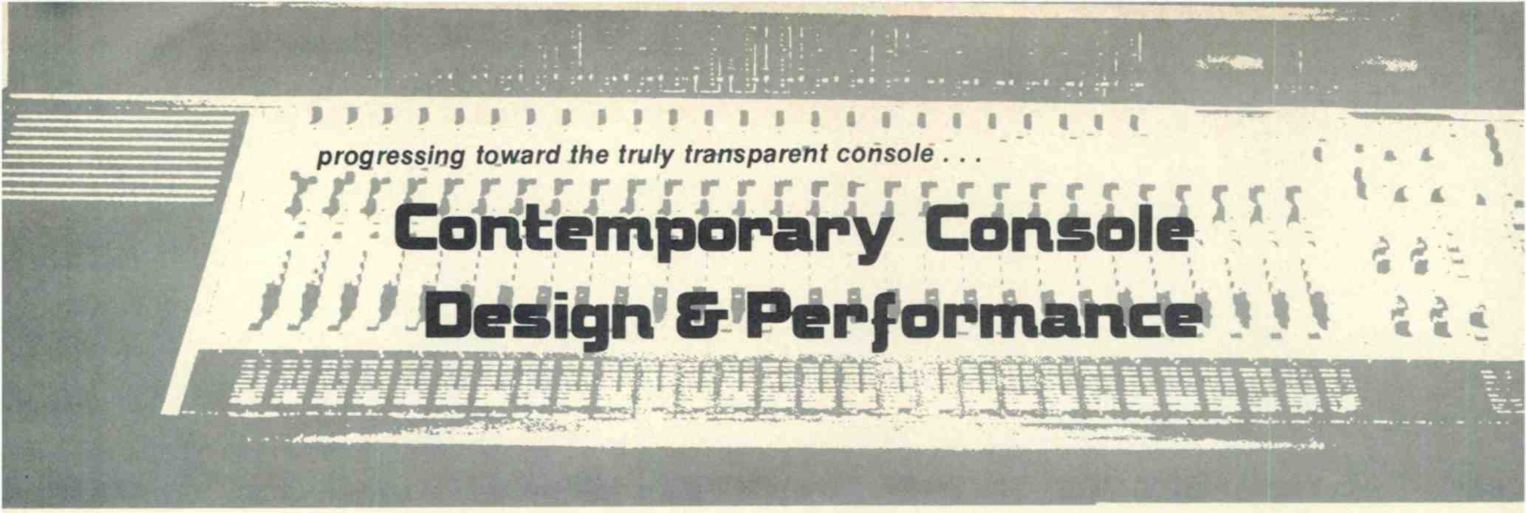

progressing toward the truly transparent console . . .

Contemporary Console Design & Performance

by Craig Connally

It is surprising to find in recent articles on recording console design that little mention is made of the real goal of designers: a console whose sound is as good as is presently possible, i.e., whose sound quality represents the state-of-the-art. In the alternative, emphasis is placed on specifications and their optimization with the implicit confidence that when it comes to measurements over sound the one inevitably leads to the other. Scant regard is shown for the wisdom among recording engineers that specifications give little clue as to how a device will sound and that similar devices, however far below audibility their distortions have been reduced, do not sound alike. Technologists pursue exotic test signals, complex computer programs, and an alphabet soup of "new" distortion mechanisms, all of which are expected to lead to formulae by which consoles and other linear audio circuits can best be designed.

Agnostic Pragmatism

There are design techniques, some examined in this article, which not only have techno-freak appeal but which, as a part of an analog design philosophy removed from the objective/subjective schism, have been conclusively shown to yield superior sounding consoles. Someday these points may be born out by a combination of measured linear and non-linear distortions, short term A/B comparisons, and long term listening tests, but for the present they are defended by the ultimate criterion: favorable evaluation by critical recording engineers and producers.

Consoles of older design suffer poor sonic quality in part because they are constrained to use amplifiers, integrated or otherwise, not well-suited to professional audio. New high-speed, low-noise parts now becoming available make the uncolored, transparent console a practical possibility. A trend toward this goal is becoming popular and may be expected to dominate in the next decade as analog audio design reaches maturity. New design techniques are necessary to exploit these components, but such techniques do not simply comprise a shopping list of features guaranteeing success. Good specifications are the consequence, not the goal, of good design.

Dance of the Seven Veils

In this article the term sonic quality will be a frequent reference. It means more than accurate reproduction, though that is a basic element. It also implies a sense of presence, aliveness, and clarity of sound, and the reality and stability of the stereo image. Many devices and circuits cause what subjectively appears to be a compression of the brilliance and dynamics of music, and freedom from this coloration is a part of sonic quality.

I am aware that many will find it difficult to relate to these ideas (the matter of measuring them with instruments for the moment set aside) for it seems absolutely necessary to listen at length to systems relatively free of strained, gritty, or dynamically dulled colorations before one comes to appreciate them with analytic distaste. Unfortunately, few professional systems are free of these defects; monitoring systems in particular, (with the exception, according to this author's experience, of the Meyer Sound Labs' speakers) are so colored and have such poor resolution that imperfections in electronics may be swamped out (as a recently published amplifier test suggests). Nevertheless, users of more transparent consoles report that some of the aspects of increased sonic quality remain apparent even to unsophisticated listeners when heard over AM radio. But ask yourself honestly: have you ever heard a system so accurate that a precocious six year old could not immediately tell that the music was not live? The natural sound of unamplified music should always be an engineer's standard, even if it is not his goal.

But It Sounds Great On The Printout

Circuit analysis is a handy technique for examining the measurements expected from a simple circuit without actually building it, but it is doomed to failure when applied to the design of good-sounding consoles. This is partly because its simplifying assumptions (all electric fields are within capacitors, all magnetic fields are within conductors, all wavelengths are longer than conductors, etc.) do not treat the normally insignificant factors which dirty-up sound in large systems like recording consoles. Besides, even with macromodeling of amplifiers, circuit analysis programs run out of nodes on a single input strip. On the other hand, boundary conditions make direct solution of the relevant Maxwellian equations impossibly complex even with large digital computers. It remains possible, in fact necessary, to do design work on certain modern computers — analog computers, i.e., the consoles themselves. Although this demands the building of an entire console to test the sonic consequences of a single design change, such costs must be born when maximum sound quality is the objective. Since the problem is so complex, it is not surprising that the matter of evaluation is also complex. Laboratory test methods and specifications become inadqequate for overall sonic judgements. What is necessary is the same precarious methodology used by consumer audio manufacturers: critical listening. This is the most common and most serious omission in professional audio design, but as skills are refined such procedures will become de rigueur. At last manufacturers will discover, if not admit, what their customers have known all along: consoles do not sound alike and few sound good at all.

". . . printed circuit board layout becomes a high art that has direct sonic consequences . . . two circuits with identical schematics and identical components can sound different . . . due only to different PC board layouts."

One Good Thing Leads To Another

From the innocent generalization that reducing distortions over a wider bandwidth seems to improve sound quality has grown the demand for faster op amps, the elimination of transformers, the application of radio frequency design methods to audio, and the more skilled control of linear distortion networks. Concomitant has been the awareness that out-of-band signals can cause in-band distortion products more noxious than simple harmonic distortion.

At low frequencies one confronts phase shifts, envelope distortion, and transformer saturation, among other ills. General solutions are to use modern op amps, eliminate transformers, and extend frequency response to well below 10 Hz. At high frequencies the problems become more acute.

- the author -

Craig Connally is president of NEOTEK, which manufactures a full line of recording, reinforcement, theater, and broadcast consoles.

After graduating from Antioch College, and earning a Masters Degree from the University of Illinois, he worked as a contract engineer in the areas of scientific instrumentation, biomedical design, and computer interface. NEOTEK was formed in 1972. Craig is a member of the AES and IEEE.

. . . continued overleaf -

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transparent consoles

As the output of an amplifier approaches a sufficiently high frequency and amplitude it reaches an internal or external limit on the rate at which it can supply current. The output voltage then ceases to be an exact representation of the input signal and will simply ramp (slew) at some maximum rate. When this happens the circuit is not under control of feedback, distortion rises sharply, and in-band intermodulation products are generated. To avoid these problems designers seek amplifiers and circuits that have wider power bandwidths and more linear transfer characteristics. This inevitably means eliminating transformers, though there are other equally compelling reasons to do so. Some console manufacturers, like Harrison, have eliminated input transformers; some, like Neve, have eliminated output transformers; a very few have eliminated transformers altogether, but the future is clear. Given the appropriate amplifiers and circuits, transformerless design is an important tool of the audio design engineer.

"good specs are the consequence . . . not the goal . . . of good design!"

Improved amplifiers also allow more sophisticated equalization networks designed around active filters. A fly in the ointment offered by better amplifiers is the necessity to re-think audio design in terms of available bandwidths in the tens of Megahertz, and this requires radio frequency techniques. Application of these many potential improvements means that it is now possible to measure the input to output performance of a relatively inexpensive console and achieve results that surpass state-of-the-art 18-bit digital systems. Such new designs can outperform old console designs and current 16-bit storage systems by a surprising degree in terms of noise, distortion, and sonic quality. Solid state design has taken twenty-five years to mature, monolithic amplifiers are younger than multitrack recording, so it is natural to expect that the quality standards for professional analog design are being pushed ahead every day.

Throw Away Your Textbooks — and Dive In

A beginning step in the art of console design is the selection of an amplifier or gain block. Knee-jerk phobia of monolithic amplifiers (ICs) is no longer sensible because increasing markets have drawn semiconductor manufacturers to offer more and better op amps for audio applications. A prime requirement of any choice would appear to be slew rate but critical listening tests have shown that speed is not the whole story. Slew-enhanced devices, like the infamous 741S, achieve speed with high input stage non-linearity at the expense of sonic quality. Devices with asymmetric slew rates also have been found to give poorer results than their speed would suggest. The popular LF356 slews almost twice as fast in the negative-going direction as in the positive-going, and so should be avoided unless better choices are unavailable. This behavior is not intrinsic to the BiFET process however, the LF347 and TL070 series op amps slew with reasonable symmetry. Although the differences may be too subtle for resolution on professional systems, it is this author's finding that feed-forward configurations, whether internal as in the NE5534 or external, are inferior to topologies which achieve their speed by other means, such as dielectric isolation in Harris parts or lower input stage transconductance in the BiFETs. Nevertheless, BiFET op amps are no panacea. For example, they may exhibit a voltage-dependent input impedance which can cause rectification of the input signal at high voltage peaks in circuits designed without this awareness and they are noisier than many bipolar parts.

Thoughtful designers will realize that for audio applications the textbook assertion that input stage distortion is most important is a red herring. The progressive decrease of input stage transconductance produces an S-shaped transfer characteristic and, like tape saturation, a relatively tolerable form of distortion readily reduced by feedback. In general, paranoia over simple harmonic distorion is probably silly. Even gross crossover distortion seems to have a short-term audibility threshold in music in excess of .1%, and low order harmonic distortion may actually be perceived as a euphonic enhancement to some programs. Playing a record with a pivoted tone arm must produce distortion well over 1%, but this effect alone is not objectionable. A designer should closely examine an amplifier's open loop transfer characteristic under various loads and at various frequencies to see what ills feedback may be hiding. Circuits that use an integrated amplifier in combination with discrete transistors to swing greater voltage or current outputs are especially prone to this source of nasty-sounding non-linearity. It is fortunate that IC manufacturers are becoming more willing to sacrifice power consumption specifications and bias output stages farther into AB operation, because this certainly improves sonics performance.

Another factor to consider when examining amplifiers is the input dynamic range. For a variety of reasons, an op amp's differential input voltage will generally not be zero; its ability to tolerate non-zero voltage will determine under what conditions the op amp will run into slew rate limitation problems. When the input dynamic range (equal to the product of peak output voltage and power bandwidth divided by gain bandwidth product) is exceeded, the amplifier will no longer be under control of feedback and it lets your ears know it. FET-input amplifiers are generally superior to bi-polars in this regard and are less sensitive to RF demodulation for related reasons. Input related problems can dirty-up an otherwise clean signal long before their gross manifestations become obvious in some specification figure.

Common mode rejection ratio and power supply rejection ratio, both functions of frequency, are other factors to consider, along with input noise, power bandwidth, power supply requirements, and cost. The bottom line is: designers' choices are limited by available commercial offerings. Recently this situation has greatly improved: the TL070 series op amps are becoming widely distributed and soon will be available worldwide; the NE5534 is now multiple sourced, although there are source-dependent differences; and the interesting new MA-332 offers the low noise of the 5534, nearly twice the output voltage into a 600-ohm load, less distortion, 60% greater slew rate, and is available in a low-cost plastic d.i.p. package. Perhaps the end of the discrete op amp is at hand.

Keeping the Horse Before the Cart

A designer selects amplifiers much like a painter selects pigments; the fruits of these efforts come in the application to specific designs. At this point a single principle has immense importance to the design of good-sounding equipment: no amplifier must ever be allowed to slew. If the slew rate of a console can be measured, the designer has made sonic concessions to specifications. Another way of stating this maxim is that the power bandwidth of every circuit must exceed its closed-loop small signal bandwidth. Slew induced distortion is known to cause highly objectionable intermodulation products that seriously degrade audio quality. This explains the requirement for amplifiers with high slew capability (about one volt per microsecond for each output volt, or about ten volts per microsecond minimum for professional levels) especially in recording consoles which get signals hot off the microphone before any other bandwidth reduction. It also suggests that consoles with unity gain structures will be more likely to be free of slew induced distortion in actual practice. It remains for the designer to utilize the slew capability of the devices he has chosen to produce a console with overall full power bandwidth comfortably beyond 20 kHz but which cannot be made to slew.

This capability can be examined by applying a very short rise time square wave to the inputs, mike, or line level, of a console and examining the out from each stage. An oscilloscope should reveal an output signal from the same duty cycle as the input, a smooth logarithmic rise and fall which has the same shape at all amplitudes, and total freedom from overshoot at all levels. In practice, there are few console which will survive this test while providing full power bandwidth beyond 20 kHz.

The networks which designers attach to amplifiers to control response at high frequencies are called compensation networks, and these play an important but incomplete part in determining the stability and sound quality of complete console systems as well as individual stages within consoles. Crafty designers will not depend on textbook solutions or IC manufacturers' suggestions when designing op amp compensation circuits; it is much more important to understand the internal workings of the amplifiers themselves and the causes of problems that compensation cures. For example, the NE5534 is generally considered to have a minimum stable gain of 10 dB without compensation and one often sees them compensated for unity gain (to wit: nearly every one in the MCI JH-600) with attendant reduction of slew capability to seven volts per microsecond. It is possible, however, to operate them in situations as extreme as unity gain followers or with inverting gains below .01 (–40 dB) with totally adequate phase margins and stability into reactive loads without decreasing their slew capability or using compensation capacitors.

". . . In general, paranoia over simple harmonic distortion is probably silly . . .low order harmonic distortion may actually be perceived as a euphonic enhancement to some programs."

The problem of compensation becomes especially important in master combining amplifiers. With a number of inputs connected, the stability problem focuses on the need to account for the stray capacitance of the distributed input bus. A well-designed console will have mute networks which remove muted inputs from the bus in order to reduce combining amp noise gain. When all inputs are muted the stability enhancing noise gain is minimized but most of the bus capacitance is likely to remain, a worst case condition especially if the combining amp drives a potentially reactive load. The trick is to insure absolute stability and freedom from slewing under all conditions without reducing slew capability or resorting to artifice like raising the high frequency noise gain. This problem illustrates the need for a thorough understanding of the internal design of whatever amplifiers are ultimately chosen.

DC To Daylight

Of foremost concern to the designer is the tremendous bandwidth available from the monolithic op amps becoming popular for audio design. These bandwidths imply that the power, ground, and signal distribution systems of a console must be conceived in terms of radio frequency signal processing. This does not mean that consoles must pass an NTSC carrier without color shift, but the availability of gain well beyond the Citizens Band poses new problems. If the design techniques used in most consoles were applied to radio frequency circuits, they simply would not work. Consoles so designed will pass audio signals and even yield acceptable traditional specifications, but critical users will notice a gritty sound quality, poor definition, image instability particularly with massed strings or horns, muffled "transient response," or an absence of presence in comparison to the live source. Radio frequency garbage generated or picked up by consoles is an often undiagnosed cause of sonic problems such as these, but designs which deal effectively with high frequency stability also serve to minimize major as well as minor RFI difficulties, which is important since RFI is a component of almost every environment.

When Is A Ground Not A Ground?

The answer is: when its impedance at any frequency is greater than zero, which is always. The design of the grounding scheme of a console so as to deal with this imperfection is an extremely crucial factor in achieving optimum sonic quality. Steady state measurements barely scratch the surface of the analyses required. Listening tests can evaluate a given design, but give little insight regarding specific points of success or failure.

Traditional single-point grounding schemes fail because of the physical size of the system; even straight wires become RF chokes instead of grounds at high frequencies well within the bandwidths of modern audio op amps. Separating signal and power grounds, effective in simple systems, becomes cumbersome in complex recording consoles where the demand for flexibility and expansion means that grounding problems encompass a plethora of signal paths, from the sublime to the ridiculous. A well-conceived single plane ground system is to be preferred, though this means trading difficulties of one sort for another. If successful, such a system gives a clean ground throughout the console and allows shields of cables to be grounded at both ends. Low frequency ground currents then flow in the low impedance ground plane and high frequency return currents flow in the shields, as in good RF design. The matter of shielding within consoles is not simple. "Grounding" both ends of cables can cause sonic interactions that defy detection by ordinary tests, yet not grounding shields increases RFI susceptibility and other problems. The matter of magnetic interference shielding is made more complex by the relatively high cut-off frequency of the foil-shielded cable commonly used — another reason to ground cables at both ends and design a ground system where doing so is always advantageous.

". . . if the slew rate of the console can measured the designer has made sonic concessions to specifications."

Most often schematics give scant insight into the problem of correct grounding; printed circuit board layout becomes a high art that has direct sonic consequences. If that doesn't shock you, think about it again: two circuits with identical schematics and identical components can sound different due only to different PC board layouts. One may reflect quintessential purity while the other rings like the service department hotline. Radio frequency engineers are familiar with this layout dependent amplifier/oscillator duality to which consoles are not immune.

Terminal Instability

The low output impedance of op amps implies the ability to deliver current to a load and since input impedances are high, the current must be supplied by at least one terminal in addition to the inputs; generally there are two such terminals. These power supply terminals connect to internal parts of the op amp's circuit and, depending on the design of that circuit, irregularities appearing on supply voltages can end up in the amplifier's output signal. Sources of such irregularities can be power supply spikes, ripple, or noise; radio frequency pick-up; other amplifiers driving loads; the op amp's own load demands, and so forth. Most op amps happen to be 20 to 50 dB more sensitive to problems on the negative supply because their integrator stage is referenced to that terminal. The NE5534 in addition employs a signal-to-positive-supply capacitor as part of its feed-forward circuit. Without correction, op amps can actually amplify signals applied to their power pins. In general, an amplifier's ability to reject power supply garbage (its power supply rejection ratio) decreases with frequency along with open loop gain, so high frequency irregularities are especially bothersome. The cure for these ills must include power supply bypassing and decoupling.

The Capacity To Solve Problems

A power supply, like any amplifier, has an inductive output impedance due to decreasing available gain at high frequencies. This means that its ability instantaneously to supply power to its load also decreases at high frequencies. To this effect add the distributed reactance of the supply wiring in a large console (keeping in mind that a PC board trace a half-inch wide is about equivalent to #22 wire) and it becomes obvious that without distributed energy storage a power supply system could contribute to serious problems, including noise, poor transient response, instability, and oscillation. The bypassing capacitors used to lower the supply impedance at high frequencies must be carefully chosen, with one eye on the textbook and the other on the actual circuit system. Even capacitors become inductive above a certain frequency; ceramic capacitors have been reported to degrade sonic purity even when used for bypassing, and tantalums may suffer catastrophic dI/dt failure due to current spikes. Capacitor selection should be based on characteristics of the signal, circuit, and amplifiers used and more than one approach will probably be required.

". . . the removal (of transformers) by itself is responsible for the removal of more sonic garbage from audio signals than any other recent development."

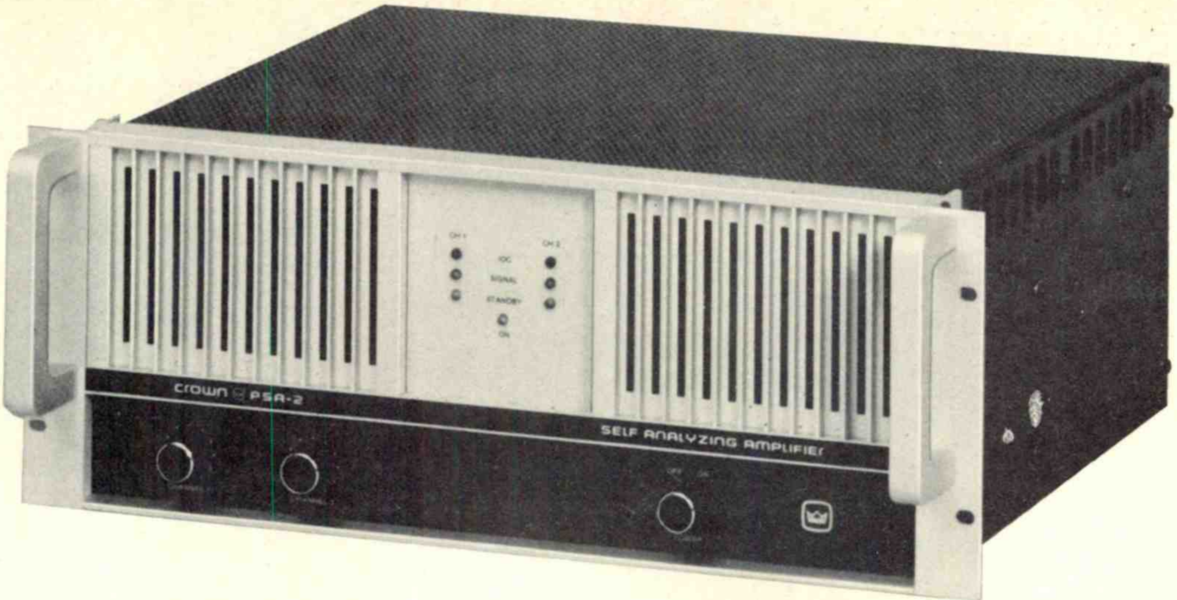
Bypass capacitors must supply energy to the load and so must be connected between the power supply terminals of amplifiers and load return paths rather than naively to the ubiquitous "ground." Again, the op amp topology and the loads must be analyzed, a difficult problem when the circuit is an entire recording console system. One problem load to consider is the capacitance of cable shields which can present a low impedance to high frequency signals; another is the virtual ground load, which returns to a power supply rail instead of to the system ground.

Ten Four Good Buddy

Decoupling is the deliberate increasing of power supply impedances between circuit stages, and thus to each circuit stage, so as to prevent one stage from "talking to" another through the power supplies. Decoupling is a concession to the realization that power supplies and grounds in real systems do not have zero impedances. Another goal of decoupling is to decrease the Q of resonant circuits in the power supply system, again to compensate for real world imperfections that even brute force on-card regulators fail to solve. One result of proper decoupling is improved crosstalk performance, but more significant in a well-designed circuit is the part decoupling and bypass play in increasing signal clarity and definition even within a single signal path. Another benefit is the increase in high frequency stability and the reduction of high frequency interference on the supply lines which will cause subtle signal pollution.

Seductive Curves

One design approach first seen in consoles in the mid 1970s, but whose popularity is rapidly growing, is the use of state variable filters in equalization networks. This basic filter topology can be configured in many varieties with varying success, but its appeal lies in reduced component tolerance sensitivity, lower op amp demands, and potentially independent control of the main equalizer parameters: gain, frequency, and bandwidth. The first of these features means that equalizers can be implemented more easily with ordinary components, and the second



This amp isn't getting a lot of attention...it doesn't need it.

The Crown PSA-2 is the most reliable high-power professional power amplifier you can buy at any price. One reviewer described it as "indestructible," since the PSA-2 accepts seemingly impossible loads, and unlike many other amp designs, will continue to produce useable output power.

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Crown engineers devised a simple test to show how the PSA-2 operates. Two flat metal bars are wired into the output circuit of one channel of the amp, with a music signal input. The other input is connected to a 1½ volt battery, requiring the PSA-2 to deal also with a DC signal. A heavy round steel bar is laid across the speaker leads. The amp *continues* to produce useable power, and the metal bar becomes a transducer, producing small sounds from the output signal!

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The PSA-2 uses its built-in computer logic and unique sensing systems to determine the limits of the safe operating area of the output transistors. The PSA-2 does not just thermal out or shut down as other designs tend to do under strange loads.

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Versions of the PSA-2 are now available with a choice of front and rear panel configurations. Users can select a model with on/off LED indicators for overload, signal and standby; or they can select the version equipped with the Crown "Dynamic Range Indicator," an LED array that displays peak/hold and instantaneous output for both channels. For the rear panel, a balanced input module (including variable gain and switchable hi/low Butterworth filters) is available, or unbalanced input only.

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for additional information circle no. 34

can give better transient sound quality compared to most other equalizer designs with identical steady state response curves. The last feature can lead designers into blissful specsmanship unless they realize that all that glitters is not beautiful symmetric curves on a graph.

State variable filters require more than a textbook approach to the design of equalizers optimized for noise, distortion, signal handling capability, minimum op amp count, control response, and so forth. The sensitivity of the filter bandwidth to amplifier gain bandwidth product requires thoughtful distribution of gain within the equalizer and crafty use of active or passive compensation, particularly in multi-mode stages (switchable peak/dip, shelving, and high-pass/low-pass characteristics). Another utility of the state variable filter is that it lends itself to digital control in that adjustments may be made at low-voltage, low-impedance points in the circuit where performance demands on VCAs and solid state switches are reduced. Experience and a critical ear determine success in the use of this complex and powerful topology, but excellent results are achievable.

Don't think that state variables are the only or even the best filters on which to base equalizers; that has not been proven, it just happens that their accessibility makes them popular. Zero sensitivity filters are now being researched in many topologies, and a bandpass filter very similar to the SV can be produced using two phase shifters instead of two integrators. This topology has many advantages to commend further investigations. Third order state variables are realizable and show reduced sensitivities, and Vogle filters using non-inverting integrators also may have audio applications. Ultimately, however, sound quality is the arbitor.

"Solid state design has taken 25 years to mature . . . monolithic (IC) amplifiers are younger than multitrack recording . . . it is natural for professional audio analog design to ahead everyday."

The No-Iron Console

The availability of better audio amplifiers has made practical the elimination of transformers from professional audio equipment and this trend by itself is responsible for the removal of more sonic garbage from audio signals than any other recent development. Transformers are a major source of linear and non-linear distortions, non-minimal phase response, hum pick-up, and other ills; if that isn't enough they are heavy, bulky, and expensive. They should be put in power supplies or museums.

It can no longer be argued that transformers are essential to building professional recording consoles and certainly no one is suggesting that they offer sonic advantages. Although transformers are slowly disappearing, the circuit techniques required for transformerless design are not new; this writer has been producing completely transformerless consoles since 1972. Until a few years ago users and designers were skeptical, but the measurable and audible gains to be made and the demand for increasingly better sounding equipment have compelled former critics to jump on the bandwagon with "proprietary" designs of their own. The skills needed to achieve better sound by designing transformers out or never designing them in are certainly not trivial or self-evident; there are as many nifty design tricks in this field as in any other, in addition to the broad considerations mentioned in this article. It is reasonable to expect that even when all consoles are transformerless sonic differences will remain.

Not only are they non-essential, but many uses of transformers actually degrade both sound quality and specifications of importance. Although it is popular to quote transformer distortion specifications at high levels, omitting low frequencies where distortion rises beyond 1%, such figures can be deceptive. It is at lower levels, -10 dB or even lower, where the specifications don't look so good and where the audible consequences are manifest. Unfortunately, this taxes the limits of most distortion analyzers so the matter is examined infrequently and specified even less. The use of step-up transformers on outputs is particularly humorous, because a slightly higher output level is traded for degraded noise, distortion, and bandwidth; headroom is actually reduced. Engineers should know that nothing in the real world is free, however appealing that concept may be; using a transformer for quick and dirty voltage gain is exactly that: quick and dirty. Another waste is raising output impedances to 600 ohms and balancing lines where it is not necessary — which is almost always. Six-hundred-ohm lines don't even relate to telephones any more and they certainly have nothing to do with professional audio.

Better Sound Starts At The Input

One aspect of console design that is particularly significant is the evolution of the transformerless microphone preamplifier. The main impetus for acceptance of this approach came from Paul Buff and his Trans-Amp™, introduced only two years ago, (though there was a 24-track NEOTEK in Nashville a year prior). The Trans-Amp™ module made it possible for users and designers to appreciate and emulate the advantages of transformerless design on a wide scale with very little effort.

Today, NEOTEK, Trans-Amp™ users, Harrison, Soundcraft, MCI, Tangent, TAPCO, and doubtless others are featuring transformerless mike preamps. All are based on a well-known topology called an instrumentation amplifier. Besides being designable for very low noise with microphone-like source impedances, designers love it because it allows gain to be adjusted over a wide range with a single resistor without seriously degrading the preamp's ability to reject unwanted common-mode interference such as hum and RFI. Transformers are sometimes defended for their ability to reject such signals, but a well-designed transformerless preamp can easily be ten times better in rejecting out-of-band signals and at the same time offer less in-band noise and distortion than the most costly mike preamp transformers. The ability of transformers to reject high DC voltages commends them to studios who experience hundreds of volts on mike lines, but in more typical mike patching situations with the ubiquitous 48 volt phantom power, transformerless preamps cannot become magnetically biased, a source of distortion that is difficult to diagnose and correct. A transformerless preamp need not have an input pad (they degrade noise performance anyway) to prevent saturation and may therefore present microphones with the designer's choice of load impedance at all gains.

The instrumentation amp topology deserves further comment because of subtleties which can be exploited to yield truly superlative sound, the kind you may not have thought your mikes capable of. In low-noise, low-impedance circuit design it is general practice to use a large total input device base area and high operating currents; nevertheless, the input impedance of an instrumentation amp can be made relatively high, an order of magnitude higher than is practical with transformers. This will reduce the load on microphones and has been found to improve the clarity and definition of nearly all mikes except older ribbon types. Quality capacitor mikes, especially the excellent Schoeps/Studer series which are transformerless themselves, really open up if the rest of the console is designed to maintain the quality.

Most instrumentation amp preamps, with the exceptions of Tangent and TAPCO, use a circuit design which inherently keeps the collector-base voltage of the input devices nearly constant and which returns feedback directly to their emitters. This results in a highly linear input stage free of Early effect, an important and bothersome cause of transistor distortion. Additionally, the second stages are made up of op amps operating in the inverting mode. Since these stages contribute the majority of the gain, operating them so that the input signal does not appear as a common-mode voltage further helps reduce distortion and improve the sound. Common mode rejection of the entire preamp can be reduced in several ways to below the second harmonic product of the input signal: virtually unmeasurable.

One other point deserving comment is the use of the LM394 monolithic supermatched transistors as transformerless preamp input devices. It has been well-reported that even with two such devices in parallel the noise figure at microphone-like source impedances is inferior to that possible with discrete transistors (in any well-designed circuit, preamp or console, the input stage should dominate the noise performance). There are, in fact, inexpensive plastic transistors which can advantageously be plugged into LM394 sockets in mike preamps, and PNP transistors are available which are better yet. Designers can avoid the selection problem by relying on the carefully matched PNP discretes in the Trans-Amp™ module.

Eliminating transformers in all audio circuits is a credible attack on sonic smut; consoles, tape machines, and other professional equipment will show increasing progress in this regard. Even though transformerless designs are not thoroughly developed, they are so superior to transformer-coupled circuits in terms of sonic performance that for anyone who listens there is no turning back.

There's More

The points raised in this article are by no means exhaustive. Recording console design poses many challenging areas that have not been mentioned at all, apart from physical design and human engineering. VCA design, which has been debated at length in this publication, is such an area, as is the matter of gain structures and impedance relationships. Passive components, particularly capacitors, have been shown to effect sonic quality, and this area is relatively unpublished. FET switching and its sonic consequences deserve intensive treatment, particularly with regard to the new V-FET devices which show promise for amplifier design as well. The many questions of sound quality vis-a-vis digital audio design loom ahead and analog designers may be expected to contribute here as well (apart from setting a high standard) in areas such as anti-aliasing filters where Lerner and other non-minimum phase filters may solve critical problems. There are even far-out possibilities awaiting research; perhaps the design of a mike preamplifier that would exploit the low noise and facile digitization potentials of parametric amplifiers and so contribute to the all-digital consoles eventually to come.

I hope the reader has by now concluded that the requirements of good console design are so many and so varied that their successful application remains an art — not yet a science and certainly not a simple matter of circuit design and component selection. The essence of art is not so much in its technical achievement as in its appreciation. In the final analysis a console must have a sound quality which skilled and experienced engineers and producers (you, the console user) will find satisfying.

Literature citations have been omitted from this article because of its non-technical nature, but it is important to understand that the points made and their underlying principles are detailed in hundreds of monographs and journals. Also understand that the concepts presented are not the exclusive property of any writer, designer, or firm. They may be, and should be, used and extended by anyone who finds them conducive to better sounding designs. □□□

"What No Man Has Heard Before . . ."

The SOUND EFFECTS for STAR TREK THE MOTION PICTURE

by Steve Barnett

Just what does it sound like "where no man has gone before?" When the makers of "Star Trek — The Motion Picture" were faced with this problem, their answer bore a close resemblance to the question.

"They kept repeating," according to sound editor Cecelia Hall, "'We want things that have never been heard before.'"

"We talked to the sound people," said the film's director, Robert Wise, "about reaching out and going as far as they could into areas that maybe they hadn't explored before; everything they could come up with in a combination of areas in sound and electronic sounds. We wanted combinations of real sounds to be put through some of the synthesizers and other gear to get something that was hopefully quite unusual and quite different, something that hadn't been heard on the screen before."

"I felt the picture should have a very unique audio style," said Todd Ramsay, "Star Trek's" editor, "and should perhaps contain sound effects that would become as characteristic of science fiction in our era as did certain sound effects from past eras of science fiction pictures, such as "Forbidden Planet" and "This Island Earth." Those films had sounds in them that were both original and indigenous to them. More importantly, these effects formed significant parts of sound effects libraries which were continually re-heard throughout subsequent science fiction films."

All parties concerned wanted to avoid sounds from sci-fi films that were either cliche or overly familiar to audiences. New sounds had to be created, and the task fell upon an array of sound creators, editors, and mixers to supply them.

"I began casting about for people to work on the sound effects," recalled Ramsay, "and many people submitted demonstration tapes of the three things that we had asked them to do as an example of their work. About this time I engaged Richard Anderson as my supervising sound editor."

Aside from discussing the sounds needed for the more spectacular visual effects in the movie, Ramsay, Wise, and Anderson also gave consideration to the basic environments of the Star Ship Enterprise, as well as the interiors of the other space craft in the piece.

"One of the concepts behind the sound in these areas," said Anderson, "was that in the future, man would be very tired of noise and would attempt to make things more quiet. We set out to create a new feeling for this film, different than other science fiction/space motion pictures."

He cited an earthbound example of quiet environment planning.

"In movie convention, whenever there's a radar screen, there's a beeping sound upon contact of an object, but that's not reality. Imagine if you were in an air traffic control room with ten radar stations and each of them was making that sound. All the little beeps and clicks would drive you and everybody else crazy, so we decided on primarily visual readouts for the bridge crew stations and screens. Noises were added for special situations, such as alarms and warnings."

These background sounds, though important, were not where the major creative avenues lay. The special visual effects for the film would require special audio effects to match. Much of the responsibility for the sound effects fell upon Anderson's shoulders as activities in the cutting room began to occupy almost all of Ramsay's time. Anderson came on the picture in July of 1979, and at roughly the same time, a number of sound effects creators were hired based on the demonstration tapes.

"We condensed it down," said Ramsay, to Frank Serafine, Joel Goldsmith, Dirk Dalton, and Alan Howarth. In addition, Wise, Ramsay,

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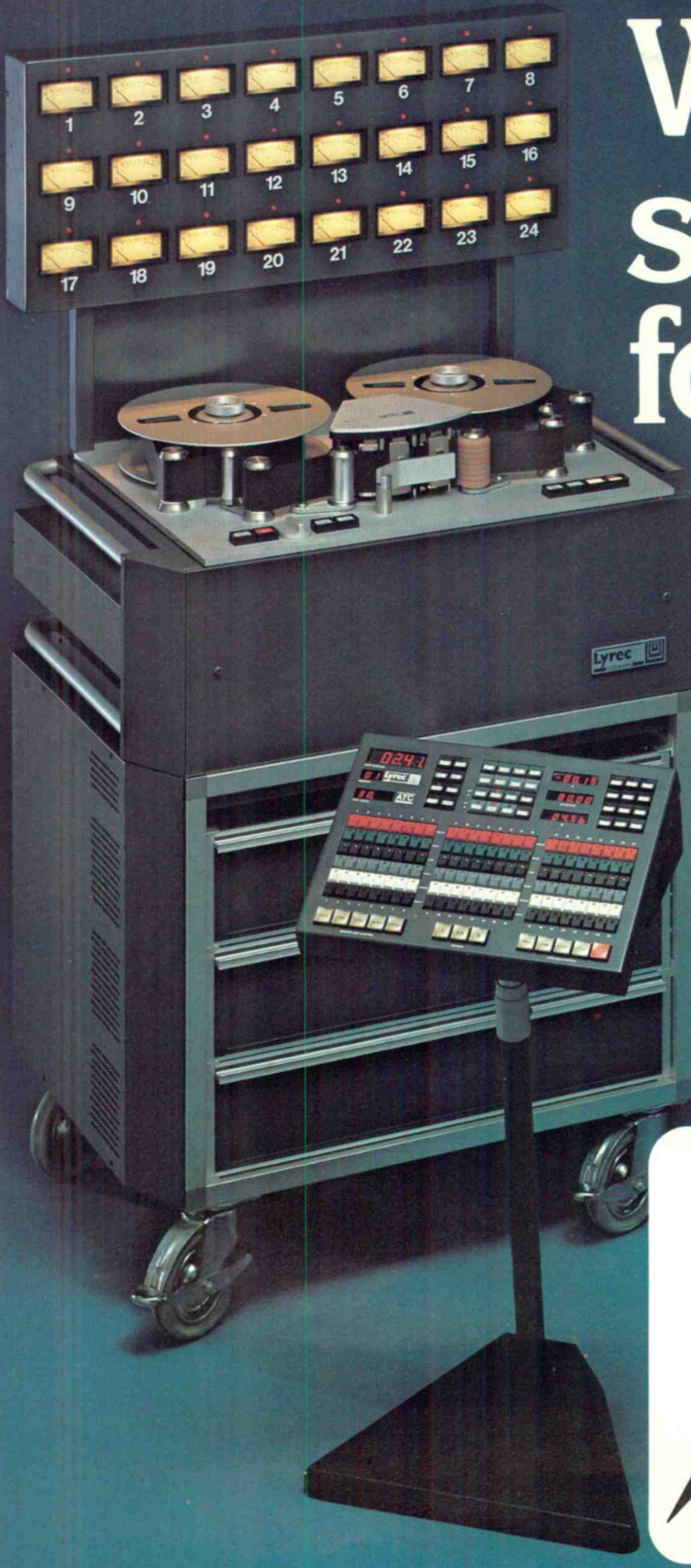

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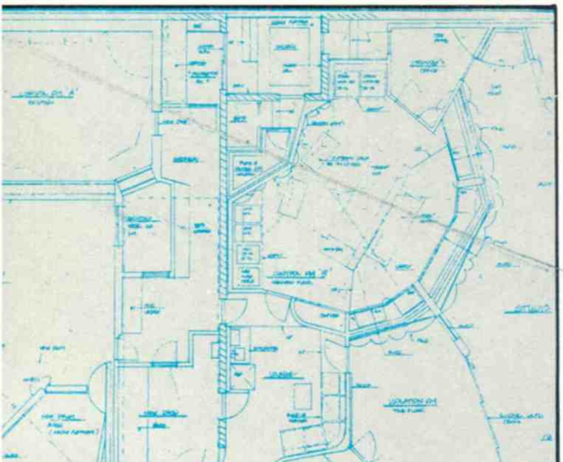

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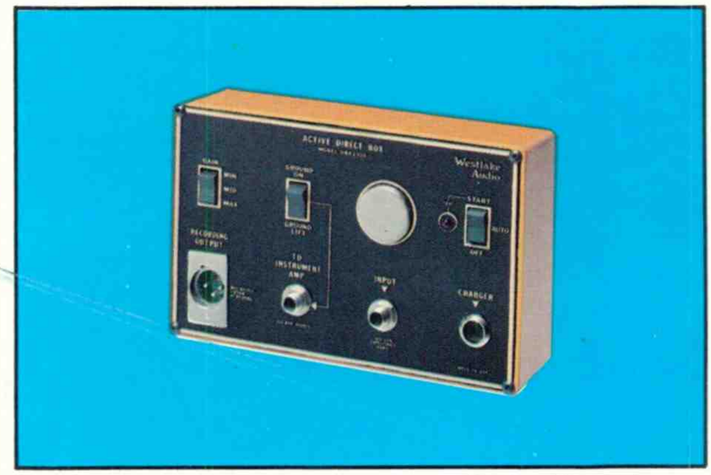

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SOUND EFFECTS for STAR TREK THE MOTION PICTURE

and Anderson had at their disposal some material by Francisco Lupica which had been submitted earlier in the film's production. Anderson put together his crew of sound effects editors, including three Paramount staff cutters who had been with the picture for some time, and two other editors from the outside. The recording team to mix the film was to be headed by Bill Varney, at Goldwyn Studios.

The different effects creators worked closely with all the editors and mixers, with creative input coming from all levels with Ramsay and Wise having approval over all.

"We got each of the sound effects creators going in a specific direction," continued Ramsay, "and got to work. We took the initial demonstration tape from Goldsmith and the one from Alan Howarth and put them together and made what turned out to be the ship's engines."

As there was no music for the film, this was the sound used in the trailer for "Star Trek."

In many cases this proved to be the working method adopted by Anderson and the other sound effects editors. Oftentimes two or three parties were engaged in creating the same specific effect for the picture. Each would submit as many as 10 to 15 elements. The editors would then listen to them and choose the ones they felt were the most appropriate, often combining and overlaying effects from different creators.

This fashion of operation grew to a large extent out of what proved to be the most frustrating problem in the film's post production: the lack of time. Delays in the area of special visual effects, so crucial to the picture's success caused the film to fall far behind schedule. This pressure could not be eased as Paramount had already booked a record 900-plus theatres for "Star Trek's" early December opening date, and the film's release could not be moved back.

"Consequently, they somewhat tried to isolate us," explains Howarth, a musician and synthesizer technician/programmer who had worked extensively with the group Weather Report, but had only done sound on student films before "Star Trek." "The opticals were coming in so late that there wasn't time for us to see them and then go home and think about them. We designed things blind, and then they were able to choose or make combinations from the submissions of all the teams when they played the effects against the visuals in the last few weeks of post-production."

This loose separation and somewhat competitive atmosphere between effects creators provided the editors and mixers with much needed options, and the competition was on more than an artistic level since bonuses were paid for each major effect that was used in the film.

Goldsmith adds, however, that had time permitted, he would have preferred more cooperation between the effects creators.

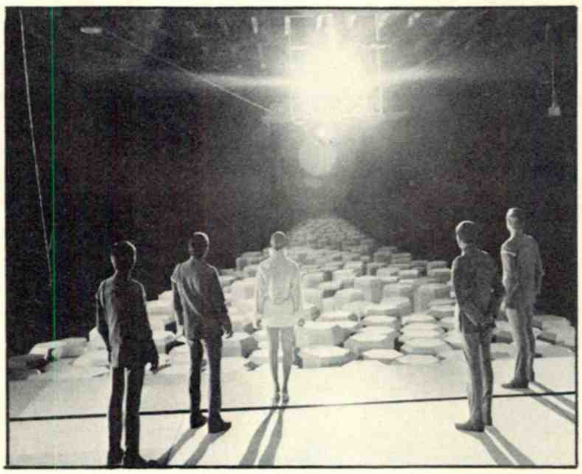
"It would have been nice," he comments, "to have feedback because you need that peer encouragement, but, because of the schedule, we were competitive instead of working together."

For ease in reference, assignment, and pre-dubbing, Anderson divided the effects into three catagories. "A" effects were the major effects, such as space ships, "wormhole" and "digitalization." "B" effects were minor effects such as doors and switches, and "C" effects were room backgrounds and crowd murmers. In the pre-dubbing for the film, when the various tracks were mixed prior to the final mix, the effects were grouped according to these catagories.

"We pre-dubbed everything separately," explained Anderson, "so that during the final mix, if they wanted to make a switch click louder, they could raise it without the background coming up as well."

"So we had in the final mix about four or five effects pre-dubs; a dialogue pre-dub; sometimes a separate processed dialogue pre-dub as with Ilia's voice; all the music tracks; and a foley pre-dub of all the footsteps and body movement that we had recorded on the foley stage."

This latter process is nearly as old as sound in movies. Footsteps and other noises, which for whatever reason could not be properly picked up in production recording, are re-created by "actors" who watch the picture to keep in sync with the action. Any adjustments in these newly made tracks are done later by the editors.



In the case of "Star Trek," much of the production recording could not be used because of noises inherent to a number of the sets. The bridge of the Enterprise where much of the action takes place, was one of these sets. It was crowded with display monitors at the different crew stations.

"For some reason, to get them to photograph well," said Anderson, "they couldn't use actual video screens, which are quiet, so they had banks of 16 and 8 millimeter rear-screen projectors behind the set."

The racket put out by these projectors made it necessary to foley most of the scenes shot on the bridge and to call the actors back after production to replace their dialogue.

Another area of the film requiring similar adjustments was the final climactic meeting with the Vger deep inside the alien spacecraft.

"They shot with these big 70 millimeter cameras with the optical effects work to be done later, but they were un-blimped," said Anderson, "so that they made all this atrocious camera noise." The 70 millimeter frame is twice as large as the normal 35 millimeter, providing the visual effects creators more area in which to work.

In addition to that situation, a special technique was used during the filming of the scene.

"Because of the nature of the sequence," Wise explained "and the awe that this inspired and to help build a whole sense of wonder about what was going on, I played back on the set a combination soundtrack that had been made up by one of the fellows who would later work on the sound in post-production.

"In a musical, you know, you play the musical track to give the actors something to dance or sing to. Well, in this instance, I played these marvelous, crazy, weird sound effects to give them something to respond to in addition to just the visual sight of the satellite. Although I don't plan every distinct sound for a picture, I certainly lean very heavily on sound for emotional response, in the theater and even on the set during filming."

Post-Production ADR

Recording of the actors' voices to match the picture after the end of filming was handled at Goldwyn Studios through a process called Automatic Dialogue Replacement, or ADR.

In another common technique of looping, the actor will hear the originally recorded line, then read it for recording, then hear it again, and read it again until the editor thinks it matches. "They're surprised when it falls in sync," says "Star Trek's" dialogue editor Sean Hanley. Using ADR, Hanley records the performers as they read to the picture. He listens to both the original dialogue and the new version as it's read by the actor.

"The way we had it set up we could hear it as they read it," he says, "so we could tell if the new one matched the production track close enough so that we could make it work."

There are several areas in "Star Trek" where the voices of the performers were processed or altered to highlight or accompany a visual effect, or simply for dramatic reasons. In these instances, the dialogue was considered a special sound effect.

Unlike the Klingon dialogue, which was spoken in their "language" during filming, the actors in the scene on Spock's native planet Vulcan spoke English during filming and were brought back to do their ADR work in a specially developed Vulcan "dialect."

"What occured," continued Hanley, "was a UCLA speech professor came in to look at the Vulcan sequence. He then created mouth movements which sounded very foreign, sort of gutteral sounds, representing the Vulcan dialogue. It then became a case of trial and error, depending on how well the performer could time it. We also had the professor's help to match the mouth movements, and then it was up to me to say, 'Yeah, that's pretty good, I think I can make that work.'"

The ADR process for that scene took about two days.

For the "wormhole" sequence, the ADR was used to adapt the dialogue for a special visual effect. This took place after Captain Kirk takes the Enterprise out of its dry dock and opts to go into warp drive before the matter/anti-matter engines are thoroughly aligned. The Starship goes out-of-phase and is caught within a treacherous rift in time and space. The crew's actions decelerate, and they shift into slow motion.

"They shot the scene and recorded the dialogue at normal speed," said Hanley, "24 frames-per-second. Then the editor cut the sequence and production track and sent the picture out to have it optically slowed down. We had to have the actors come in and read to the slowed down mouth movements. Using lines looped to the normal picture and then slowed down was considered," according to Hanley, "but it tended to sound too mechanical, whereas the ADR had more feeling to it. We got a few little extra slurps and gurgles and things like that as we went along. Those sounds wouldn't have come out as well otherwise."

The interior scenes of the "wormhole" sequence were some of the first with opticals completed, having been done by an earlier visual effect team.

The effects editors under Anderson's supervision were Cecelia Hall, George Watters, Alan Murray, Stephen Flick, and Collin Waddy. The first three are on staff at Paramount, and the remaining two are independents. These editors were each assigned the responsibility for individual reels from the picture, a reel being roughly 10 minutes of film. Fortunately, according to Hall, the major effects were nearly all contained in their entirety within individual reels. Separate reels with similar effects were handled by the same editor. They would then work with the effects creators on the different sequences in each reel. The "wormhole" sequence was assigned to Stephen Flick.

"We wanted these ascending tones to the point of Warp 1," explained Flick, "then we go into Warp 1, and then the "wormhole" appears on the view screen, and then all of a sudden the Enterprise starts chattering. When Alan Howarth and I talked about this sequence I discussed it as a Volkswagon whose engine is out of time."

The sequence developed into a rather complex combination of effects. A swarm of bees from the sound effects library were adjusted by Flick and fellow editor Collin Waddy through a vari-speed device supplied by Glen Glenn Sound. Other low-end "home brew" elements, Anderson's term for sounds created by the editors themselves, Howarth's original effects for the sequence, and tracks by Joel Goldsmith were also used. Howarth processed his tracks to the picture through a harmonizer and a pan-pot to change the stereo phasing so it would add and cancel left to right. Later, Howarth submitted a reconstructed or "retrofit" group of effects which were pre-dubbed down to four differently harmonized mixes. These final five units were hung in the machine room with the others for the initial interior mix of the "wormhole."

Most of the effects in "Star Trek" were constructed in a similar fashion with contributions from every sector, including the re-recording team. Material from an individual creator, however, would often end up as the audio focal point for a given scene in the film.

Howarth's effects for the "wormhole" were created primarily on a prototype Prophet 10 synthesizer, the more advanced version of Howarth's own Prophet 5. "It was basically a pulse modulation," he says. "It makes dash lines out of the voices (synthesized tones)."

All of this material was created, adjusted, and premixed to the picture before the exterior shots of the Enterprise in the "wormhole" had been delivered by the visual effects teams. Flick, as did the other editors, had to put slugs of black leader in where the as yet unseen opticals were to be placed in the film. "We played the interior 'wormhole' audio material over the entire length of the exterior. In some cases they were shorter, and in other cases, they were longer, which meant that I had to slide tracks one way or the other before a final mix was made."

"For the exteriors, we had an incredible amount of material that had been turned in by everybody," says Flick. "So we just sat down at a bench with all this material and played it to the picture." Among the sounds used in this sequence was an unusual effect produced by Frank Serafine. It was originally submitted to go with asteroids bouncing off the deflector shields.

"The interior sounds were actually going even when the focus is on the exteriors," explains Howarth, "but then Frank's cowboy sound was added to it to make it bigger and boomier."

"I took the sound of a barroom brawl from an old Paramount western and reversed the direction of playback," explained Serafine, who had worked laser shows before doing sound on a TV documentary, "and then ran it at half speed. If you analyze the sound of something being hit, of the actual blows, there is a moment of impact, then air moving, then cracking and bashing. Reversing the cowboy fight created the pulling effect we wanted." This signal was laid down on eight different tracks with varying starting points and processing.

Because the most complicated visual effects in "Star Trek" were in the opening and closing reels of the picture, these proved to be the last of the opticals to be completed. This created problems for the sound effects people at every level, since they were essentially flying blind against extreme time deadlines.

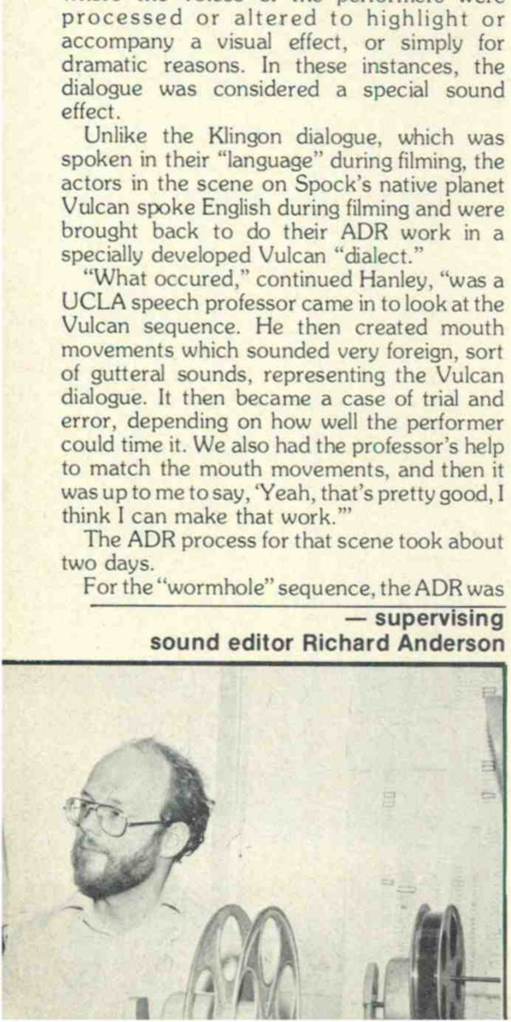
To try to keep ahead of the game, the sound editors and creators would do their best to see the opticals in progres. "We would make trips to both John Dykstra's and Douglass Trumbull's places," explains Hall, "and we would view the dailies of the opticals so that we would have some idea of what it would look like, but before that, we would just see production drawings."

"A lot of what we saw down there," adds George Watters, "was completely different from what we finally received, so right at the last minute, we had to scramble."

"Some things," continues Hall, "you could work on without really knowing what it was going to specifically look like. For example, we knew that in some scenes there was going to be a major effect called 'digitalization.' This

— sound editors George Waters and Cecilia Hall

— supervising sound editor Richard Anderson



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was the process by which the Klingon space ships were completely destroyed by being dematerialized in a computer-electric manner. That, by itself, is a place to start. We knew there was a lightening bolt, although they very specifically did not want a real lightning sound used."

For "sort of like" sound effects, the editors developed a technique of playing back a library sound to give the effects creators a point from which to depart in the creation of their effects.

With regard to the "digitalization" of the Klingon vessels, the sound editors realized that there was really no set beginning or end to the process, so this made the development of the audio effect somewhat easier. For his work on 'digitalization,' Serafine created roughly 40 elements, some of which were the result of a number of mixes of various synthesizer tracks from his 8-track recorder.

"So that the editors would have some concept of what I was trying to do," he explains, "I would take all those elements and make a composite mix that could be played with the rough cut visuals so that the editors would have an idea of what all these elements would sound like together."

When the final opticals came in, the editors would then have a place to begin in building the sound. If necessary, they would have the creator go back and re-do his individual tracks of material so that they would more precisely fit the newly arrived visual. All the elements were then hung in the re-recording room, and the mixers and sound editors, along with Ramsay and Wise, would select the combination of elements which worked the best. For the sake of identification these individual sounds had to be named, usually in an arbitrary fashion.

"Frank made an element that was called 'electrical element number one,'" explained Hall. "Another was called 'digitalization low technology number 2' while another was labeled 'high frequency whine.'"

"You had to call everything something," adds Serafine.

Serafine's contributions to the "mind meld" sequence in the last reel of the picture involved nearly 360 individual tracks created and mixed with Japanese producer/engineer Miki Curtis. The material was generated on a Moog synthesizer, along with a Prophet 5, which is an analogue synthesizer with a digital memory. A Prophet 5 was also used extensively by Alan Howarth in his development of sounds.

The Prophet

Perhaps one of the most promising innovations to come out of "Star Trek" was the use of synthesizers in the area of sound effects. The Prophet 5 was distinctly well suited for the task.

"I had been using huge modular systems," explains Serafine. "They required lots of patch work to give me the kind of variation I needed, they were not very quick to set up, and they were monophonic. Another problem was that I'd spend all night to create these sounds, and I'd take them in the next day, and they'd say, 'Oh, that's really good, but can you change this or that?' "

"On a modular synthesizer," continues Howarth, "you have little phone plugs and a patch panel, so that if you do something on the synthesizer and then unplug everything, finding the exact same sound again is difficult, especially if they say, 'that's the sound, just change this one element.' On the Prophet, the exact sound is stored in memory. You just call that sound up, push an edit button, and start changing it."

Forty different programs can be stored in the system's memory. In addition, all the programming for the Prophet is internally routed. No mechanical patching is used, saving set up and development time.

— sound effects creator Joel Goldsmith

The other effects teams also used synthesizers in some of their contributions to the sounds of "Star Trek." Joel Goldsmith did not have a Prophet at his disposal, but managed to get around the retrieval problem in another fashion. "Using a 24-track recorder with synthesizers," he explains, gives you the option of keeping elements by storing them on individual tracks. It gives you a lot more control over the sound, though it is time consuming. It's an anlog way of storing your synthesizer sounds, and it gives you a lot of freedom." The 'sonic shower' accompanying the 'Ilia probe's' return to the Enterprise was a synthesizer effect by Goldsmith.

Francisco Lupica, who had been engaged originally to do the effects for "Star Trek" also used some synthesizers in his development of sounds. One of the elements of Lupica's in the film is the 'Vger light probe,' which enters the Enterprise and kidnaps Ilia, the ship's navigator. The sound was built on a 24-track recorder and produced in collaboration with programmer Malcom Cecil at Tonto Studios, in Venice, California. The tracks were generated via techniques involving Oberheim Expander Modules, Modular Moog, and the ARP systems.

Other methods of sound development were utilized by all the creators involved. Two such sounds were combined to create 'Vger's Sentry Bolts' which were shot out of the huge alien vessel to orbit the earth in the film's last reel. The 'Sentry Bolts' were similar to, but in theory, 100 times larger than the 'whiplash energy bolts' of pure plasma energy which shot out of the alien craft to 'digitalize' the Klingon battle cruisers in the first reel. The two elements were conceived separately by Howarth and Serafine, who were both working independently in the same area.

"I did my contribution with a pedal steel guitar," explains Howarth. "I put it through a Vocorder while running the slide up and down the strings. Then I gave it the rock and roll treatment by putting everything on 10 and distorting the hell out of it." This track was one of the lower elements of the effect, while Serafine's was in a higher frequency range. He ran a microphone through a harmonizer and simply made the sound by passing his mouth by the microphone with a screeching sound.

Serafine was also responsible for the sounds of the transporter and its subsequent malfunction, an effect that had such a high level of recognition from the television series that due homage had to be payed. After exploring the possibilities of a digitally controlled digital synthesizer, Serafine turned to a Roland JP-4 compuphonic synthesizer to arrive at the original sound. For the malfunction, the transporter's normal sound was transfered from 2-track to an 8-track recorder. He then ping-ponged the signal onto tracks three through eight running the signals separately through flangers, harmonizers, a wavemaker, a dual phasing device, and a reverb unit.

For another element to the malfunction, microphone feedback was generated from an AKG-414 held at 10 feet in front of a twin reverb guitar amp with a compressor as part of the signal loop. This sound was played back in a bathroom, re-recorded, and then the

— sound effects creator Frank Serafine

— sound effects creator Alan Howarth

resulting sound was reproduced at half-speed to provide the desired effect.

Home Brew

The effects creators, as they were credited in the end titles, were by no means the only individuals generating sound effects for "Star Trek." Anderson's crew of editors produced a large number of 'home brew' sounds by processing effects found in the Paramount sound effects library with variable speed transfer devices and filtration systems, as well as by changing the direction of playback. Sound editor Collin Waddy became well known for experimenting in this area. One of his assigned reels included Spock's space walk, which, in the pre-dub stage, relied heavily upon 'home brew' effects.

"For the rocket pass-by," he says, "I had about six or seven elements going. Some of them were synthesized, and some were not. One of the things used, was actually a ripping sheet as the rocket went past. I sort of slowed it down and speeded it up, and it worked quite nicely."

"Others that were involved here," added Ramsay, "were Bill Varney and the mix team who did an absolutely brilliant job, and who culminated the production of some of the effects with additional engineering on the re-recording stage."

The mixing was an on-going process throughout the post-production period of the film, with its numerous pre-dubs to combine various elements of sound for an effect or effects. Consequently, Varney, who handled the final dialogue mix, along with music re-recording engineer Steve Maslow and effects mixer Gregg Landaker, were to a tremendous degree responsible for the final outcome of all the sound effects in "Star Trek."

Before Varney could get the various reels of effects hung in the machine room for mixing, cue sheets had to be checked by the editors who cut the reels. These indicated what sound was located where on the individual tracks in relationship to the reel of picture.

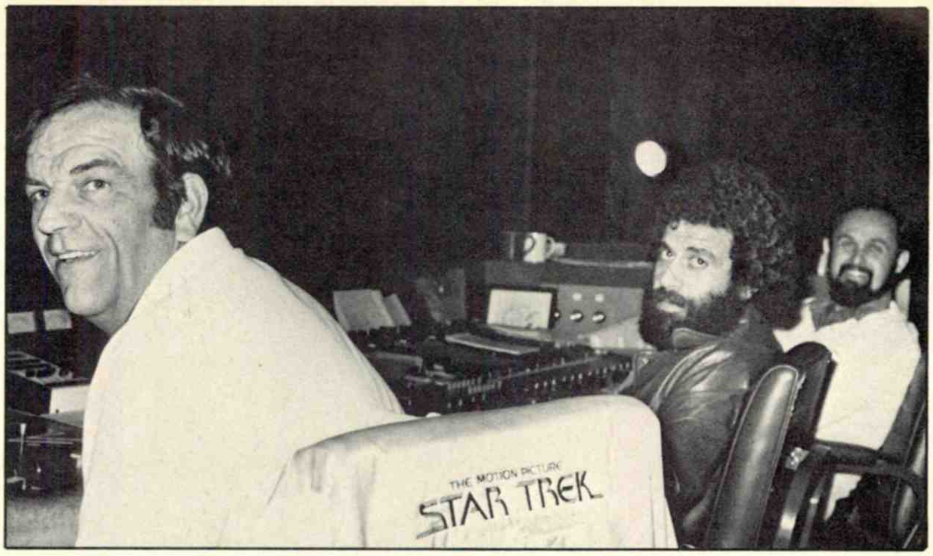
"Each audio reel had to go on a specific machine," explained Hall, "and the mixers had to know which tracks were Dolby, which tracks were non-Dolby, which ones were stereo and which ones were non-stereo. The non-Dolby effects were usually those pulled directly from the sound effects library and had not been Dolby encoded when transferred to 35 millimeter stock, which in some cases had occured years before. Most of the ¼-inch tapes turned in by the effects creators were not Dolby encoded to avoid problems with matching Dolby tones between each of the studios and the mix rooms at Goldwyn.

In addition to high-speed recording to avoid tape noise, explains Howarth, "a technique that I used to generate multiple synthesizer effects without multiple individual tracks was to combine the Prophet, the ARP Quadra, and the two ARP Avatars, have them run simultaneously, and directly record them to two tracks. I'd end up with a nice stereo mix of the four synthesizers, and I saved a transfer generation right away."

When all of the original sound effects tracks were transferred from ¼-inch to 35 millimeter magnetic stock, they were Dolby encoded, and of course, the Goldwyn mixes to 35 millimeter were Dolbyized throughout.

As with the other participants in the making of "Star Trek," Bill Varney's worst enemy was time. Because so much of the optical work had not been completed by pre-mix dates, many of the more involved sound effects could not be created, let alone cut into the reels. Varney and his team left gaps where the effect would later be added both in the picture reel, and in the sound reels. The optical and its sound effect would be carefully added later. This called for the implementation of some techniques to avoid recording over the mixed sections of the reel when the visual effects eventually arrived. One must keep in mind that Goldwyn's re-recording stage "D" is not automated. All the moves must be done manually.

"We generally would work up to a point where the scene would change," Varney explained, "so when we picked up recording again, it would be in a new environment or a new circumstance in the film, but the mixing of some sequences became more involved and complicated than just slugging in the effect. For instance, we would record up to the end of a scene where the music would have to dribble into the next scene which would be this optical that we didn't have yet. In those cases, I would get my dialogue comfortable and done, Gregg Landaker would get his sound effects down and done, and Steve Maslo would just let the music cue play on out. Then we would take that recording and use it as a unit, in essence a pre-dub. When the opticals showed up, Steve would make his fade out on this pre-dub working against our new material, and we would go on. It meant we had to go another generation, but in some cases it was the only possible way we could do it, and we were pretty careful to make sure you couldn't detect any generation loss. Each individual moment utilized a new technique. There was never any one technique that would work for every reel."



— re-recording mixers (L to R) **Bill Varney, Steve Maslo, Gregg Landaker**

"We got into such a crunch during pre-dubs," continued Varney, "that we also utilized two studios at Goldwyn. While I was concentrating on dialogue and things like that in studio "B," the other guys would be going strong in the other room."

Usually Wise and Ramsay would attend pre-dub sessions for the major effects along with the sound editors responsible for cutting the reel being mixed.

"A giant unknown that existed while mixing effects, and one that continued through the final mix was the fact that we never knew what the music was going to be like in any specific sequence," added Varney. "Jerry Goldsmith was literally writing and recording the score while we were doing the final recording for the picture. We didn't exactly know what we were going to be working against, and that's what this business is all about. The whole layering technique comes from knowing what's going to be there. So when we got into the pre-dub stages, we had to take gambles. We'd guess, 'Well, Jerry will probably be doing this or that,' so we'd adjust the effects accordingly. And then there was the possibility that once Bob Wise heard the score, he might not like it. He might have opted to just go with the sound effects, or the other way around, or a different combination."

"I never heard the music until it was time to put the reels together during the final mix," recalls Steve Maslow. "I didn't know how the music was going to be structured until everything finally came down. It wasn't until then that we got a chance to hear how the music was doing."

Final Re-recording

Signal processing on the re-recording stage was common to further adjust the effects to better fit the picture. At times, an optical would come in with a little element that had not been taken into account. The mixers would make little enhancements with harmonizers and other gear to sculpt the sound around the previously unseen element.

"The 'wormhole' is a good example of dialogue enhancement," explained Varney.

"There had been a lot of lowering of the pitch of the dialogue there when it had to be replaced in the ADR room. We ended up modulating the voices with the aid of a vocorder using the 'wormhole' sound effects as the exciter."

One of the more critical areas of signal processing concerned the voice of Ilia after she returns to the Enterprise as the Vger probe.

"We wanted to give her some treatment," described Wise, "that had a little sense of something mechanical, but without getting into the very familiar sort of computer voice that we all know so well."

"That was originally in the dialogue department's hands," explained Howarth, "and they did some very interesting things. Mark Mangini, the assistant dialogue cutter had cut Ilia's dialogue into vowels, consonants, and less important syllables, and split these off onto three different tracks trying to come up with this broken, segmented dialogue. They also tried vocording that and all this other stuff, and they were just not happy because it was too mechanical.

"Richard Anderson finally called Howarth to bring his gear down to the re-recording stage to work something up for the voice.

"Ultimately," explained Howarth, "the sound they selected was created with a flanger that had its input controlled by a 16-step sequencer. Each of the steps gives off a specific voltage that I had tuned to the equivalent of a melody line thus creating this little flange melody. Ilia's voice was flanged in the low frequencies, then it skipped to the high frequencies, and then jumped down to the middle, rather than a smooth swept oscillator sequence. This jumping effect ultimately satisfied what Mark Mangini was trying to achieve by dividing the syllables of the words. When we finally came up with the thing and everybody liked it, we came in the next day and ran all her dialogue through the process, which we called the 100% effects voice. So we had that on one channel, the original un-processed voice on another track, and we made a 25% track for a third channel. Now Bill Varney had the choice of dialing up either direct, 100%, or 25% for both intelligibility and to change her character throughout the sequence. You could then tell if she was reacting to Decker's attempts to revive her memory of the Enterprise and her previous life. When she started getting those feelings back, she would sound human, and when she reverted to being the Vger probe, she would be very mechanical. So those three tracks were run simultaneously, and then it was a matter of Varney dialing up the necessary character."

Another section of the film's success was the voice of Vger itself. As the deadline for the final mixing of the last reel approached, no one was completely satisfied with the various possibilities.

"I felt very strongly," commented Wise, "that I wanted to arrive at something that through those pauses and light and sound effects changes would give us a sense of Vger himself trying to communicate with us. It had to be something in his voice."

"When you examined the rules set by Wise and Ramsay," commented Varney about the voice of Vger, "you found that it was an impossible thing to create. There were so many things that it couldn't be. At one point we had Bob Wise talk out what he though Vger would be saying. Then we used his voice to modulate all kinds of things. For a while we thought we were pretty close to it, but none of them made the movie."

"I got a call one day," recalled Howarth, "and Richard Anderson said, 'Well, Al, why don't you bring all of your stuff down to the stage and do something right to the picture. We need something that works.' So I hauled my stuff down, and we tried this and that, and we decided on some sounds that were generated on the Prophet. The next day I took the Prophet down all by itself, and it was a matter of designing programs that had a sort of 'vowelly' sound to them. It was a manipulation of both keyboard and program switching. One hand was playing the programs, and the other was playing the note and the initiation of the note."

"Interestingly enough," adds Anderson, "we recorded on the soundstage with Al while he was playing to the scratch track of Bob Wise as the voice of Vger."

"I had the dialogue in one side of my earphones," continued Howarth, "while watching the movie and hearing the synthesizer through the other side of the phones. I tried to do the cadences of the language in my playing. What's in the movie was done in about four hours, and for me it was just intense pressure."

In addition to the voice of Vger, the scene involved a rather elaborate ambience track. While addressing the problem of Vger, Joel Goldsmith also developed material that was used for this effect during the encounter.

"I took an ambience recording of a street at about four in the morning, no cars," Goldsmith explains. "There's a certain sound about it. So I started processing that. Then I put my voice into it, slowed it down real low to sort of a groan sound, and put that through a lot of digital delay and a little bit of flanging and pitch change.

"Then I made a stronger, more turbulent ambience with the synthesizers and sequencers and processed that using a lot of reverb, delay, flanging, and phasing. With these tracks down, I took some Emile Richards percussive elements and layed them down after processing. He was doing things like dropping super balls on timpany and then adjusting the pitch of the drum head during the attacks. I made it real thick by adding a lot of delay with a large amount of regeneration to get these swells with percussive intent."

Along with these sounds from Richards, Goldsmith generated some more experimental percussive elements and used them as the exciter signals for a Kepex and various VCAs through which the ambience tracks were routed.

"The ambience sounds took on a color," says Goldsmith, "a percussive quality and then the slow delayed percussive color, and then move back into normal ambience that would slowly die out. Each time Vger spoke, the percussive elements became more and more violent, and the ambience became more turbulent. The effect was an entire environment that was Vger."

One of the more startling occurances during the process of mixing the last reel's effects was the discovery that lightning bolts were eminating from Vger during the climactic sequence. No mention had been made to the sound effects teams of their existence prior to this point.

"At two o'clock in the afternoon," recalled Howarth, "Richard turned around to Frank and I and said, 'I need a sound by four o'clock.' So Frank ran back to his equipment and I ran back to mine and it was do whatever you could do. Frank was the first one back. He came up with something and got it transferred, but I didn't get back until 9 o'clock that night."

Serafine had developed a rumbling, thunder-like sound, that was fine for earthbound lightning, but viewed as to familiar for Vger.

"I had done the opposite," continued Howarth. "I had worked up a sparking sound, but I hadn't had it transferred. So here's this whole movie hung up on lightning bolts for a whole day while Richard was there in the back taking these little pieces of tape with lightning bolt sounds and fitting them into the movie so they could finish dubbing the reel."

It was in this frantic air that the team concept for the effects creators proved to be valuable. It provided the editors with badly needed options as opposed to one sound that they may have been forced to use. Different creators supplied different points of view and the luxury of choice.

In the end, "Star Trek — The Motion Picture" was finished at the last possible moment, with both the first and the last reels of the picture being mixed on the last day. A courier from the Dolby Labs stood by at Goldwyn to take the completed reels of sound

— Star Trek director Robert Wise, producer Gene Rodenberry

to the lab for transfer to the optical stereo format found on the 35 millimeter release prints of the film that saw general circulation. A single 70 millimeter, six channel magnetic Dolby print was generated for the picture's premier in Washington, D.C. It later filled the screen in the Avco Theater, in Westwood Village, Los Angeles.

"Star Trek — The Motion Picture" did not mark the first venture for Robert Wise into science fiction. In the past he had directed "The Day The Earth Stood Still," and "The Andromeda Strain," and he commented on how sound for motion pictures had changed.

"We've really come so far with what can be done with sound and what the boys are able to achieve with the various instruments, synthesizers, Vocorders, and all kinds of other things that I had never heard of before I started on "Star Trek." The variations of the sounds that can be done impressed me tremendously. It has opened up a whole new field of possibilities for me as far as sound for the motion picture." □□□

MICROPHONE OFF-AXIS COLORATION

by BRUCE BARTLETT
Senior Development Engineer, Microphones
Shure Brothers Incorporated, Evanston, Illinois

The frequency response of a microphone that is most often published in a manufacturer's data sheet is the response to sounds arriving from straight ahead (on axis). However, while a microphone may have an acceptable frequency response on axis, it can have a different response to sounds reaching it from other directions. Even if a microphone measures "flat" on axis, at some other angles there may be an uneven or "colored" response. These variations from the desired on-axis frequency response, therefore, are called "off-axis coloration."

This article will take an in-depth look at the problem of off-axis coloration: its detection, causes, and solutions. Methods will be suggested to minimize this problem through judicious microphone selection and placement.

Examples Of Off-Axis Coloration

A recording engineer expecting a microphone to sound a certain way, based on its on-axis frequency response, may be in for some surprises or disappointments when the microphone is placed in use. Even though he usually intends to aim the microphone straight at the instrument(s) to be recorded, sounds are stil reaching the microphone from a variety of directions. The result might be an unwanted tone quality.

For example, a piano produces different spectra from different areas of the instrument. A closely-placed microphone "hears" the sound not only from the piano strings it is aimed at, but also from the other strings, sound board, and lid. These sounds arriving from off-axis may be colored.

Even the human voice, which is considered a point source, can cause problems. A performer using a hand-held microphone will not always stay on axis. They may vary their position by as much as 90°, which would result in tonal changes. Or, if stage actors are picked up by a distant microphone as they move around while talking or singing, the reproduced timbre of their voices may change if the microphone does not have the same response over a broad range of angles.

A microphone placed some distance from a musical ensemble will pick up room reverberation along with the direct sound from the instruments. This reverberation is a combination of sound reflections off the walls, ceiling, and floor of the recording room, which arrive at the microphone randomly from every direction. The "random-incidence" response of a microphone gives a good indication of the tonal coloration the microphone will impart to recorded reverberation. The farther a microphone is placed from a musical ensemble, the greater is the ratio of reverberant-to-direct sound. If the random-incidence response of a recording microphone is unlike the on-axis response, the effective frequency response of the microphone will change as the microphone is placed farther from the ensemble.

When several instruments are recorded simultaneously with multiple microphones, each microphone will receive off-axis sound leakage from instruments it was not intended to pick up. Poor off-axis response in any microphone can degrade the quality of the total pickup.

Coloration also can occur when many instruments are covered with only one or two microphones. For example, if a flat-response microphone is placed over a drum set, aiming straight down at the snare drum, the cymbals may sound harsh or dull since they are not on axis to the microphone. If the same microphone is placed in front of an orchestra, some instruments might be reproduced with higher fidelity than others, depending on their position relative to the microphone.

Clearly, it is important to know the response of a microphone over a wide range of angles. This data is seldom shown in published specifications, except in the form of a "polar-response plot" (also called "polar pattern" or "directional characteristics"). A review of this subject is offered to help the reader interpret polar-response plots in terms of off-axis frequency response.

Microphone Polar Response

A polar response is a plot of the sensitivity of a microphone versus angle of sound incidence. That is, the sensitivity of the microphone at one frequency or frequency band is plotted for every angle at which sound arrives at the microphone. Note: this is generally done in one plane since almost all microphones are symmetrical about their axis of greatest sensitivity. The polar response usually is drawn on polar graph paper,i.e., a graph made of a series of concentric circles, as shown in Figure 1. The outer circle is 0 dB, which is a maximum sensitivity reference. The next circle inward is -5 dB, next is -10 dB, and so on. Lines passing through the center indicate various "angles of incidence," or angles at which sound approaches the microphone. On the graph, 0° represents a sound source position in front of the microphone (on-axis); 90° is a position at the side; and 180° is to the rear of the microphone.

The higher the output of a microphone is at a particular angle, the farther from center is its polar pattern at that angle. For example, a typical measured cardioid (or heart-shaped) polar pattern is shown in Figure 1. The test frequency is 1 kHz. The highest sensitivity of the microphone is on axis at 0°, so the sensitivity is nearly the same up to about 45° off axis. Then the plot curves inward to a point 6

— *continued overleaf* . . .

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dB less sensitive at 90°. Continuing around, the sensitivity at 180° is about 20 dB below what it was at 0°. The polar pattern is symmetrical on each half of the circle. This pattern is uni-directional or cardioid; that is, most sensitive to sound coming from one direction (the front, or 0°).

Various other polar patterns are shown in Figure 2. The omni-directional pattern indicates equal sensitivity for all directions of sound incidence. Bi-directional or figure 8 indicates greatest sensitivity in two directions (front and rear). Also shown are two other common variations of uni-directional patterns: the super-cardioid and hyper-cardioid. These last two have rear lobes with greatest cancellation (lease sensitivity) not at 180°, but at other angles. Although the plots are drawn on flat paper, the actual polar patterns are three dimensional. This pattern could be generated by revolving the indicated polar pattern around the 0° - 180° axis of such plots.

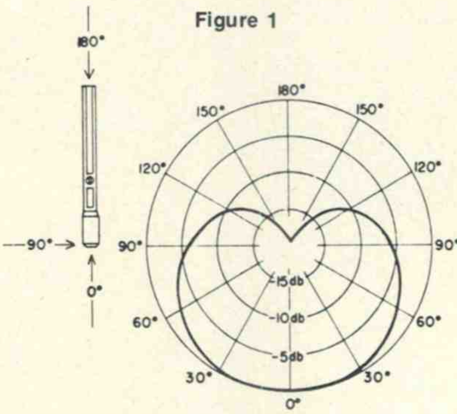
A polar response plot shows sensitivity at *one* frequency for *all* angles of incidence. A frequency response curve indicates sensitivity at *all* frequencies for *one* angle of incidence.

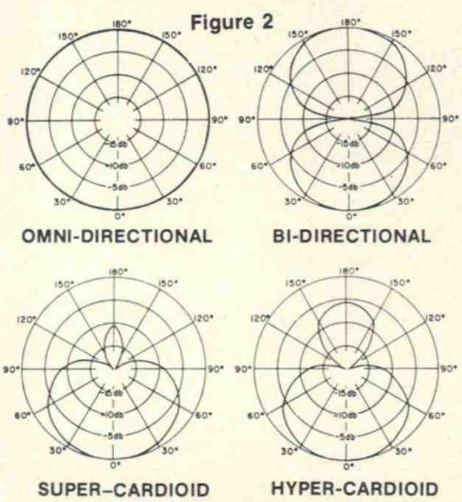
Frequency-Dependent Polar Patterns And Off-Axis Response

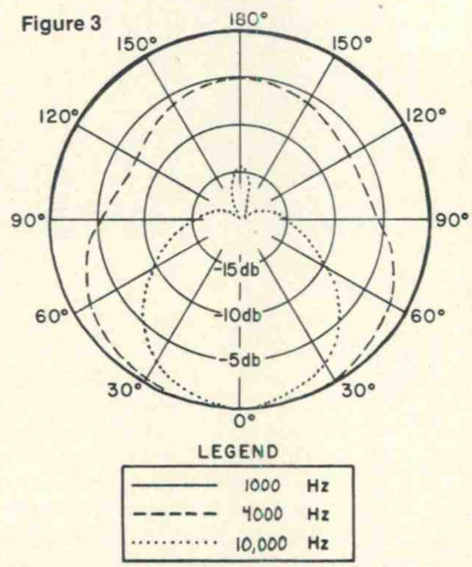
The polar pattern of most microphones is different at different frequencies. For example, a microphone that is omni-directional at low frequencies may be uni-directional at high frequencies. In other words, its polar pattern is a circle at low frequencies, but is narrower at high frequencies (see Figure 3).

What does Figure 3 tell us about the off-axis frequency response of this microphone? At a test frequency of 1 kHz (the solid pattern), the 90° sensitivity is the same as the 0° sensitivity. However, at 10 kHz (the dotted pattern), the 90° sensitivity is 16 dB less than the 0° sensitivity. These two data points roughly indicate how the 90° frequency response is similar to the 0° frequency response. The 90° frequency response is similar to the 0° response at 1 kHz, and is 16 dB lower than the 0° response at 10 kHz (Figure 4).

Figure 1: **Cardioid Polar Response Plot.**
Figure 2: **Various Polar Patterns.**
Figure 3: **Polar Pattern of a Microphone Which is Omni-Directional at Low Frequencies and Uni-Directional at High Frequencies.**
Figure 4: **Frequency Response of the Microphone Producing the Polar Pattern of Figure 3.**







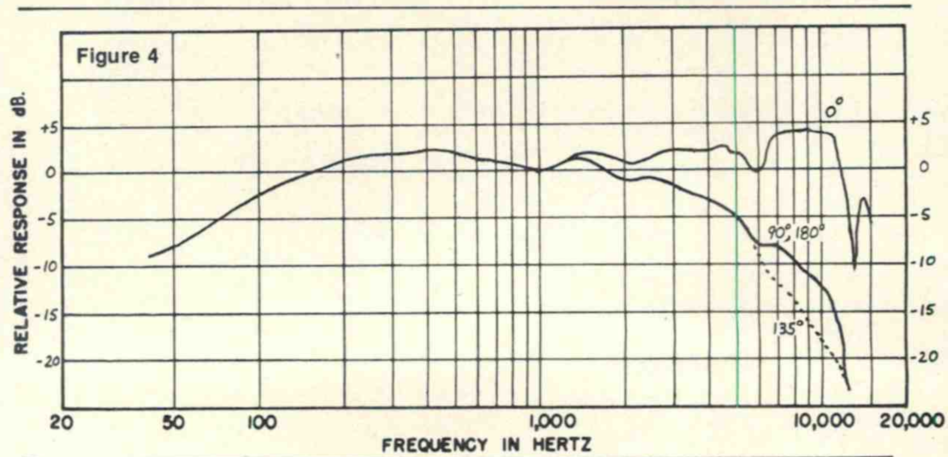


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Figure 5: An Omni Directional Polar Pattern Which is Nearly Uniform With Frequency.

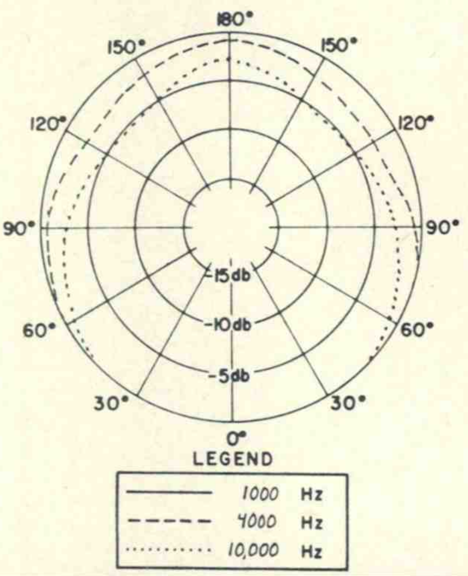


Figure 6A: Frequency Response and Polar Pattern of an Omni Microphone With Severe Off-Axis Coloration.

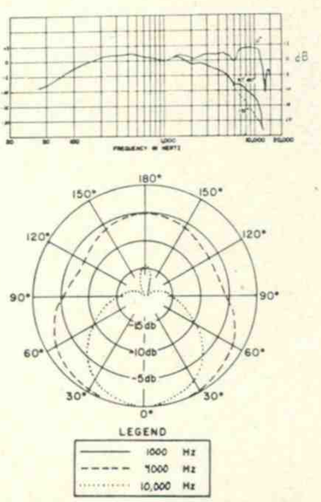


Figure 6B: Frequency Response and Polar Pattern of an Omni Microphone With Minimal Off-Axis Coloration.

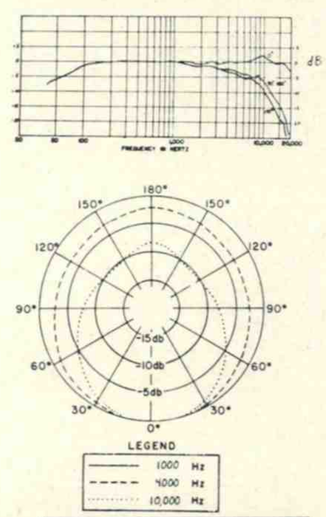


Figure 6C: Frequency Response and Polar Pattern of a Cardioid Microphone With Severe Off-Axis Coloration.

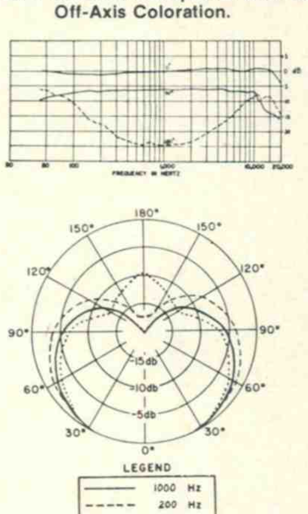
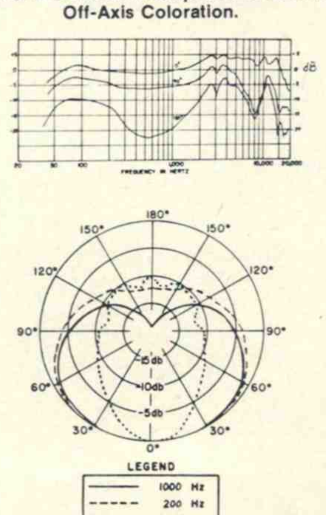


Figure 6D: Frequency Response and Polar Pattern of a Cardioid Microphone With Minimal Off-Axis Coloration.



Subjectively, this microphone should sound "duller" from the side than from the front. A source on-axis would probably sound quite acceptable, but the same source at the side of the microphone may lack brilliance. This can be verified by a simple talk test.

Now, suppose a different microphone is omni-directional. Its polar pattern at both low *and* high frequencies is a circle, rather than a narrower pattern (Figure 5). At 90°, the sensitivity is approximately the same at 1 kHz and 10 kHz. These two data points tend to indicate that the 90° frequency response is similar to the 0° frequency response. This microphone will sound about the same at 90° as it does at 0°, which is highly desirable for an omni pattern.

In general, if the shape of the polar pattern is approximately the same at all frequencies, the microphone will have about the same frequency response at all angles. Thus, it will not suffer seriously from off-axis coloration. Conversely, greatly different polar patterns at different frequencies indicate a microphone with noticeable off-axis coloration.

Figure 6 (A,B,C,D) shows the polar responses and the corresponding frequency responses of several microphones. For each microphone, the polar patterns at several frequencies are shown, along with the corresponding frequency responses at several angles. Note that the microphones with similar polar pattern shapes at various frequencies have similar (parallel) frequency-response curves at various angles.

Causes And Cures Of Off-Axis Coloration In Omnidirectional Microphones

As an aid to selecting a microphone with minimum off-axis coloration, it helps to know how sounds at different frequencies create different polar patterns. Polar patterns are produced by the variation in sound pressure on the microphone diaphragm as a function of angle. In other words, a polar pattern is a plot of the net driving force acting on the diaphragm versus angle of incidence. How does this force-variation-with-angle arise?

Let's start by looking at the forces acting on a single-diaphragm omni-directional microphone. The microphone diaphragm has two sides: the front which is exposed to sound, and the rear which is not. Sounds arriving from all directions excite just the front surface of the diaphragm. Even sounds from the rear engulf the microphone case and thus exert pressure on the front of the diaphragm.

Low-frequency soundwaves can bend around obstacles the size of microphones almost as if they were not there. So, at low frequencies, sounds from any direction excite the diaphragm equally. As a result, the sensitivity of the microphone is constant at all angles. This produces an omni-directional (circular) polar pattern at low frequencies.

At high frequencies, however, diffraction effects can disrupt the perfect omni polar pattern. *Diffraction* is the disturbance of a sound field by an obstacle. This disturbance is greatest at frequencies where the wavelength is comparable to the dimensions of the obstacle. With an obstacle the size of a typical microphone, diffraction effects start occuring above about 1 kHz and become greatest between 10 kHz and 20 kHz.

What are some effects of diffraction? At high frequencies, the microphone case obstructs sound waves arriving from angles toward the rear so as to reduce the sound pressure acting on the diaphragm. Consequently, the

for additional information circle no. 52

Figure 7A: Approximate Pressure Build-up Due to Diffraction for a 1" Diameter Omni Directional Microphone (averaged over the diaphragm surface).

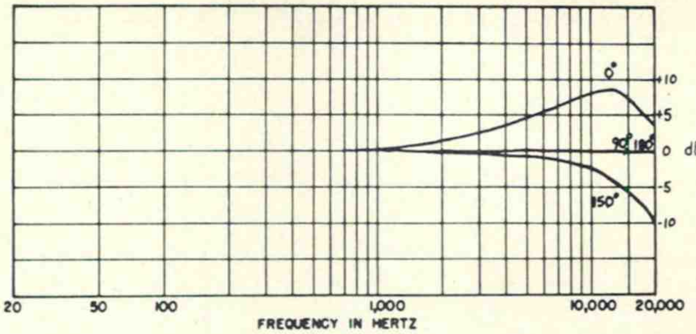


Figure 7B: Approximate Polar Pattern Produced by Diffraction at13.6 kHz with a 1" Diameter Omni Directional Microphone.

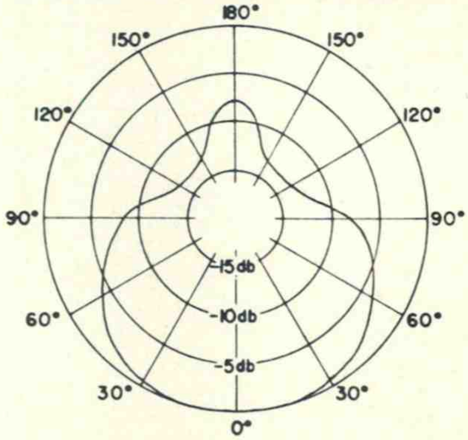


Figure 8A: Approximate Pressure Build-up Due to Diffraction for a 1/2" Diameter Omni Directional Microphone (averaged over the diaphragm surface).

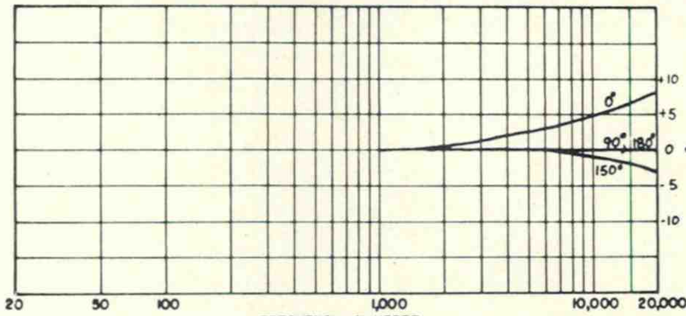
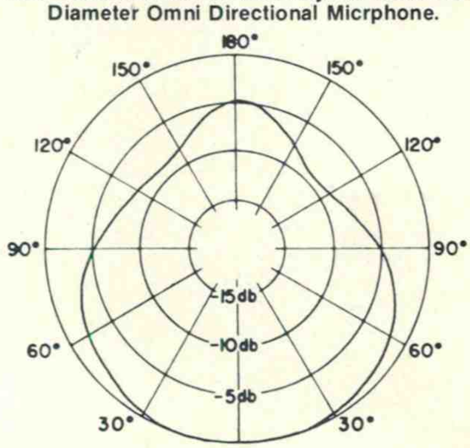


Figure 8B: Approximate Polar Pattern Produced by Diffraction at 12.6 kHz with a 1/2" Diameter Omni Directional Micrphone.



rearward frequency response of a microphone may be weaker at high frequencies than the front (on-axis) response. The polar pattern will then no longer be a perfect circle.

Other diffraction effects occur when high-frequency sound waves arriving from the front strike the diaphragm and are reflected off its surface. The reflected waves and the direct waves combine at the front of the diaphragm surface, increasing the pressure above what it would have been if the microphone were not present. At low frequencies, this reflection is insignificant. The net effect is a pressure boost at high frequencies, for sounds arriving on axis. There is less of this high-frequency pressure rise off axis. Thus, the microphone will be more sensitive to high-frequency sounds arriving from the front than to those arriving from the side. The subjective effect is a "brighter" sound on axis, a duller sound off axis.

Figure 7-A shows the approximate pressure increase on a 1-inch diameter microphone cartridge. The pressure increase is greatest at frequencies where the wavelength is comparable to the cartridge diameter. In this case, the greatest pressure boost is at 13.6 kHz on axis. At 90°, there is no pressure increase. At 150°, the microphone case obstructs high-frequency sound waves, causing a pressure decrease on the diaphragm.

The resulting polar pattern at 13.6 kHz is shown in Figure 7-B. Pressure is greatest on axis, so the polar plot is farthest from center at angle 0°. Pressure is less at 90° so the polar plot curves toward the center. Pressure at 150° is least so the plot is closest at that angle. The overall pattern approaches a super-cardioid characteristic.

So, the microphone which was omni-directional at low frequencies becomes a weak super-cardioid at high frequencies, due to diffraction. This narrowing of the polar pattern at high frequencies corresponds to a dulling of the sound off-axis.

Now, consider the pressure increase on a ½-inch diameter microphone cartridge (Figure 8-A). Maximum pressure boost is at 27.2 kHz. But, there is much less boost at 13.6 kHz than there was with the 1-inch diameter microphone. Consequently, the polar pattern is more omni-directional at high frequencies with the ½-inch microphone than with the 1-inch microphone (Figure 8-B).

The tone quality of the ½-inch microphone would not change as much off-axis as would the tone quality of the 1-inch microphone. Clearly, the smaller the diameter of an omni-directional microphone cartridge, the less the off-axis coloration, with everything else being equal.

Another cause of changes in frequency response with angle of incidence is the *phase difference across the diaphragm*, illustrated in Figure 9-A. Shown is the instantaneous pressure distribution of a low-frequency soundwave traveling across a diaphragm from the side (90° incidence). The wavelength is many times the diaphragm diameter, so the pressure at the outer circumference of the diaphragm is about equal to the pressure at the

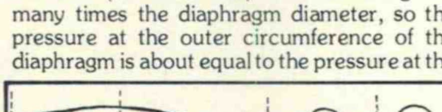


Figure 9: (A) Instantaneous Pressure Distribution if a Low-Frequency Sound Wave Travelling Across a Diaphragm From the Side. (B) Same for High-Frequency Sound Wave.

center. Thus, the average force acting on the diaphragm is relatively high.

At a higher frequency, where the wavelength is comparable to the diaphragm diameter, there is an uneven pressure distribution across the diaphragm (Figure 9-B). Here, the center of the diaphragm is at maximum positive pressure, and the outer circumference is at maximum negative pressure. These phase differences partially cancel each other, causing a considerable loss in the average driving force on the diaphragm. The higher the frequency, the less the average driving force.

This loss is greatest at ±90°, tapering off to no loss at 0° and 180°. So there is a variation in high-frequency response with angle, due to the phase-difference effect. The smaller the diaphragm diameter, the higher the frequency at which the average driving force begins to diminish.

In review, diffraction and phase-difference effects cause a high-frequency roll-off off-axis, relative to on-axis. The smaller the microphone cartridge diameter, the less off-axis roll-off (coloration). Note that some microphones have small cartridges inside large housings which, if properly designed, will be acoustically transparent.

Referring back to Figure 6-A and 6-B, the frequency responses at various angles and the corresponding polar patterns are shown for two particular omni-directional microphones. Note that the microphone in Figure 6-A has much more off-axis roll-off than the microphones in Figure 6-B. The polar pattern of the microphone in Figure 6-A stays fairly broad at high frequencies, but that of the microphone in Figure 6-B becomes narrow at high frequencies.

It may seem reasonable to make omni-directional microphones as small as possible to minimize off-axis coloration. However, the smaller the diaphragm diameter, the less operating force there is on the diaphragm. Thus, microphone sensitivity decreases as diaphragm diameter decreases, all else being equal. Microphone design becomes a compromise between maximizing sensitivity and minimizing size.

Since most omni-directional microphones have brightest frequency response on-axis with roll-off at other angles, the random-incidence response has a high-frequency roll-off compared to the on-axis response. Thus, the tonal quality of the recorded reverberation may be dulled. Some microphone manufacturers put a high-frequency peak in the on-axis frequency response so that the random-incidence (reverberant) response will be more uniform.

Microphone designs that minimize off-axis coloration also tend to have a random-incidence response which is similar to the on-axis response. Figure 10 shows the on-axis and the random-incidence frequency responses of a particular large-diaphragm omni microphone and a small diaphragm omni microphone.

Causes And Cures Of Off-Axis Coloration In

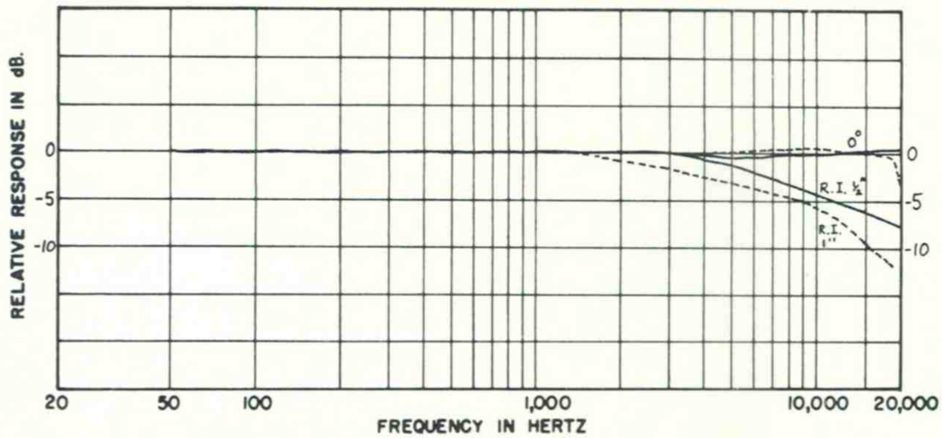


Figure 10: 0° Response and Random Incidence Response of a 1/2" Diameter Omni Condenser Microphone (solid line), and a 1" Diameter Omni Condenser Microphone (dotted line).

(Source: Bruel & Kjaer Instructions and Applications Manuals for Models 4133 & 4131, June, 1971.)

Unidirectional Microphones

As an aid to understanding this section, preliminary discussion of unidirectional polar pattern formation is helpful. Although a single-diaphragm omni-directional microphone has only one sound entry to the front of the diaphragm, a single-diaphragm uni-directional microphone has more than one sound entry. Both the front and rear of the diaphragm are exposed to sound pressure. The rear entry leads to an acoustical phase shift network made of screens, tubes, and cavities. This network delays low to mid-frequency sounds and attenuates high-frequency sounds reaching the rear of the diaphragm. This acoustical signal processing contributes to the formation of the directional polar pattern.

At low to mid-frequencies, the front and rear pressures partially cancel each other. This cancellation is made to vary in a controlled way as the sound source angle varies, so that a particular directional pattern is produced. At high frequencies, the sound pressure on the rear of the diaphragm is greatly attenuated so that the frontal pressure prevails to move the diaphragm. Thus, at high frequencies, the microphone acts like an omni-directional unit. In this audio range, the polar pattern is achieved the same way as it is with an omni-directional microphone: by diffraction and phase differences across the diaphragm.

Referring back to Figures 6-C and 6-D, the responses and polar patterns of two uni-directional microphones are shown.

Uni-directional microphones tend to have a flatter random-incidence frequency response than omni-directionals, since the off-axis sensitivity for uni-directionals is diminshed not only at high frequencies, but also at low to mid-frequencies. In an ideal cardioid microphone, the random-incidence response is 4.8 dB below the level of the on-axis response and is fairly similar to the 90° response. Figure 11 compares the random-incidence response of a particular uni-directional microphone to an equal-sized omni-directional microphone and to a double-diaphragm omni-directional microphone.

Obviously, a uni-directional microphone will have more cancellation of off-axis sounds than on-axis ones. The design objective is to have all frequencies cancel equally, at a particular angle of sound incidence. For example, if all frequencies are canceled by 6 dB at 90°, the 90° response is similar to the 0° response. Then no off-axis coloration occurs at 90°. But if 1 kHz is canceled 6 dB, and 100 Hz is canceled 8 dB, and 10 kHz is canceled 12 dB, the 90° sound quality is different than the on-axis sound quality. What are some causes and cures of uneven cancellation at various frequencies?

The uni-directional microphone operates best with carefully controlled elements in the acoustical phase shift network. However, the values of the network elements vary with frequency and internal standing waves can develop. These effects degrade the desired polar pattern at certain frequencies. The dimensions and damping of the acoustical elements must be adjusted to provide phase shifts which, in combination with diffraction effects, perform as desired at high frequencies.

MICROPHONE OFF—AXIS COLORATION

The performance of a directional microphone also depends on well-behaved pressure and phase differences between its front and rear entries. However, diffraction can cause a nearly unpredictable distribution of sound pressure and phase around the microphone cartridge, disrupting the ideal performance. So, in a microphone design, the sound entry configuration and placement are critical for optimum results.

As mentioned before, high frequency polar characteristics are controlled by diffraction and phase cancellation effects across the diaphragm. These effects boost the pressure on-axis and reduce it off-axis at high frequencies. The design objective is to match the amount of high-frequency roll-off produced by diffraction, with the amount of low to mid-frequency cancellation provided by the acoustic phase shift network.

For example, if the low-frequency cancellation at 90° is 6 dB, the diaphragm should be dimensioned so that the 90° roll-off due to diffraction is also 6 dB. Usually this calls for a small-diameter diaphragm. However, at 180° (that is, at the rear of the microphone), the mid-frequency cancellation may be 20 dB, so a 20 dB roll-off due to diffraction would be desired at this angle. This would require a large-diameter diaphragm or, at least, a large microphone handle to block sound from the rear. As in the omni-directional microphone, sensitivity decreases as size decreases, all else being equal. So, as earlier stated, microphone design often becomes a compromise between maximizing sensitivity while minimizing size.

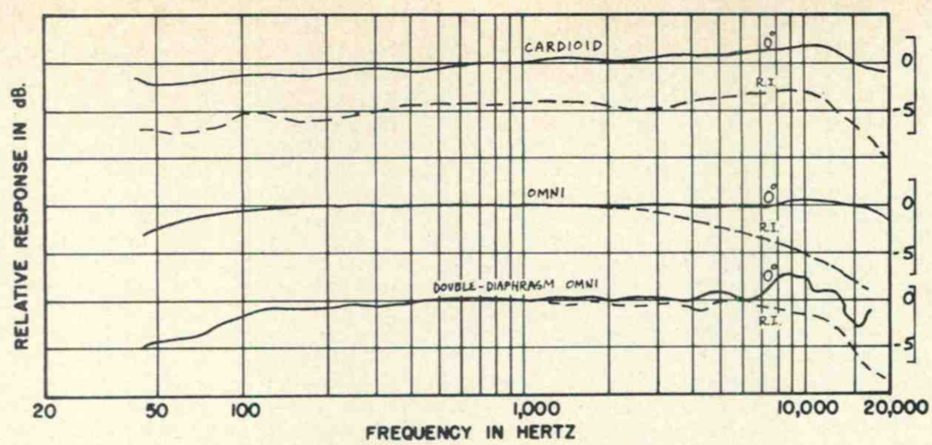


Figure 11: 0° Response and Random-Incidence Response For A Cardioid, Omni and Double-Diaphragm Omni Microphone. (All 3/4" Diameter)

Referring back to Figures 6-C and 6-D, the responses and polar patterns of two uni-directional microphones are shown.

Uni-directional microphones tend to have a flatter random-incidence frequency response than omni-directionals, since the off-axis sensitivity for uni-directionals is diminished not only at high frequencies, but also at low to mid-frequencies. In an ideal cardioid microhone, the random-incidence response is 4.8 dB below the level of the on-axis response and is fairly similar to the 90° response. Figure 11 compares the random-incidence response of a particular uni-directional microphone to an equal-sized omni-directional microphone and to a double-diaphragm omni-directional microphone.

Off-Axis Coloration In Bi-Directional Ribbon Microphones

This type of microphone uses no phase-shift networks, and it employs a very narrow vertical metallic ribbon as a diaphragm. Consequently, diffraction and phase-difference effects are minimal. Polar pattern shapes are constant up to a very high frequencies in the horizontal plane. Since the ribbon is much longer vertically than it is horizontally, some pattern narrowing occurs in the vertical plane (Figure 12). Fortunatley, instruments are usually arrayed horizontally around a microphone, so off-axis coloration is very minor with this type of design.

Microphone Selection And Technique To Minimize Off-Axis Coloration

The degree of pattern control in a microphone depends on the design, and is not always predictable by appearance. However, microphones with small diaphragms (including ribbons) tend to exhibit less off-axis coloration than large-diaphragm microphones. It probably is best to go by results and choose a microphone which tends to maintain similarly shaped polar patterns at all frequencies.

With multiple-instrument coverage by one or two microphones, instruments that are affected most by off-axis coloration should be placed on axis, if possible. Many microphones have an off-axis coloration that is a high-frequency roll-off. Consequently, instruments with spectra strong in high frequencies (such as percussion) should be placed directly in front of the microphone. Bass instruments can be placed anywhere around the microphone with little change in timbre.

In multi-microphone coverage of a drum set if, for example, the snare drum microphone is intended to pick also up the high-hat cymbals. It should be aimed somewhat toward the high-hat so that its high-frequency "sizzle" is not lost.

Sometimes a single microphone is used to pick up a round table conference discussion. Placing it so that the diaphragm plane is horizontal will make it pick up all participants with the same timbre, although the sound may be dull due to off-axis high-frequency roll-off. Some electronic high-frequency boost may be required to restore a natural tone quality. The same technique can be used for instruments surrounding an omni microphone.

When several instruments are recorded

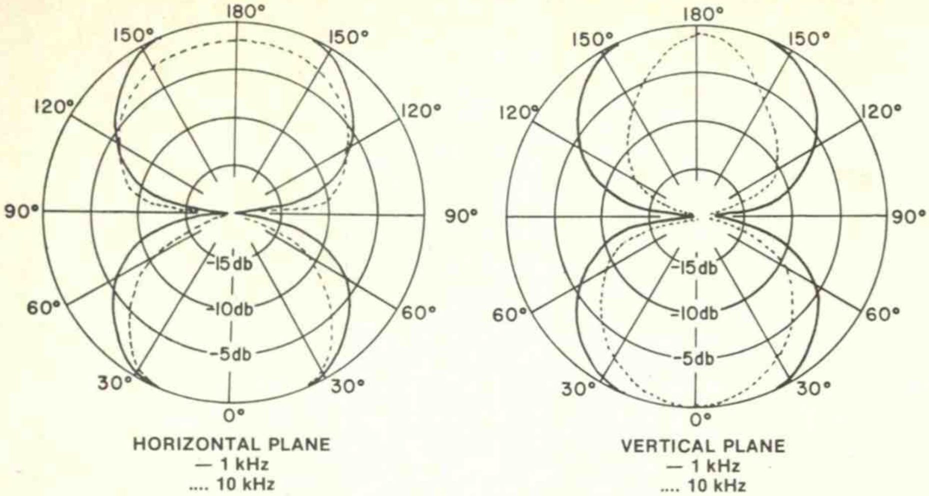


Figure 12: Bidirectional Ribbon Microphone Polar Patterns

simultaneously with individual uni-directional microphones, each microphone still receives off-axis sound leakage from instruments it is not intended to pick up. By aiming the rear of each microphone at nearby unwanted instruments, the audibility of the off-axis coloration can be reduced. Placing the microphones closer to their intended sound sources accomplishes the same thing.

In overdubbing situations, where leakage is not a problem, the recording microphone(s) sometimes can be placed relatively far from the instrument. This places more of the instrument's radiating area within an "acceptance angle" of uncolored response. For example, an acoustic guitar picked up by a closely-placed microphone may sound harsh, due to sounds from different areas of the instrument arriving at the microphone from wide angles. Placing the microphone farther away will pick up the entire guitar within a smaller angle which is more on-axis.

Conclusion

Microphone off-axis coloration can produce unexpected and undesired tonal degradation in a recording. However, with some understanding of the causes of this problem, and with some knowledge of proper microphone selection and placement, the sound engineer can minimize the effects of off-axis coloration. □□□

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Facility open daily during summer season. Call for appointment to inspect stage.

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Orchestra pit: 60' wide x 15' deep seats 92. Can be manually raised by adding scaffold. First row is about 6' from edge of pit.

Stage to proscenium height is 23'. Actual proscenium width, 54' 6'.

Moveable pipes located above stage are available for hanging monitor speakers or mikes.

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Main loading door (10' x 10') is located at end of corridor backstage left and opens onto Vermont Street.

Dock is about three feet above street level to facilitate unloading from semi's.

Another loading door is at stage level 3' above ground off stage left and measures 18' high x 8' wide. An elevator with a 10' x 10' door goes to stage level from ground level. Small equipment van can drive up to stage left loading door and elevator. Truck parking available in front of theatre on Vermont Street.

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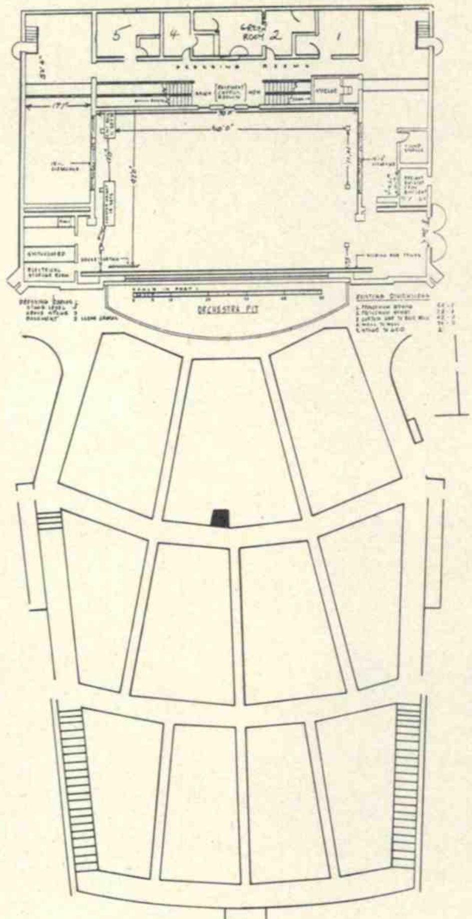
House system consists of two large side stacks on drapped scaffolding and a center overhead cluster.



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- house console -

Will accept balanced line level signal with a XLR 3-pin connector.

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"Clean, even sounding with good transparency. Not a dynamic system due no doubt to 95 dB City sound limitation. Therefore, PA basically reinforces vocals and supports sound coming off stage. You can hear everything, but physical feel is inhibited. House sound system has a good, gentle sound with no apparent horn influence. Mixing area is a little cramped and could do with a lower protective wall." ***Perry Cooney,*** *Britro, Inc., New York (Blondie).*

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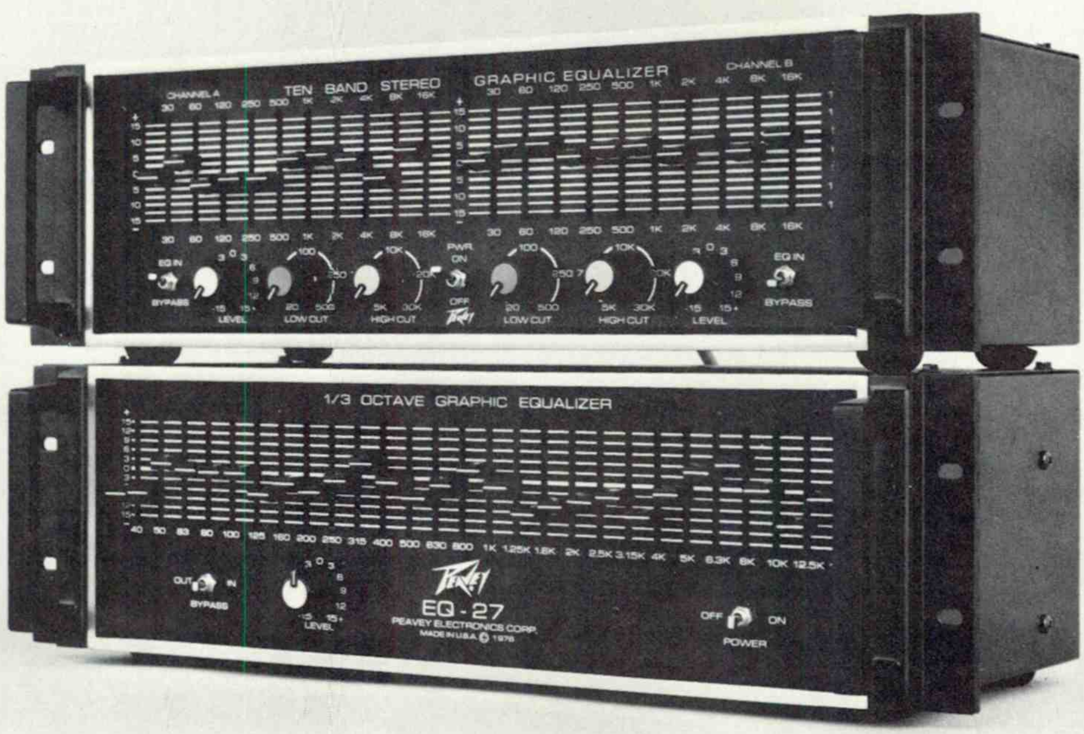

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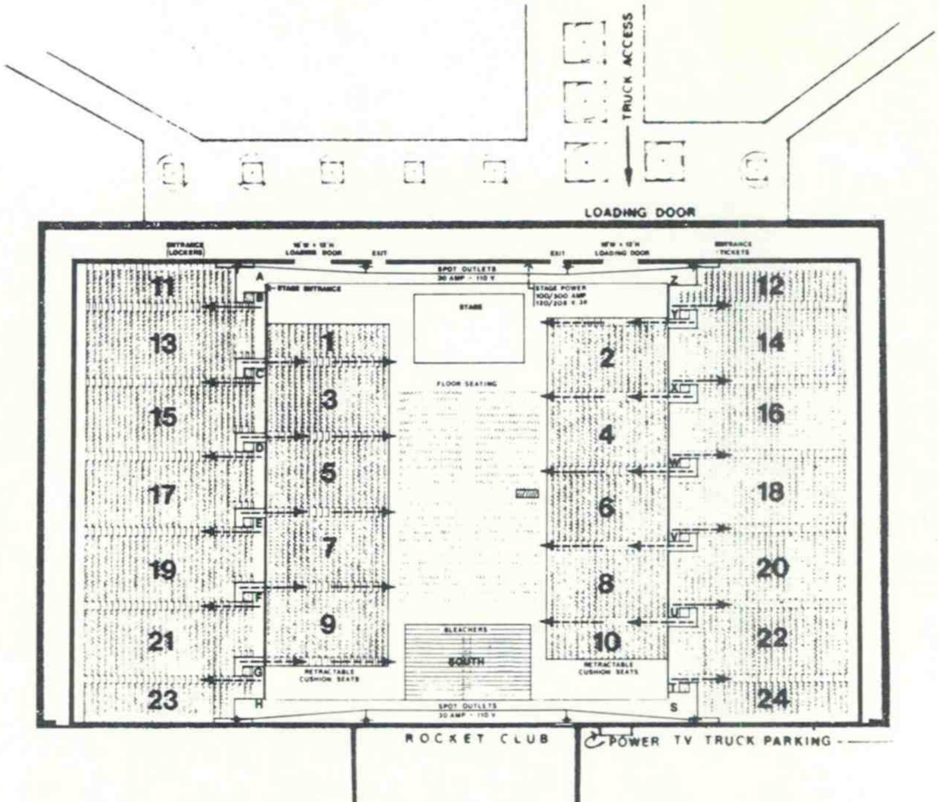

CENTENNIAL HALL
University of Toledo
2801 W. Bancroft
Toledo, Ohio 43606
(419) 537-2825
(419) 537-4338 (Backstage)

Directions

From closest airport: Take Toledo Express Airport Highway (2) into Toledo; left at Reynolds Road; right at Dorr Street; left at Secor Road; right at West Bancroft Street; right at Towerview Boulevard (East); left at Stadium Drive over Ottawa River. Take immediate left to Hall.

From closest highway: From Ohio Turnpike (80) get off at Exit 4 and head north on Reynolds Road. From here follow above directions.

From Toledo-Ann Arbor Expressway (475) exit at Toledo Express Airport Highway (2), head east to Reynolds Road and follow above directions.

Facility

Open from 8 a.m. to 5 p.m., Monday through Friday. No specific closing time for shows. Rectangular shaped coliseum hall will seat 9,294 when stage is at north end; 9,608 when set up for shows "in the round." Level stage size is variable and is assembled from SICO 6' x 8' x 4' high sections. A width of 68' is possible if seating plan is modified.

Distance from apron to first row of seating: 6' - 8'.

Height from stage to grid: 52'.

Acoustics

2,736 seats are padded; the remainder are hardbacked. Overall acoustics are said to be very good for this type of hall although speaker deployment is a key factor for good sound quality here. Some slapback possible off window (15' H x 75' W) located on the south wall about 150' away from the stage. Position speakers to avoid this window if possible.

No RT_{60} reverb measurements available. However, reverb time is less than normal for this size hall due to spray insulation and carpets on all walls.

Loading

Loading door located at N.E. corner of building. Size: 12' H x 16' W. Smallest opening between loading dock and stage: 12' H x 16' W. Parking available for trucks in N.E. parking lot #3. Direct access to stage at arena floor level (4' below stage level).

Set Up

Sound console is generally set up at floor level in center aisle at house right. Distance to stage: 75'. Length of cable to center stage: 100'. Closest AC: 100'. Run cable under retractable seat section to stage.

If speaker wings are used, some sightlines will be affected. Hall will generally block off seats at sides of stage until stage is set and sightlines are determined. Approximately 1,000 seats can be affected. Risers are available on which to stack speakers.

Riggers and equipment are available to hang speakers. Contact hall in advance as total loads are analyzed on an individual show basis according to established hall guidelines.

House Sound System

Overhead speaker cluster in center of building uses two-way active crossover. System includes: 4 Dukane 6A405 bass cabinets; 4 Dukane 6A415 bass cabinets; 4 Dukane 6A436 bass cabinets; 12 Dukane 5A530 radial horns; 12 Dukane 5A421 exponential horns; 12 Dukane 5A422 exponential horns. All high frequency drivers are Dukane 5A540s. System is powered by a total of 3,800 watts. Provides very good sound for sports events and some MOR-type music shows "in the round," but was not designed for high level music reproduction. System utilizes 2 Dukane 3A235 1/3-octave filter sets; one for speech, the other set for music. Two AKG D-224E and 2 AKG D-558 mikes are available. There is no in-house monitor system.

Main house mixing console is a Dukane Model 2A75 with 5 mike or line inputs and 4 outputs. Two band EQ on master outs only. Console will accept a balanced line level signal on an XLR 3-pin male connector.

There is a separate charge for the use of this house system.

Electrical

Three 3-phase 120/208Y supplies available at 100A, 300A, and 600A portable respectively for 3,000 amps total. Main breaker box located 10' from stage requires pigtail connectors.

Personnel

Non-union house.

Building Manager: Mike Barber, (419) 537-2825.

Operations Manager: Don Warner, (419) 537-4338 or 537-2825.

Piano tuner available: Donald Seeman, (419) 242-5246.

Traveling Soundman Reaction

"I did Andy Williams there 'in the round' using the house PA cluster. It was adequate for this kind of show, but I couldn't punch a lot of level through it. It's definitely not suitable for rock 'n' roll. I mixed from the bleachers at eye level with the speaker cluster. Acoustics are average for this type of place — not great; not terrible. The sound did improve noticeably with a full house. Acoustics and overall PA quality similar to the San Francisco Civic Auditorium. Crew of University students is good — no problem getting in or out." ***Ken Cox,*** *Independent Engineer.*

FEEDBACK ISN'T ALWAYS A DIRTY WORD!

The information contained in these surveys is as accurate as possible at the time of printing. However, there will always be changes and improvements made to in-house sound equipment and acoustics as more and more venues become conscious of high quality sound. Many of these halls rely on the comments and reactions of visiting engineers such as yourselves and make changes accordingly. So, if you should come across a situation that you feel is contrary to what is printed, please drop me a note and I'll print an update. Also, if you have any reactions to a venue (pro or con) that you'd like to see surveyed in an upcoming issue, please address them to:

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RECORDING engineer/producer
P.O. Box 2449
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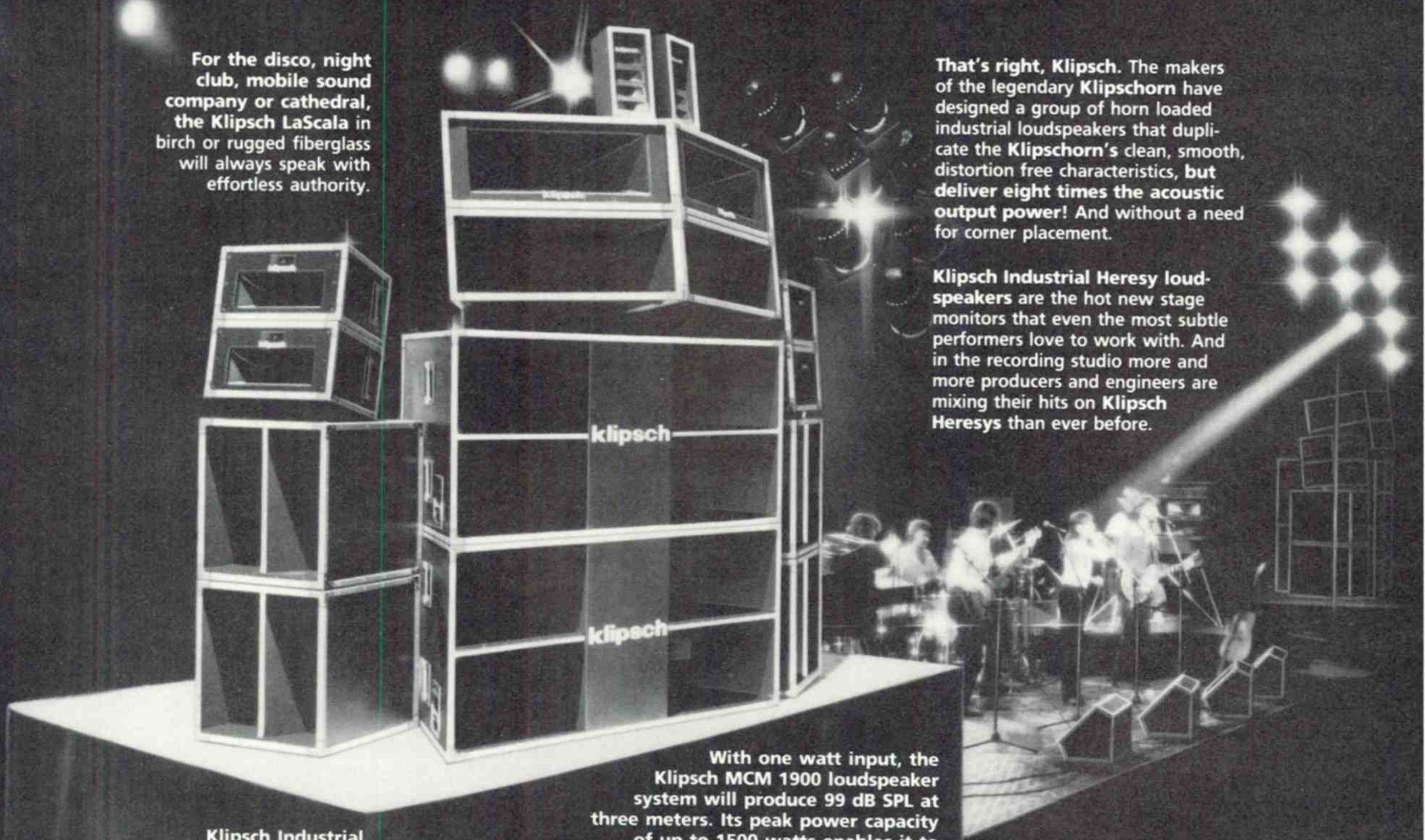

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. . . thoughts and experiments that may lead to a re-evaluation of the revered and established basis of psychoacoustics —

HASSLING HAAS

by

STEPHEN ST. CROIX

We all grew up in an analog world; a world of 'natural' laws of acoutics and acoustic events, a world where all overtones and distortion products are directly (harmonically) related in some fashion to the signal that is generating them.

Further — all pitched events occurring in nature, by that I mean all events which contain perceivable periodicity, such as a dog barking, a train whistle, or human speech have a tracking set of harmonics or overtones in a reasonably simple combination of even or odd relationships. This simple rule is very important when it comes to developing a good understanding of the brain's pattern or event recognition systems. This rule of related tracking harmonics is important both in it's existence and in it's exception: METAL. What? Yes; metal — if you hit a chunk of metal, a bell, a cymbal, or a 10 x 10 x 10 chunk of bronze machine stock, you get a series of overtones that do not relate to the fundamental in any simple way. This condition of an impact excited mass producing primarily bastard harmonics is not actually unique to metal, as hitting a rock with a hammer, for example, will also produce a harmonic package rich enough in bastard overtones to surpress any sensation of true pitch.

There are, of course, exceptions to these statements, too. One can say that coffee cups, plastic letter in/out trays or chunks of firewood also have bastard hamonics, while a tuning fork or the wood blocks of a marimba do have a clearly perceivable pitch.

These statements are all true, but one must realize that it is the long term listening experience that is important in pattern recognition; not the occasional exception.

the author —

Stephen St. Croix is the president of Marshall Electronics, and is the designer of the Marshall Time Modulator series of delay processors. He is also a studio musician and engineer.

He has recently completed a private studio (front cover) for the purpose of completing his first solo album and to aid in his current work for other artists and movies.

He has done psychoacoustic research in the areas of underwater medicine, air traffic control, defense, and music.

A Sound By Any Other Name

The psychoacoustic process of hearing, and interpreting what you hear, is, of course, a chain of events starting with the generation of a signal by natural physical excitation or the electrical excitation of electromechanical transducers producing an encoded output (compression and rarification of the atmosphere surrounding us).

This encoded information is then captured by the outer ears, where the first signal processing or conditioning takes place. The small folds and bumps (known medically as small folds and bumps) in the outer ear serve to impose complex comb filter responses on the incoming signal, as a function of interception angle. Various frequency components of the signal presented to the outer ears are attenuated or reinforced by reflection and acoustic recombination. The end result is that the signal then has a series of comb filter notches and peaks superimposed on it.

To get a better feeling for how this works, imagine several flangers set up in series, some set for very deep flange, and others set for a less pronounced flange. These flangers are not set to the same delay times, but several differeent delays. They are not sweeping, they are just sitting there generating their assigned comb filter notches. The units are then slaved to one master control that can change the delay times of all of them while maintaining the exact relationship that they started out with. You now have a table top covered with flangers, and a model of the outer ear.

Far From Flat

Of course, from this point on the frequency response of the receiving system (your chain of flangers or the outer ear) is far from flat as it has all these notches cut into it. The placement and relationship of these ntoches is a function of the angle of interception of the acoustic signal by the outer ear, or the position of the master control for the chain of flangers.

This is the first signal processing done by your body; the generation of varying frequency response curves; primarily as a function of the location of the sound source relative to the position of your head. As the sound source moves, your system's frequency response is altered.

The original audio information along with its newly generated location information is then presented to the eardrum where through a series of levers that we all learned about in school it is coupled to the inner ear. Here, in a much more efficient liquid environment, a respectably rapid analysis of amplitude versus frequency is done along with some very impressive phase analysis. This information is then outputted to the brain chemically/electrically for the next stages of processing. This ends the mechanical signal conditioning.

The next step in processing (if things in there are reasonably normal) is a comparison of the input from the two receiving transducers (ears). Factors that receive priority are overall amplitude and the ratio of amplitudes of the signals presented to each ear, time differences between when each ear received a common signal, the placement of the notches that the outer ears generated as a function of receiving angles, and equally as critical; what you EXPECTED to hear.

You Are What You Hear

This is where psychacoustics really start. All this great flow in encoded electrical input to the brain doesn't mean a thing unless there is a set of decoding algorhythms and look-up tables to process against.

Understanding what we hear is an eductional process. We learn through accumulating a lifetime of acoustic events, analyzing how they relate to the ones we have heard before, and how they relate to what we see and feel at the time we hear them. Each and every time that we hear something this process of correlation and reinforcement is repeated and filed away in a reference bank for look-up and comparison later again.

Actions and reactions become linked and triggered by recognizing acoustic event patterns. We hear one sound and we know that it is a violin, about ten feet in front of us and a bit to the left. We hear another sound, and we know that it is voice communication, and we understand it (sometimes).

For example, let's say that you are driving to work in the morning and suddenly you hear a particular sound. You immediately and automatically know that a person next to you is speaking to you, in fact yelling at you, claiming that you have just cut him off at the last intersection without looking. From the noises that this guy is making (the acoustic event that he is generating) you know that he thinks that you have done something to him, (cut him off) and that he is very upset with you, and that it probably would be a good idea to roll up your window and lock your door. You may even make surprising discoveries about the legal arrangements that your parents had when you were born.

A lot of information has been collected from a one-and-a-half second sample of audio input.

This information was extracted from the sound by your brain comparing the encoded audio input derived from your ears to a look-up table — finding the proper matching sounds, and calling up the proper meaning to the words while examing the harmonic structure, comparing again to look-up tables, and telling you that this is a person (of

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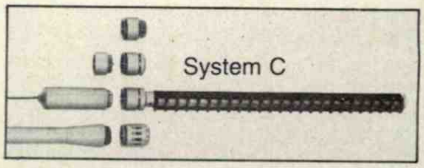

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sorts), and further a full grown male, and he is at the moment (judging from the overall amplitude and rich harmonic structure of this audio sample) capable of possibly inflicting physical harm upon your engineer or producer body.

An amazing amount of information has been extracted by use of mental search and compare techniques that are usually completely forgotten or taken for granted.

It should seem clear now that if we have the power to compile this magnitude of past experiences in a look-up table and instantly do an extensive search and retrieve to do work and emotion recognition, we also have quite an impressive library of acoustic conditions and events in there, too. In fact, it is next to impossible to generate a sound electronically with a synthesizer, let's say, without the brain going into search mode and immediately coming up with some past acoustic experience that it "sounds like." The strangest sound that you can come up with will still yield some correlated event, even if it is a weird "science fiction" type space sound. The listener will generally experience a weird "science fiction" type space feeling; by automatic correlation with previous science fiction films that he may have seen or by correlating this sound with other similar sounds that he has heard in his life that have remained a mystery to him as to their origin, such as the strange "singing" that telephone lines sometimes make in the wind, or the unexplained howls and wails that he may have heard on his first night camping out in the woods as a boy.

These sounds are logged in the "unexplained" files, and if the correlation factor with the sounds now being synthesized is high enough, are sure to trigger to some lesser degree those same feelings of panic and awe that were originally felt and stored in direct relationship with the original acoustic event.

From Scared In The Woods To Better Music

All this is directly applicable to the real world or recording studio applications. In fact, whether it is consciously realized or not, it is critical to them. That sometimes all too evasive factor in a piece of music, called "mood," is, of course, a function of such obvious things as modality (is the piece written with minor sixths?), tonality, tempo, instrumental choices, and so on, but it is also very important to not forget the factor of evoking predictable emotional responses by providing acoustic information for the express purpose of correlation with the listener's past emotional experiences; most of which can be recalled and re-experienced with carefully formed audio triggers.

Throughout history, the great composers have been striking these deep psyhic chords in us, though I suspect without conscious specific knowledge of how and why this works.

Today we are sort of blindly searching for that same skill or power, and usually never really finding it. Done with enough skill, the concept of evoking desirable emotional conditions or reactions to a piece of music by inserting the proper audio information to trigger these desirable correlations is amazingly reliable. The trick is to think it out and do it consciously, not to leave it as a little understood skill that some musicians, composers, and producers can get to work (sometimes). Not only is it legal, but it is and always has been considered brilliant when achieved.

In my work I have discovered that, as an example, you can quite reliably build clear anxiety and awe in a section of score for a science fiction movie where it is wanted simply by constructing synthesizer voicings that contain a small bit of the proper correlatable audio. It usually doesn't even have to be enough to become consciously noticable; just enough to start the process and the brain will do the rest.

I recently found myself faced with an interesting problem. After mixing down a song which was to have a section in the middle that produced a strong feeling of a sort of "cosmic or religious" power and magnitude, I discovered it to be very clearly missing. As this was a rock and roll tune, it was not realistic to actually score this section with the major block chord progressions that will typically produce those feelings, so I was left with the dreaded "Fix-It-In-The-Mix" approach.

A little thought yielded the concept of going for the giant stone church effect. You can't get that by just slopping on more reverb, so how do you get it? I thought about what music sounds like in an ancient Roman stone cathedral and why; how the hard surfaces and large spaces do unique things to high frequency versus low frequency decay times, and planned to call up that association in my listeners.

The rest was easy, and amazingly effective. I put a Bode Linear frequency shifter on the reverb send and shifted everything up a slight bit. Then I mixed the two returns together and fed a little back into the input of the frequency shifter through a Marshall Time Modulator set at around seventy milliseconds of delay. This, of course, set up a feedback loop that sharped the reverb and linearly increased the harmonic spacing as it decayed.

It was not enough to be heard as a detuned condition, but it must have satisfied the conditions to trigger the proper search and correlate pattern, because it worked, without changing the music or the mix. As a sidenote, I also discovered that shifting down produced a heavy, depressed feeling.

You can rely on the brain's power to search and retrieve in another way. Whether done consciously or not, the trick in electronic synthesis is to provide enough correlation, that is accumulate enough factors when trying to synthesize an acoustically natural event (say a trombone) that do exist in the real audio package delivered by a trombone so that your mind sees a reasonable percentage of what it is looking for to identify that sound as a trombone in what it hears.

When a high enough percentage (correlation factor) is reached, the brain identifies: Aha! Got it — a trombone! Once this has happened the brain will actually fill in, that is provide the missing information to complete the trombone acoustic package.

This system of early identification triggering a filling-in function enjoys an amazingly good track record. In fact, it is so good thaqt many people cannot ever remember it making a mistake or it being fooled. If it were often fooled the recognition process would just become more and more intricate, and, of course, slower and slower until it again became reliable. As it is, however, we possess a recognition system that is not only 'instantaneous' in daily application but is essentially never confused.

Those of us in the recording industry have all experienced these confusions though — while the general public probably hasn't.

The Black Wire Goes On The Red Lug

A simple thing like reversed L/R program phase entirely destroys our sound source or image location ability — as we do not have the education (information in our look-up tables) to deal with it. If this situation were to occur in nature often enough we would probably build up the proper look-up tables and we could then deal with it.

Those of you who scuba-dive may be aware that we have an almost useless sense of sound source location under water.

This is due to the fact that sound travels a great deal faster in water than in air, in fact, at about 4,800 feet per second, compared to the sea level air speed of about 1,130 feet per second. It is so much faster than in air that it is outside of the capabilitites of our detection and look-up system.

The delay time generated by the distance between the two ears is shortened to the point of uselessness, and the comb filtering that the outer ear is designed to generate is now shifted beyond usability.

Without these systems to aid in the computation of sound source location, we do a dismal job of it. This is because in a normal land-bound life, we have never had to deal with sound traveling that fast, so we have not developed the look-up tables, nor is the outer ear proportioned correctly to give useful phase difference information at that speed.

Because our recognition system enjoys such a track record (it is essentially never wrong) it has gotten away with sub-grouping and early branching techniques in the interest of simpler processing and faster recognition.

Not only does the system fill in the information that's not there when it determines that enough properly correlated information to identify the sound does exist, but it will actually throw away or suppress information that it sees as wrong once it has made its decision, identified and categorized the sound (actually it just doesn't bother to process some of this 'garbage').

Even harmonic distortion is a good example of this. While a given amount of even harmonic distortion may be unnoticable, the same level of bastard harmonic distortion would fall well outside of the trained acceptable window of naturally possible events, and be clearly noticed. It is generally felt, for example, that a given amount of second harmonic distortion is far

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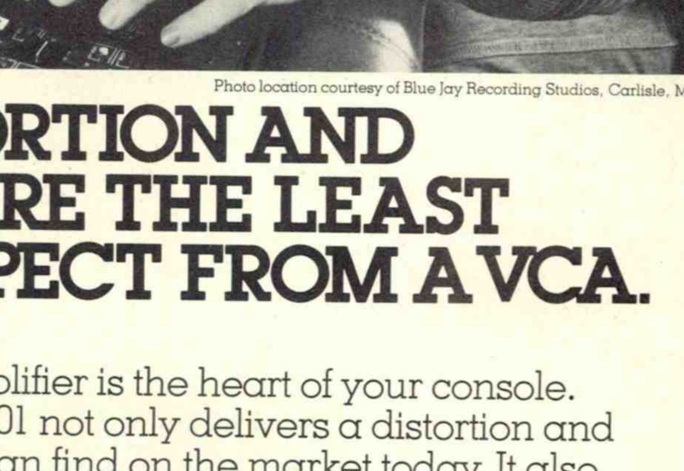

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less objectionable than the same amount of third or higher order distortion. It is felt that the second harmonic distortion is more completely "masked." This effect of masking is a psychoacoustic event generated by such things as the limitations of the actual inner ear pressure to the electrical transducer network and, once again, past education reinforcing the high probability of lower order even harmonics existing in nature.

Bite The Byte

It is this phenomenon that explains why even very low levels of distortion in digital delay lines and other digital processing equipment are immediately heard and so objectionable to many. The distortion products of these devices are primarily sideband products caused by interaction between the audio and the quantized samples taken as a given clock or sample frequency. They therefore bear no direct harmonic relationship to the original input signal and, in fact, are very similar to what you hear when you hear single sideband radio transmissions. This is a condition that almost never appears in nature, except in the earlier example of struck metal, and therefore is picked up and processed as correlating to the only thing that you have in your look-up tables that possess a similar structure: metal.

To make matters worse, these devices increase distortion as the signal level is decreased, again an effect that does not occur in nature. Perhaps this is why so many find the sound of digital delay lines to be "harsh or cold."

The psychoacoustic interpretations of equipment side effects should be considered all the way from the point of designing equipment to sweetening the mix if you want the highest probability of controlling the listener's reactions.

As discussed above, our recognition systems are very touchy in some areas, but in others like recognizing a musical instrument or a simple apparent sound source location where they have developed short cuts from experience, they get a bit looser.

Hitting The Brain With A Pan

Because our recognition systems are so sure of themselves in these areas it is actually very easy to fool them. The panning that is done with a pan pot is a good example of this. When a sound is conventionally panned from one side to the other, essentially only one of the functions that would actually take place in a real situation of a moving sound source takes place: a change in power levels or the power ratio, presented to each ear. Completely missing is all of the phase shift, frequency changing, sharping and flatting, reflective phase delta, phasing, locational change of standing wave nodes, and change of frequency dependent time delay to each ear.

Yet with all this missing your brain still buys this as a moving sound source, because the single most obvious condition has been sort of satisfied.

This tendency to 'jump to conclusions' as the information starts to come in has another interesting aspect. That is that once the decision has been made the system has a very long release time. That is, you can actually start subtracting conditions that satisfy a decision, until what remains is less than necessary to identify that sound. The brain will still continue to hear it as it did when the needed parameters were present.

To show this in another way; we have all seen the optical illusions where a little cube or some other form is seen as solid or hollow, up or down, or something. The first way that you happen to latch on to it when you see it may be 'solid' or 'hollow.'

Whatever way you happen to first percieve it is the way that you will continue to percieve it, even if someone else is there telling you the other way that it can be seen. This will continue until you provide a break in the visual input by blinking or looking away. It is hard to shake it and see it the other way; just because your system is satisfied with its earlier decision.

This can be good for those of us interested in producing synthetic acoustic experiences (records).

It often allows us to get away with serious shortcuts in construction of synthetic musical instrument voices (let's not fool ourselves; with today's technology, we are not capable of producing an acceptable copy of any musical instrument without this system's help) and it allows us to create synthetic copies of natural events that we actually don't even know how to create. We just start guessing at the components that might be present in the signal that we wish to synthesize, and keep building until our brains say, "That's enough; I give up; I'll take what you've got and do the rest myself."

— PART 2 —

The parameters discussed in part one of this article did not lend themselves well to graphs or charts; so there weren't any. The information and test results that I will put forth in this following section, however, (which I call "Part Two"), may best be conveyed in this way; so it will be.

Experiences in the studio over the years have raised certain questions in my mind that finally made me curious enough to actually do some experiments. The specific purpose of these experiments was to obtain much more specific and musically usable information regarding the apparent source of a signal eminating from two transducers; a left and a right, with delay on one of them. More specifically, I wanted to know just how much amplitude boost it would take to re-center the apparent source position of specific types of signals under these circumstances, with the other variables removed, as these other variables are not controllable when the end audio product is a tune playing on someone's automotive FM stereo system.

Setting Up

Test program material was recorded on the MCI multi-track, (see Figure 1), with each program isolated on its own track. Each return to the Midas console was bandpassed to roll-off directly below and above the energy present in the test program dedicated to that input channel, so that any background hum and noise artifact would not appear in the headphones and distract the subjects (as the artifacts would appear at drastically different right/left levels as the tests progressed).

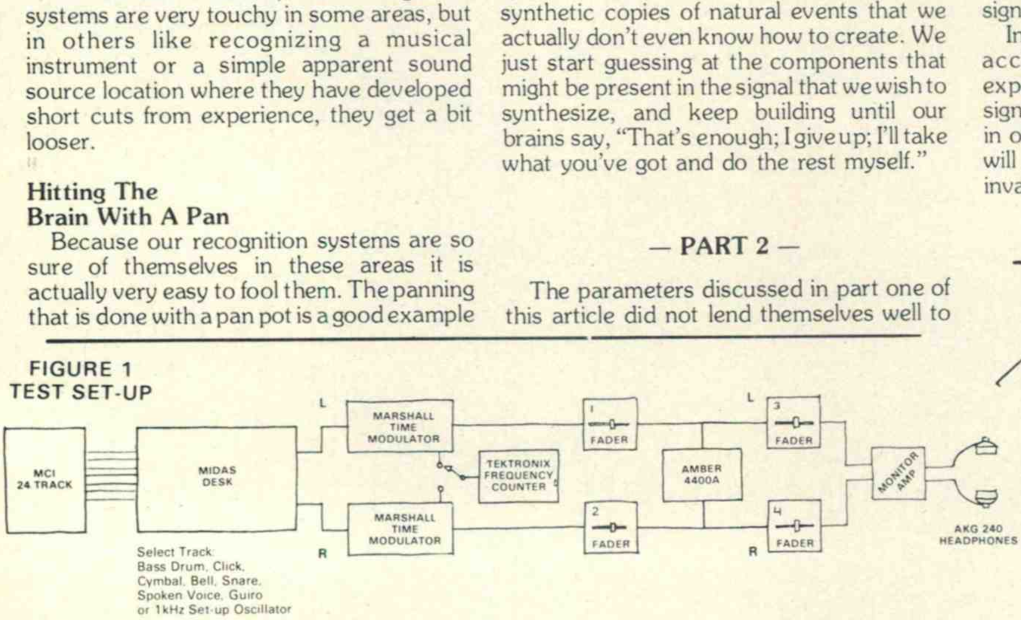
These band limited programs were then made available to two Marshall Time Modulators. The left one was permanently set for 1.00 milliseconds which we decided to make our reference delay.

The need for a reference delay exists for two reasons. First, it is not realistic to attempt to approach the extremely short delays needed for these experiments (as short as one microsecond) using a delay technique which is capable of the extremely long (100 millisecond) delay needed later in these tests; and second: to assure identical signal paths for both the left and right signals.

Interestingly (we discovered this quite by accident though it should have been expected) distortion products and other signal anomalies such as noise, etc., present in one channel but not present in the other will very clearly throw off the figures, invalidating the research.

— continued overleaf . . .

FIGURE 1
TEST SET-UP



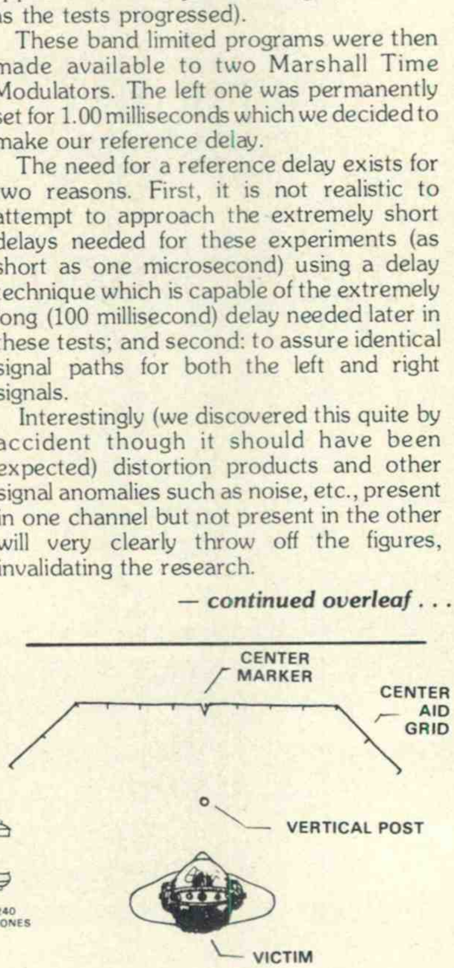
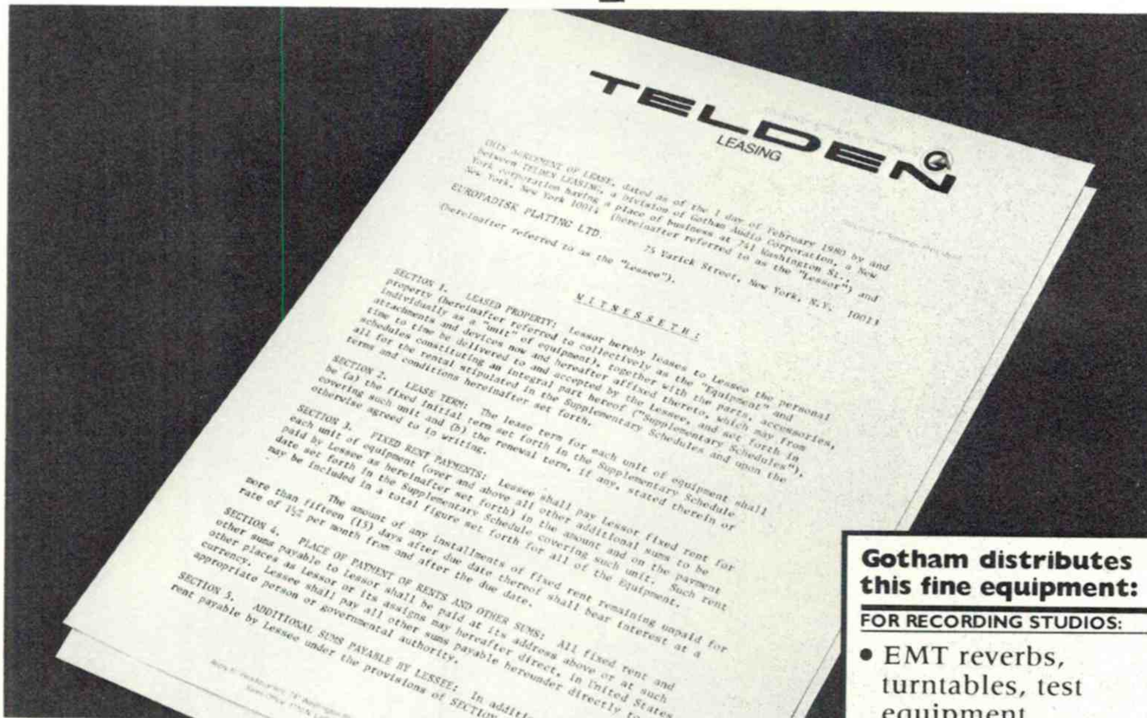


FIGURE 2
CENTER AID GRID

Every studio should be equipped with this outstanding Gotham product.



Everyone knows Gotham distributes the world's finest audio equipment.

What everyone may not know is that we also have an unusual leasing program.

For more than ten years, Telden Leasing, a division of Gotham Audio Corporation, has helped fledgling studios, disk cutting facilities and plating plants get started and enabled established operations to expand.

Unlike the usual leasing organizations with little understanding for the workings of the recording industry, we're audio professionals. And, unlike outside leasing companies only interested in bank statements, Gotham's Telden Leasing is primarily interested in *you*. Your artistry. Your skill.

We have the expertise to assist you in determining the best leasing terms as well as equipment. Even if it's not in our line. And it isn't unusual for us to start individuals in their own business just on the basis of an interview.

For more information about the numerous advantages which a Telden lease offers, write for our brochure. You'll find out why one of Gotham's finest products is made of paper.

Gotham distributes this fine equipment:

FOR RECORDING STUDIOS:

- EMT reverbs, turntables, test equipment
- K & H monitor speakers
- Neumann condenser microphones
- Telefunken master recorders
- telcom noise reduction system
- TTM universal noise reduction frames.

FOR DISK PRODUCTION:

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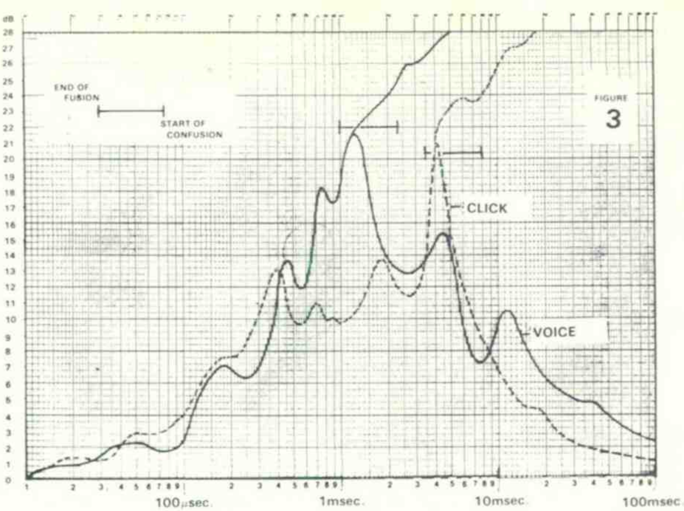
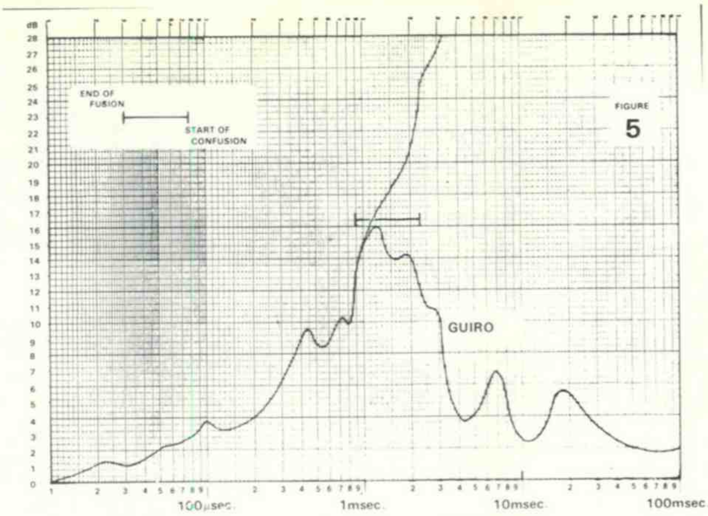
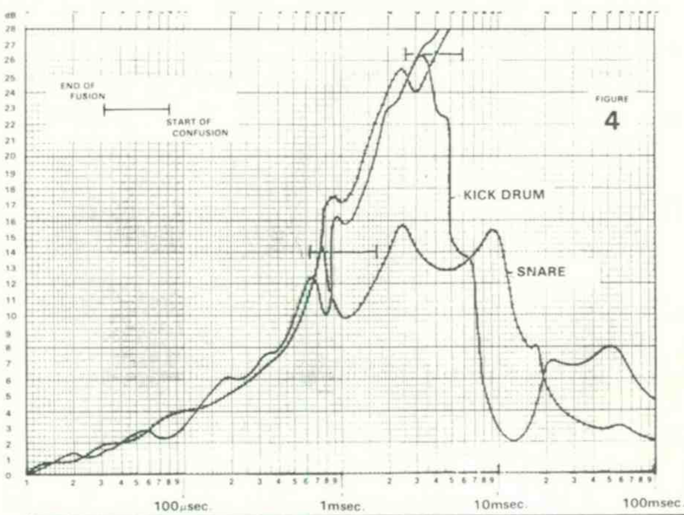
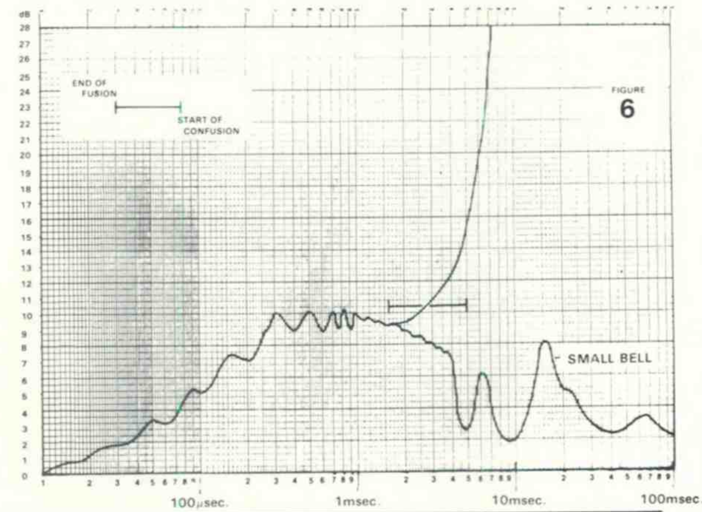
GA-3









HASSLING HAAS

We noted in fact that the total combined noise and distortion products of the two channels must match within 0.01% in order to keep the deviation spread 3 dB wide at the most narrow point. Introducing a distortion mismatch alone of 0.1% total added another 3 dB to the entire deviation spread.

The right channel Time Modulator was the swept one. This unit was used to provide the delay times relative to the left channel from one microsecond to 100 microseconds. The Time Modulator was chosen as it is the only piece of gear capable of producing the exact delay times needed, over the range needed for these experiments.

The delay increments of 3, 2, or even 1 millisecond available from digital delay lines are not sufficient (there are 24 sample increments in the first millisecond alone!).

The Time Modulator is capable of continuous (non-stepped) changes in delay, with more than an 80-to-1 ratio. This is needed in order to get the very short test times required at the one microsecond accuracy required, while also delivering the delay sup to 100 milliseconds without changing signal paths.

Signal paths (delay lines) could not be changed at any time during the experiments as this would introduce changes in any distortion and noise characteristics; possibly invalidating the test results.

The Tektronics frequency counter was tapped on to the clock of the swept unit — reading out clock frequency to 5 figures. This allows precise setting of delay times, which obviously must be very accurate as the interval between the first two sample conditions required is only one microsecond.

Faders one and two were first set with the 1 kHz test tone from the Midas to show 0 dB on the Amber with both Time Modulators set at one millisecond (no differential delay). Then the victim was allowed to trim faders 3 and 4 to generate what he felt was the most satisfying center placement. Then these two faders were covered so he couldn't touch them.

The test programs and delays were then executed, while the victim did whatever he wanted to with both faders 1 and 2 to re-center the image. I discovered that this is not really all that easy for many people, so I built a grid that was placed in front of the listener to help them visualize the movement and more easily feel the center placement (see Figure 2).

A vertical post allowed the victim to align himself properly by parallax with the center of the field. As he shifted the image around it seemed to help to have him envision the image placement against that grid.

Once he felt satisfied as to its location, the audio was killed, the test tone was selected, and the tone level adjusted until the left channel shows 0 again on the Amber. Then the right channel was read, showing the amplitude difference.

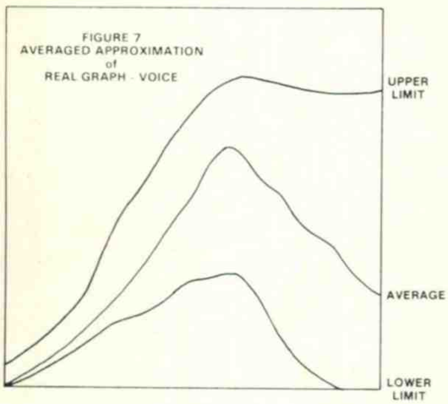
The earphones had wedges installed to angle them so that the actual drive plane relative to each ear was displaced slightly forward. This was done in an effort to use more of the outer ear structure to help reinstate some of the natural comb filtering that takes place in the outer ear when sounds come from the front quadrant. However, any location aid as a function of dynamic changes in outer ear comb filtering was, of course, impossible as the true physical sound source remained stationary relative to the ear.

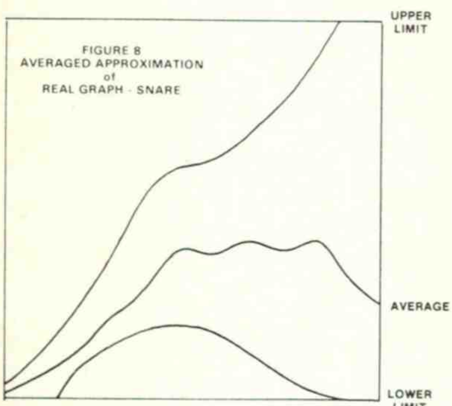
This left us with a resonably clean shot at the variables of interest for our tests; interactive location sensing as a function of full bandwidth delay time and full bandwidth amplitude differences at the two transducers. Further, we have essentially removed the variables that we did not wish to experiment with, such as outer ear dynamic comb filtering, room reverb, echo, doppler shifts, and spectral changes as a function of source location. We also removed any possibility of vertical image displacement by removing these variables.

Do not fear; what follows is not another set of graphs showing the generalized math of the precedence effect or the effects of temporal fusion, but some specific exmaples showing precendence compensation and the onset of temporal fusion collapse, followed by replacement of relative power levels as a correction variable as two discrete events become audible (see Figures 3 through 6).

As the spectral density differs from sample signal to sample signal so do the end figures and the amount of deviation from the center figures.

As the different frequencies (or harmonics) contained in each sample are percieved differently by each sample victim (due primarily to varying degrees of age or occupation related high frequency roll-off), each requires a different amount of amplitude correction to re-center them at any given common delay. Once can see that the room for subjective errors or deviations increases as the harmonic content and transient information of the sample signal increases. This means that (as the Figures 7 and 8 show) the more complex the signal the broader the spread of test response errors.





Also note that as the delay times increase, so does the spread. In other words, the test victims agreed much more closely on the amount of amplitude change that it took to re-center an image at the shorter delay times than at the longer delays, with the spread notably increasing as the point of confusion is approached.

Other factors such as whether the interpreted center was approached by lowering or raising either channel's amplitude from some arbitrary point also come into play.

It seems that as one approaches the illusion of center from either amplitude direction the mind makes a jump and forces the image to subjectively jump to a satisfactory center. Of course, the tendency then is for the test victim to stop at that point and announce that he has reached a satisfactory center image.

If one is approaching the center, let's say by raising the amplitude of the left channel, he may feel he has reached center while he is still as much as several dB low. This is a factor in the development of the lower limits shown in these graphs.

The inverse is, of course, also true; playing a large part in dictating the upper limits of the spread.

We have learned that the most accurate (by accurate we mean most repeatable) system for locating the satisfactory re-centered condition for a given subject with a given test signal is reached by a sort of successive approximation; i.e., having the subject overshoot what he feels is the center and rocking the level fader back and forth in successively smaller increments until the center of the peak-to-peak fader throw is discovered. Of course, as stated before; the more complex the signal the more amplitude spread appears in the final readings.

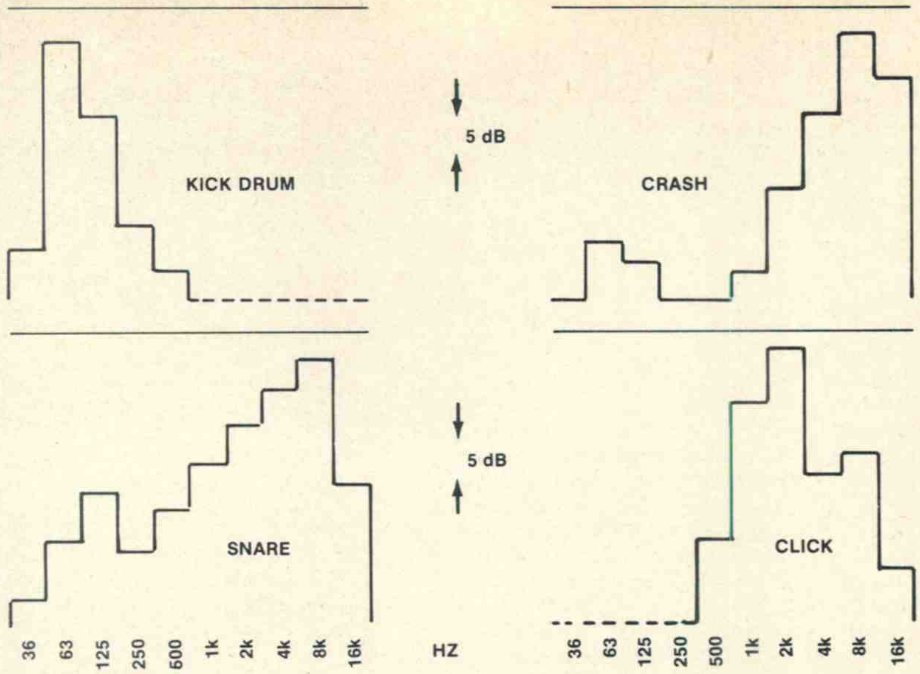


FIGURE 9

SELECTED SPECRUM ANALYSIS
OF
TEST SIGNALS

One might think that the more complex the signal that is presented to the ears and brain, the more accurate the results would be, as the brain has much more in the way of phase relationships to work with. Our findings show, however, that this is not true. In fact, we found that the narrower the power bandwidth the easier it was for the subjects to determine what they felt was a satisfying and repeatable center.

It is well known that the human system of source location depends on phase or timing differences only up to approximately 1 kHz. Above that frequency amplitude difference becomes the major determining factor.

This explains many of the surprising spreads shown in our graphs. Older subjects with high frequency roll-off actually did better in establishing repeatable center figures than did younger "full bandwidth subjects." Younger subjects whose material has been electronically rolled-off to simulate old age high frequency loss also did better.

Those subjects with a flatter high frequency hearing response were more prone to confusion when sensing the difference between transient spread and power balancing at the point of confusion.

There is a point (not surprisingly) where relocation to center becomes more difficult and finally impossible. This is the point at which amplitude cannot compensate for the precendence effect because sufficient delay exists for the brain to just begin to interpret the composite signal as two separate events while still attempting to process it as a single horizontally displaced event. This is the point which we feel is the "end of fusion — start of confusion" threshold.

This condition did not really appear where we expected it to, at the point of obvious separation of the two events, but considerably before. This point is displayed on the amplitude versus delay graphs as a horizontal line, the length of which denotes the sample spread of our victims.

After this point, one generally gives up on trying to re-center the image and begins attempting to adjust for equal apparent loudness on both sides.

If, however, the subjects are prompted to go back and try the tests again, this time attempting to continue re-centering the image; the upper portions of the graph to the right of the "end of fusion — start of confusion" appear.

Figure 9 represents a roughly averaged spectral analysis of the various program materials used in the tests for comparison purposes.

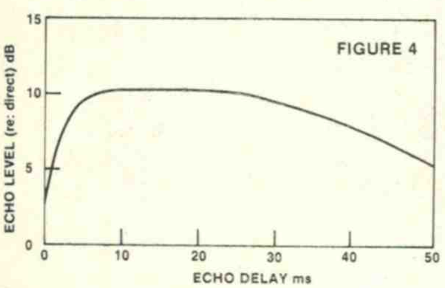
Though I am guilty of perpetuating in casual conversation the broad generality that the onset of perception of echo or discrete delay is at around 20 to 30 milliseconds, I present Figure 10 in the interest of being a little more specific. These figures were averaged from our test results. At first glance they may seem to be quite low, but you must realize that these tests were given after the very, very long ordeal of the other tests, and though the test subjects walked in with varying degrees of audio experience, they all walked out quite practiced in hearing discrete delays.

FIGURE 10

	Milliseconds Delay	
	Minimum Point of Separation	Point Where All Subjects Heard Two Clear Events (With Practice)
Bass Guitar	7	18
Cymbal	8	17
Click	7-3.5	7-14
Snare	9	19
Guiru	13	25
Voice	6-8	23

Hassling Haas

You may have noticed that our test results differ greatly from the much published and accepted general curve of Haas precedence correction amplitudes (see Figure 11). Believe me, this surprised me more than it does you. The precedence zone curve determined by Helmut Haas shows level increases up to almost the ten millisecond point. A study of our results, however, will reveal that we discovered this amplitude peaking point to be much nearer one to four milliseconds (depending on test program content); quite a difference.



The precedence zone as determined by Haas (Ref. 2). This curve describes the amount the delayed signal must be increased in level to sound as loud as the undelayed, direct sound. The delayed signal is perceived as an echo only if it exceeds this value.

At this point it is worthy of a reminder that we are working with earphones, and it is safe to assume that we have access to much more sophisticated and accurate test equipment than was available to Haas.

Still we use this curve as a reference for this discussion. Even though as it is usually seen today it is averaged, smoothed over, and really contains general results found by others at that time, it is the popular reference today. Many times this author has had people bring this very curve to me saying, "See, here it is; the answers are all right here."

The early work in this field was primarily centered on speech, which is but one example, and one curve. Other work done at approximately the same time did deal with other types of signals and did yield results suggesting that such variables as transient formatting and periodicity durations might affect these findings, but nowhere near the amount that we discovered in our specific testing.

Again referring to the Haas precedence curve; we see that from the ten to twenty-five milliseconds, there is shown basically no change, while a gradual reverse downwards slope is shown from twenty-five milliseconds on. Further, this curve shows *NO* information in the region of the first couple of milliseconds, where we clearly found the most action.

It is interesting to note that the maximum horizontal surface distance across the face between the two ears on a person who is assembled more-or-less conventionally is roughly one foot. This correlates roughly with up to one millisecond of differential delay potential. This is curiously close to what we found to be the practical peak or limit for clean location determination by phase or delay techniques, just before the onset of our "end of fusion — start of confusion" zone; and, curiously, far from the ten millisecond figure shown on the Haas curve.

We feel that the balance of this Haas curve may reflect an unfortunate averaging of pattern transient phase compensation and attempts at balancing left-right acoustic power levels (remember the zone of confusion).

Further, Figure 10 shows our test subjects as averaging a much lower threshold of discrete echo detection than is generally felt to exist.

While pointing out that the earlier experiments in this field may suffer severe inaccuracies from attempts to average and simplify figures (then and through the years since then) and a general tendency to apply these loose figures derived from voice experiments to other types of program material; it is one of the points that we wanted to make in this article.

I often hear of integration or fusion thresholds of fifty, eighty, or even one-hundred milliseconds being quoted, with delays under fifty milliseconds definitely falling well within the zone of fusion. While I assume these to be (reverberant) field figures, they still fall way beyond the limits revealed by our tests, and are generalized beyond any usefulness.

In fairness to all involved it must be remembered that no reverberant field interaction was present in our tests; as earphones were utilized. Since reverberant field tests reflect (no pun) the deviations caused by these reflections which, of course, vary greatly in real applied situations, the only really significant testing in this area must be done with earphones. This yields repeatable results. □□□

New Products

PUBLISON MULTI-FUNCTION AUDIO EFFECTS GENERATOR

Identified as the D.H.M 89 B2, the newly introduced unit, from France, can operate in Dual Digital Delay mode, Dual Pitch Shifting mode, Dual Memory Latch mode, making it an extremely versatile piece of equipment.

Dual Digital Delay: In quasi stereo mode two independent delays operate on the same sound from 1ms to the maximum value. The maximum value of delay depends upon selected bandwidth. 1200ms for 5 kHz, 600ms for 10 kHz, 300ms for 20 kHz. Optional addition of memory can extend delay to 5000ms.

In true stereo the two channels are completely separated.

The D.H.M. 89 B2 offers a unique feature in the delay mode which allows a continuous variation of delay without doppler effect nor switching noise. It is possible, according to the designers, to change delay during operation without audible effects. Dual echo is effected by use of dual delay with feedback.

Dual Pitch Shifting can be accomplished from -2 to +1 octave. Use of a sophisticated micro-computer provides phase coincidence at joining points, eliminating glitches. The serial delay is adjustable. Dual automatic arpeggio, a pitch-increasing or decreasing echo is accomplished in this mode, as is dual reversed sound - the electronic equivalent of a magnetic tape running reversed. Reversed sound can be pitch-shifted.

The Dual Memory Latch mode enables sound to be repeated indefinitely. the borders of the repeated sound as well as the reading sense and pitch ratio are continuously selectable.

THE KB2000 EXTERNAL PROGRAMMING UNIT

The KB2000 is an external programming unit for the D.H.M. 89 B2 which includes a three octave keyboard and a control panel for setting the functions as follows:

Three Voice Chorus is accomplished by adding two pitch shifted voices to the original sound. The resulting chord is controlled by keystroke. A serial delay can be added, which is continuously variable.

Reverse Synchronization: The reverse mode from the DHM alone affects the original tempo in a random manner. The reverse synchronization mode triggered from the KB2000 synchronizes this effect with the attacks of the sound, so that the reversed sound has the same tempo as the original. The added delay is adjustable.

Biphonic Memory Synthesizer: Any existing sound, supplied by tape or by microphone, is memorized by depressing the "Memory Latch" button. The KB2000 synchronizes memory reading with attacks of the notes. "Attack Point" sets the point at which each note begins. "End Point" and "Return Point" select the part of the memorized sound which is repeating continuously when the note is sustaining. By means of an internal phase tracking computer the sustain is said to be very clean. The "Speed" control sets the reading speed of the memorized sound.

Two Envelope Generators drive VCAs to control the envelope of the notes. The

attack, release and hold times are adjustable.

Dual Evolving Vibrato is controlled through three basic parameters. Frequency sharpness and depth are modulated function to time by a form generator initialized by attacks of the notes. Glissando time from a note to the following note is adjustable. Sustain Mode is controlled by keystroke and continues even when the key is released. The real duration of the note depends on the settings of the envelope generator. Push-Play Mode enables the notes to be heard only when the keys are continuously depressed.

Available from selected professional audio dealers.

PUBLISON AUDIO PROFESSIONAL
5-7-9-11 rue Crespin-du-Gast
Paris, 75011, France

for additional information circle no. 68

UREI INTRODUCES NEW ELECTRONIC CROSSOVER

Identified as their Model 525, the unit features four, panel selectable operating modes: stereo 2-way or 3-way, and mono 4-way or 5-way. Crossover frequencies are continuously adjustable from 50 Hz to 10 kHz, with the actual frequlency measured and displayed on a digital frequency counter, with 1 Hz resolution.

Mode select and crossover frequency controls are front panel screwdriver slot adjustments for greater security. A subsonic filter is included to roll-off frequencies below 30 Hz providing protection of the low frequency transducers in the PA system. This is switch selectable on the rear panel. Inputs and outputs are XLR/QG connectors or terminal strips. Security covers are also avilable for the Model 525.

U.R.E.I
8460 San Fernando Rd.
Sun Valley, CA 91352
(213) 767-1000

for additional information circle no. 69

SHURE SM63-CN SMALLER SIZE, HIGHER OUTPUT MICROPHONE

Being less than six inches long and weighing only 2.8 ounces, the newly introduced microphone is said to be particularly well suited for on-camera, on-stage use.

The SM63-CN is a dynamic, omnidirectional type with an output that is about 6 dB higher than larger, comparable units. Other performance advantages include a controlled low-frequency rolloff to insure natural sounding music and voice pickup, as well as smooth high frequency response for an overall clear, crisp clean sound similar to

some condenser microphones.

Additionally, the SM63-CN features a hum-bucking coil that makes it insensitive to strong hum fields, such as those produced by lighting; a mechanical-elastomer isolation system that makes it resistant to handling noise; built-in breath and pop filter; rugged VERAFLEX® polyester grille that is impervious to dents, rust, and moisture; supplied external windscreen accessory for outdood use.

Price: $100.00, which includes a swivel adapter windscreen and professional three-pin audio connectors on both ends of the cable.

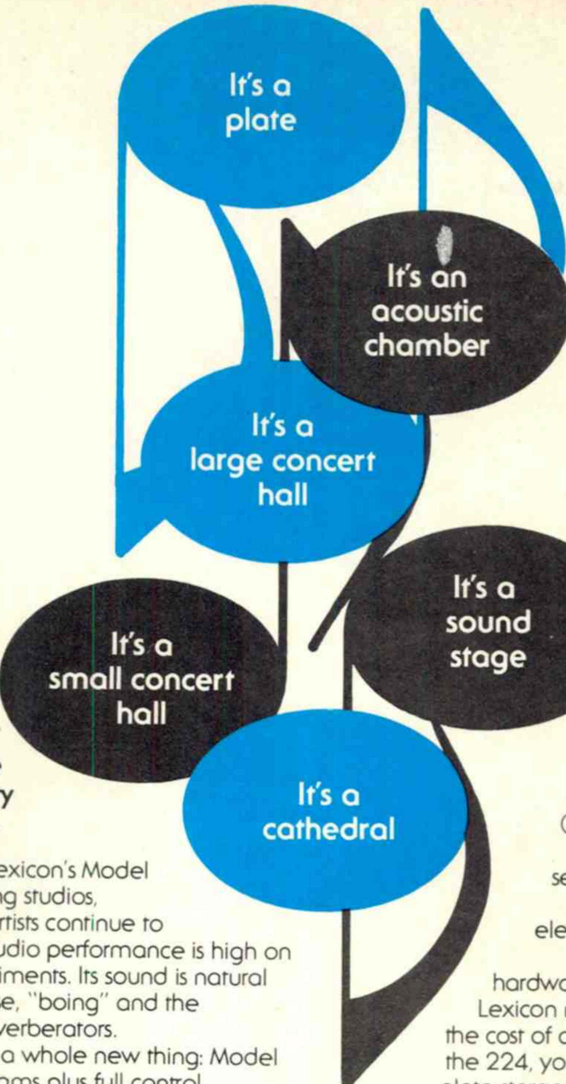
SHURE BROTHERS INC.
222 Hartrey Avenue
Evanston, IL 60204

for additional information circle no. 71

CROWN INTRODUCES AUDIO COMPUTER FOR ANALYSIS OF SOUND

As announced by Max Scholfield, president of Crown the Badap 1™ audio computer will be manufactured and distributed by Crown. The system was developed, in association with members of

for additional information circle no. 70



... A complete repertoire of the very highest quality reverb at your fingertips.

Rave Notices for Lexicon's Model 224 from the world's leading studios, broadcasters and musical artists continue to pour in. The 224's superb audio performance is high on the list of everyone's compliments. Its sound is natural and completely free of noise, "boing" and the problems of mechanical reverberators.

And its versatility is a whole new thing: Model 224's multiple reverb programs plus full control of all reverberation parameters allow audio engineers to create the sound that "is exactly right for the material." It's a whole stable of quality reverb capabilities in one compact package.

The model 224 Digital Reverb is available with 2, 4 or 6 basic reverb programs (easily updated in the field by simple program plug-in) and comes complete with extensive self-test diagnostic programs that take the guess work out of maintenance. Its mainframe, all electronic, requires only 7 inches of rack space.

If all this sounds expensive, it isn't. Digital hardware and software technology breakthroughs by Lexicon make the 224 surprisingly affordable ... about the cost of one good plate. And after you've experienced the 224, you just might find more productive uses for your plate storage rooms and acoustic chamber!

For full details write or call — better yet contact your leading Pro Audio dealer for a demonstration. Once you've heard it you'll never want to do another session without it.

WORLD CLASS REVERB

Lexicon, Incorporated 60 Turner Street Waltham, Massachusetts 02154/(617) 891-6790/TELEX 923468
Export: Gotham Export Corporation, New York, New York

lexicon

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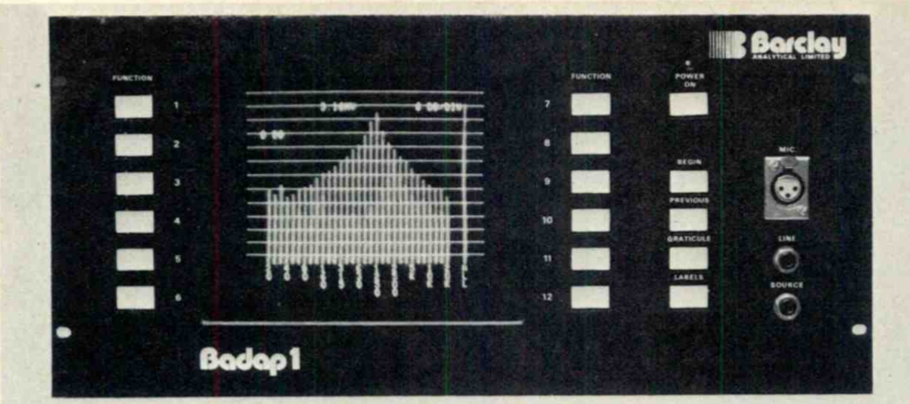

the Crown engineering staff, by Dr. Clay Barclay of Barclay Analytical Ltd., Wynnewood, PA. Early production of the system was at the Barclay facility. Increased sales and demand for the equipment dictated a move of production to the Crown plant. Dr. Barclay will continue to assist in development of new software for the system.

Badap 1 was designed to replace a series of dedicated test instruments, relieving audio engineers of a burden of investment in test equipment. The current mainframe of the Badap 1 system includes software for RTA and RT60 analysis; new software is now being developed for additional functions, such as acoustics distance measurement and stereo analysis. Early announcement of the software for a 32-input multi-plexer is expected, providing real-time analysis capabilities for recording studios.

Badap 1 uses a single-gun color CRT for data display with all data being captured digitally and stored in memory. This enables the operator to expand or contract the display to include only that segment of the data which is of immediate interest. This, it is claimed, is a significant improvement over current memory systems which store only the data in the format in which it is being displayed, making it impossible to expand stored data.

Up to eight data sequences can be displayed at one time through the use of a system of bars and dots and four NTSC compatible colors. Data can be stored and recalled later for comparison with real-time data. The NTSC-compatible display system makes it possible to videotape the CRT display or to display it on a larger remote monitor for instructional purposes, or to record it on video cassette for later analysis.

Software for the Badap 1 system is on circuit boards. As new software becomes available, Badap owners can simply add boards instead of buying new test instruments.

CROWN INTERNATIONAL
1717 Mishawaka Road
Elkhart, IN 46514
(219) 294-5571

for additional information circle no. 73

TONGARINGS™ PROVIDE IMPROVED TONETONE FOR STUDIO DRUMS

TongaRings™, a new patented acoustic drum accessory developed expressly to provide pitch modulation from tom-toms in addition to increased punch and definition from bass drums, are now available.

According to Tonga president, Calvin Rose, "drummers can now get more tone and less bang from their drums quickly and inexpensively without second heads, complicated tuning proceedures, and home-made foam rubber strips." In addition, TongaRings allow studio drummers and recording engineers to achieve natural, unsynthesized pitch modulation instantly for easy, single head tuning, more bass drum punch, and a downward sliding pitch effect.

Easy to apply, pressure sensitive, computer designed TongaRings™ are produced from mirror-finish Mylar in a range of diameters from 6" to 24" to fit the majority of popular drums. The company reports that the durable, stretch resistant disks were research tested at Northern Illinois University's Acoustics Laboratory.

Retail prices range from \$2.50 for 6" TongaRings to \$12.00 for 24" models. Available directly from the manufacturer or from selected music dealers.

TONGARINGS
P.O. Box 3803
Stamford, CT 06905
(203) 329-1838

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SYMETRIX ANNOUNCES NEW SIGNAL GATE

Designated the SG-200 Dual Signal Gate the new unit is claimed to exhibit

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Our new computer system tracks over 20,000 in-stock parts daily to give you fast, accurate feedback on your inquiries. Additionally, Scully dealers now maintain a well-balanced parts inventory. This streamlined system coupled with a thoroughly trained parts and service staff are your assurance of prompt service...whether it be one year or 20 years after the sale!

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Because our machines are manufactured with your future in mind, we build in the flexibility for you to add parts as new options are developed. Whether you bought your Scully in 1960 or up until 1979, we will show you how to update your machine in your studio with state-of-the-art technology. Or our technicians can make the necessary alterations in our factory...from a major overhaul to a minor adjustment, rely on Scully. We want to keep you up to date on the latest technology and you don't have to go out and buy a new machine to do it.

Update Bulletins

Timely bulletins listing the newest options are available to customers free of charge for the following series: 100, 270, 280B, 284, 288 and 500 loggers. For your copy of any of these bulletins simply write to us on your letterhead. These are some of the ways we are continuing in the great tradition of Scully.

AMPRO Scully

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for additional information circle no. 76

considerable advances in both performance and ease of operation over existing designs. The SG-200 contains two totally independent gates sharing a common power supply in one, 1-¾" high, rack mount chassis. Operating controls for each gate include attack time, release time, range of attenuation, threshold, a totally silent in/out switch, and an internal/external gating command control.

Symetrix claims problems inherent in many previous "noise gate" designs have been solved in several ways. First, gain control in the SG-200 is accomplished by true voltage controlled amplifiers (VCAs) rather than FETS or optical devices, thus eliminating the distortion problems inherent in FETs, and the slow response times of the optical attenuators. Secondly, traditional noise gate problems such as "turn on thump", "chatter", "falsing", and low frequency distortion have been completely, it is claimed, eliminated by the device's unique *program controlled attack logic* circuitry.

Suggested list price: $399.

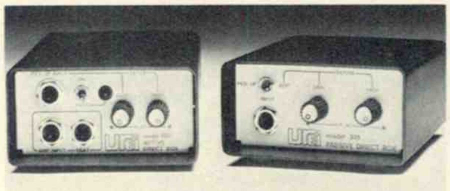
SYMETRIX, INC.
109 BELL STREET
SEATTLE, WA 98121
(206) 682-3076

for additional information circle no. 77

NEW DIRECT BOXES FROM UREI

Added to UREI's broad line of products are two new direct boxes; Model 315 is a passive design, while the other Model 325 is an active component. Both have been proven by in-studio use at United Western Studios for over a year.

The Model 315 passive box uses a proprietary transformer, designed by the newly formed transformer division of UREI. This transformer allows the Model 315 to accept more level and exhibits less roll-off of the low frequencies than other competitively priced direct boxes. According to the manufacturer, when the Model 315 is plugged into the output of an amplifier it provides the user with extremely smooth low and high EQ.



The UREI Model 325 active direct box is transformerless from the instrument output to the mixing console input. To assure there are no ground loops, the amplifier output and input are separately transformer isolated. The 325 can be powered by the 48 volt phantom power supply in the mixing console, or by a pair of standard 9 volt batteries. The company claims that, "to equal the performance of the 325 with a microphone, the user would have to spend over a thousand dollars and it still wouldn't have the dynamics or transient response the 325 is capable of delivering."

Both the Model 315 and the 325 direct boxes are assembled in rugged steel cases, with all switches and connectors recessed and finished with hard epoxy paint, capable of standing the rigors of touring concert service, or constant studio movement.

U.R.E.I.
8460 San Fernando Rd.
Sun Valley, CA 91352
(213) 767-1000

for additional information circle no. 79

GARNER BULK TAPE DEGAUSSER

The high-speed Model 1100 Bulk Tape Degausser has been designed especially for high density audio tapes, video cassettes and 1" high-coercivity video tapes. It will erase up to thirteen reels per minute and

can handle reels up to 10½" in diameter.

Constructed of heavy-gauge aluminum, it is equipped with an internal fan for heat dissipation. An automatic thermal cutoff switch is also provided to prevent overheating.

GARNER INDUSTRIES
4200 No. 48 Street
Lincoln, NB 68504
(402) 464-5911

for additional information circle no. 80

AUDIOMARKETING DEVELOPS TIME/SYNC™ ELECTRONIC CROSSOVER

Developed as a new electronic crossover that corrects the inherent disadvantages of point source configuration speaker systems, the new unit is called the

for additional information circle no. 78

Would You Use An Elephant Gun To Shoot a Duck?

If you answered yes, you're probably using a 50 watt stereo amp to power your headphone system. If you answered no, what are you using? The point is different products are designed to accomplish different tasks. These headphone amps are not designed to power monitor speakers, they're designed to power headphones in either a stereo or mono mode. The AP-10 will power four headphones minimally, and the HA-100 will power at least eight with individual volume controls. The AP-10 can be rack mounted or outboarded, and the HA-100 naturally mounts in 1-3/4 inch rack space.

We are not suggesting you ditch your elephant amp powering your headphone system, but as long as you're hunting ducks, drop by one of the stores listed below. Not only do these fine stores handle large power amps for big kill, they are also sensible enough to carry headphone amps for the more thrifty among you.

CALIFORNIA

Alco Paramount
San Jose

Castle Music
Goleta

Catania Sound
San Rafael

Central Calif. Electronics
Fresno

Sound Foyer
Los Angeles

Union Grove Music
Santa Cruz

Wah Sound
Sacramento

CONNECTICUT

Audiotechniques, Inc
Stamford

Vinco Electronics
Hamden

FLORIDA

Danforth Sound
Orlando

Grice Electronics
Pensacola

Harris Audio Systems
N. Miami Beach

Nelron Communications
Ft. Lauderdale

Sound & Signal Systems
Tampa

GEORGIA

B & H Sales
Tucker

ILLINOIS

Bridgewater Custom Sound
Harvey

Palm Music
Effingham

INDIANA

Bowman Electronics
Valparaiso

MARYLAND

Stereo Equipment Sales
Baltimore

MASSACHUSETTS

Cramer Video Systems
Needham Heights

Springfield Audio
Springfield

MICHIGAN

Hy James Enterprises
Ann Arbor

NEW JERSEY

Stuarts Audio
Westfield

NEW YORK

Advance Sound
Dix Hills

Fisher Electronics
Syracuse

Harvey Sound
New York

OHIO

Gattusco's Music Center
Canton

Music Man Sound
Columbus

Reddish Stereo, Phil
Parma Heights

OKLAHOMA

Ford Audio & Accoustics
Oklahoma City

PENNSYLVANIA

Pro Sound, Inc
Wynnewood

TV Parts
New Brighton

TENNESSEE

Strings & Things
Memphis

TEXAS

Fingers
Houston

Marvin Electronics
Ft. Worth

McAllen Radio
McAllen

Milltronics
Dallas

Southwest Sound & Elect.
Austin
San Antonio

WEST VIRGINIA

Black & Sons, J.M.
Wheeling

Electronic Specialty
Dunbar

Mack & Daves
Huntington

CANADA

Audesco Electronics Ltd.
Montreal

Electro Tec Marketers
Vancouver

Thorvin Electronics
Toronto

PUERTO RICO

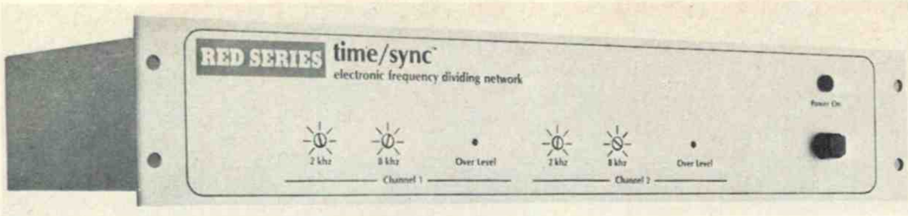
Williams Tech Sales Co
Rio Piedras

Headphones Courtesy AKG

Edcor

16782 Hale Avenue, Irvine, California 92714 (714) 556-2740

for additional information circle no. 81



Time/Sync™ Frequency Dividing Network (FDN).

Time/Sync is designed to be fully compatible with Audiomarketing's "Red Series" studio speaker systems, as well as any 604-type loudspeaker. The design philosophy follows: The strength of the Big Red 604E monitor system is said to be its co-axial point source configuration. The woofer cone is positioned ahead of the tweeter diaphragm, thus signals produced by the woofer and tweeter leave the drivers simultaneously, but actually arrive at the listener's ear at slightly different times. To all but the most sophisticated listeners the difference is imperceptible. However, this difference is not imperceptible to the recording engineer. Thus, Time/Sync™ incorporates time delay in the low pass section enabling the tweeter signal to catch up. The result is said to be a completely coherent sound.

"The key to the system is the 'time delay'," said Rick Anderson, AudioMarketing's vice president and general manager. "Time/-Sync is an active crossover which electronically synchronizes the speaker signals. It's far more effective and efficient than a passive crossover. It has a 95dB dynamic range and a crossover frequency of 3000 cycles."

Anderson points out that an active electronic system gives the recording engineer more control than the traditional passive system.

The new unit incorporates LED over-level indicators. Input impedance is 25k Ohms balanced; hi frequency control is dual high pass shelving 2kHz and 8kHz; power required is 110/220 vac at 20 Va. The unit is housed in a standard 19" x 3½" rack mount.

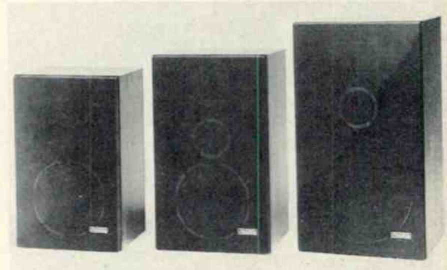
Cost of the dual channel Time/Sync™ is $1775. This includes two Big Red speakers and a Time/Sync electronic crossover. Time/Sync alone will sell for $495.

AUDIOMARKETING ,LTD.
652 Glenbrook Road
Stamford, CT 06906
(203) 359-2315

for additional information circle no. 82

NEW MID-SIZE PRO LINE
FROM ALTEC

The three models in the new series incorporate sound reproduction technology which was previously only available in larger sized cabinets. The series consists of Model 4, a 10 inch, two way system; Model 6, a 10 inch 3 way system; and Model 8, a 12 inch three way system. All three models feature Altec's exclusive Mantaray horn and Tangerine radial phase plug, plus LZT (lead zirconate titanate) ultra-high-frequency compression drivers. LZT is a space-age semiconductor material that directly converts electrical energy to physical motion, replacing the conventional magnet and voice coil.

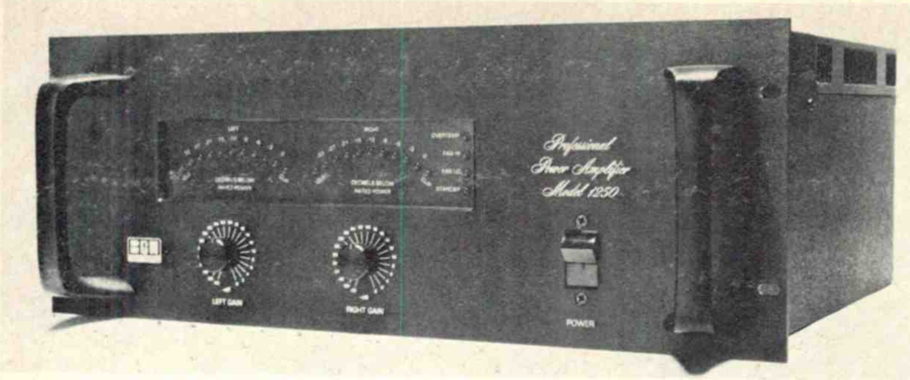


Other features new to this size range are Altec's exclusive Automatic Power Control, which prevents power overload, and anechoic damping, an acoustically absorbent baffle covering that reduces difraction distortion.

ALTEC CORPORATION
1515 So. Manchester
Anaheim, CA 92803
(714) 774-2900

for additional information circle no. 84

for additional information circle no. 83



THEIR LARGEST AMP EVER, BGW INTRODUCES MODEL 1250

The new Model 1250 offers 400 watts RMS into 8 ohms with no more than 0.03% distortion. Also at 8 ohms, the 1250 is said to deliver 1200 watts mono at no more than 0.05% distortion.

The 1250 continues BGW's philosophy of using only full comlementary circuitry. The newly introduced unit uses a total of 48,200-watt power transistors and is cooled by a three-speed solid-state controlled fan.

Other features of the Model 1250 are: Mains voltage changed from rear panel with no additional parts. New multicolor LED peak reading power meters with true clipping, temperature and mode indicators. Standby mode during turn-on, and when DC offset or over temperature conditions occur. Soft turn-on for low mains inrush current. High input sensitivity, full output for 0 dBm input. True step-attenuator gain controls calibrated directly in decibels.

And, XLR-type and ¼" phone input connectors. Separate circuit and chassis grounds at rear of panel barrier strips. Speaker protection using arc-interrupting techniques. Modular construction. All steel package for strength and RFI shielding.

BGW SYSTEMS
13130 So. Yukon
Hawthorne, CA 90250

for additional information circle no. 85

AMPEX INTRODUCES MULTI-CHANNEL AUTO BIASING FOR ATR-124 RECORDERS

"This plug-in accessory further demonstrates the forethought that has gone into the ATR-124, which allows an auto bias feature to be incorporated," said Ed Engberg, audio products manager.

Auto biasing quickly sets the right bias for any tape's characteristics. According to Ampex, the ATR-124 equipped with this new accessory increases the bias level, stops and stores the correct amount of overbias as selected. Once bias is set for a single channel it automatically becomes a ganged setting for other selected channels. Typical processing time required to accomplish the automatic biasing is 10 seconds.

Overbias is selectable from 1 to 6¾ dB in ¼ dB steps. Storage of the selected level is permanent, even in the event of a power failure or accidental shut off. Accuracy of settings is better than 0.1 dB.

The auto-bias accessory generates test signals internally and automatically selects the right one for the three tape speeds.

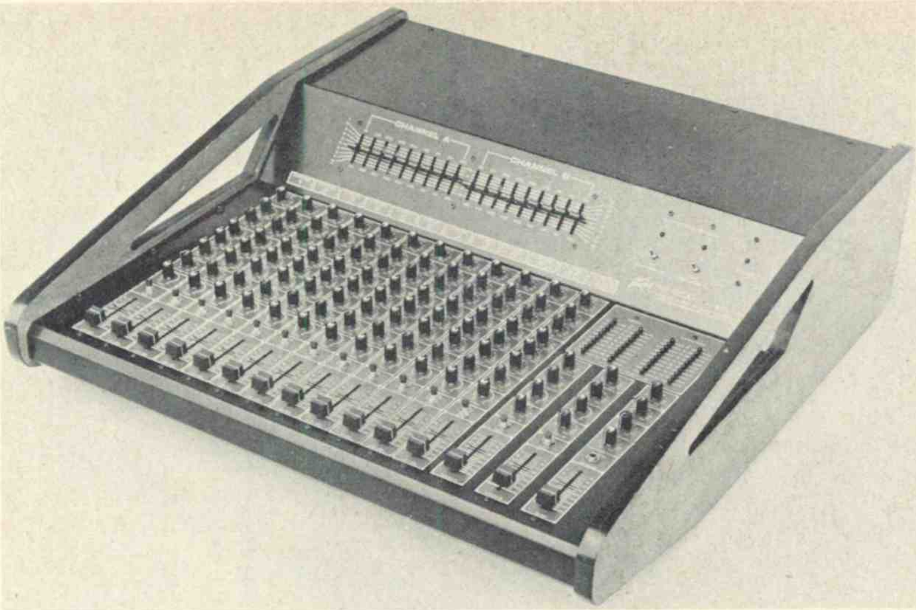
Manual bias levels, as set by the engineer on trim pots, are always accessible by pressing the manual switch.

AMPEX CORPORATION
401 Broadway
Redwood City, CA 94063

for additional information circle no. 86

PEAVEY POWERED MIXING CONSOLES INTRODUCED

Said to be the most sophisticated powered mixing consoles available today, these new mixers feature balanced low impedance as well as unbalanced high impedance inputs and individual pre in/out jacks on each channel. There are two independent pre monitor sends, a post effects send, a PFL/cue button, and active



3-band EQ on each channel.

The master sections feature a unique headphone monitoring system, ten-position LED ladder displays for mains and monitors, dual 9-band graphic equalizers, and a unique control that allows mixing the reverb and effects back into the monitor buss.

The XR-800 (8 channel mixer) is powered by Peavey's new 240SC 100 watt per channel stereo power amp and features DDT compression circuitry. This feature electronically senses the onset of power amp clipping and automatically compresses the signal only at those times when distortion would be produced by the power amp..

The XR-1200 (12 channel mixer) is powered by the field proven CS-400 circuitry featuring 200 watts RMS per channel and DDT compression.

Both models utilize modern fiberglass/-polyester woodgrain panels to give a combination of good looks and ruggedized construction to withstand the most rigorous continuous professional use.

PEAVEY ELECTRONICS CORP.
711 A Street
Meridian, MS 39301

for additional information circle no. 88

LEXICON ANNOUNCES NEW MODEL 122 SERIES DIGITAL DELAYS

The newly announced Model 122 replaces the Model 102 series and features 14 bit floating point digital encoding with 6 dB gain steps for wide dynamic range free form noise pumping and artifacts. Nine pole Butterworth anti-aliasing filters are used on all inputs and outputs for flat frequency response. PCM digital encoding, it is stated, ensures uniform superior performance on all audio frequencies within the bandwidth, free from slew rate and high frequency/amplitude distortion inherent in delta modulation units.

The Model 122 series is available in mono or stereo. Model 122 (mono) has 1-5 outputs, each individually adjustable. Delay memory capacity is modular with up to 320ms of delay available in incrementsof 40ms in each Model 122 frame. Model 122 is intended for large sound reinforcement applications or professional audio applications requiring many delay outputs or longer delays.

Model 122 (stereo) is provided in a stereo frame and has two independent delay lines each with one or two outputs. It contains a

VCO Module as standard for time modulation and other special time based studio effects. Delay memory is 160ms per channel in 40ms modules. Outputs are adjusted in 2.5ms steps. Model 122-S is intended primarily for studio applications.

Both models are fully modular. Field reliability and serviceability are significantly enhanced by module redundancy and Lexicon's advanced repair/exchange module service. Units may be field expanded to maximum configurations by simple module "plug-ins".

Lexicon has a special program, available through Lexicon dealers to allow present Model 102 Series owners to upgrade to Model 122 performance.

LEXICON, INC.
60 Turner Street
Waltham, MA 02154
(617) 891-6790

for additional information circle no. 90

SEQUENTIAL CIRCUITS ANNOUNCE REVISED PROPHET-5 SYNTHESIZER

As announced by the manufacturer, major technological advances have made possible a number of significant improvements and additions to the Prophet-5. These include:

A built-in CASSETTE INTERFACE enabling the transfer and storage of complete sets of 40 programs on to regular cassette tape. The programs can then be re-loaded or transferred to any other Prophet. It is possible to store an unlimited number of programs for use at later dates. Virtually any cassette recorder may be used.

A totally unique feature of the Prophet is the ability to use different tuning scales. A special "variable scale" allows tuning each of the 12 notes in an octave to different frequencies. The range for each is about±½ semitone form its normal equal tempered value. These tunings can then be programmed into memory, which enables instantaneous switching from one scale or key to the next. Since they reside in memory, they can be stored on cassettes with the other programs for future use. For the fist time, in a standard commercially available instrument, the musician can use other scales; such as just intonation, mean tone, Pythagorean, etc., as well as their variations (and different keys).

The EDIT feature has been improved to provide the utmost simplicity in program modification. Turning a knob or hitting the desired switch will immediately modify any program in memory. No special 'edit' mode switches need to be touched.

A VOICE DEFEAT system enables disabling of a defective voice in an emergency situation, even while playing.

Other advances allow for fewer electronic parts in the Prophet-5. A radically new

oscillator design, coupled with a unique new computer-correction scheme, is said to, practically eliminate tuning problems.

Sequnetial Circuits has established a service center network in eight major cities, nationwide.

The new Prophet-5 lists for $4595.

SEQUENTIAL CIRCUITS, INC.
3051 North First Street
San Jose, CA 95134
(408) 946-5240

for additional information circle no. 92

STEREO GRAPHIC EQUALIZER FROM PEAVEY

The newly announced graphic features two independent ten-band sections with 15 dB cut or boost at ten center frequencies. Low and high cut filters are provided for each channel with 12 dB per octave roll-off capability.

The input circuitry can be matched to a wide range of signal levels thanks to special gain/attenuator level controls. Balanced and unbalanced outputs are equipped with protection for any accidental overvoltage or short circuit situation.

Because of high level transformer balanced output circuitry the Stereo Graphic has the capability fo providing greater than +16 dBm into 600 ohms making it an excellent high quality line amplifier.

PEAVEY ELECTRONICS CORP.
711 A Street
Meridian, MS 39301

for additional information circle no. 93

OBERHEIM INTRODUCES OB-SX POLYPHONIC SYNTHESIZER AT $2995

TheOB-SX(6), and the OB-SX(4), use the identical proven circuitry hence have the same 'fat sound' except that the (6) has six voices and the (4) has four voices. [The (4) can be upgraded to six voices by adding two additional voice cards.] The OB-SX series differs from the OB-X series in that it is much smaller and easier to use since user programmability is not necessary. The OB-SX is a true polyphonic synthesizer for the keyboard player, whereas the OB-X is primarily used by the synthesist.

Features include: Four or six voice option; 24/48 program option. Four octave keyboard. Operates on line voltages from 90-130 volts. Pitch blend and modulation levers. Auto tune, hold/chord feature. Edit mode, unison, portamento, LFO rate, osc. 2 detune, filter frequency, attack, decay and release. Transpose.

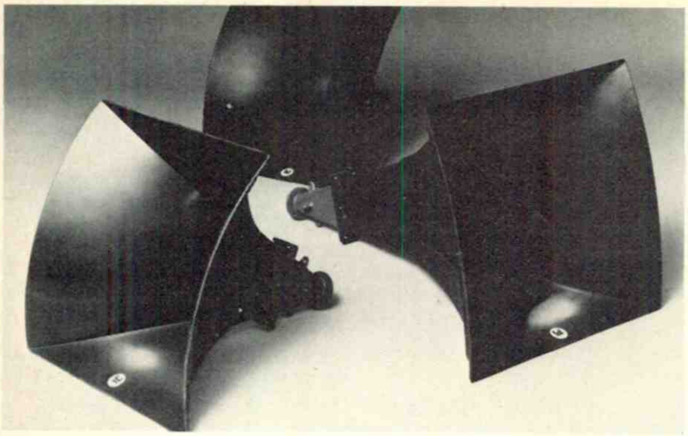
Rear panel interfaces: Filter Pedal, sustain foot switch, modulation pedal, CV in/out, gate in/out, group a/b program switch and Oberheim computer interface.

The OB-SX(4) lists for $2995. The OB-SX(6) lists for $3495.

OBERHEIM ELECTRONICS, INC.
1455 19 Street
Santa Monica, CA 90404
(213) 829-6831

for additional information circle no. 95

for additional information circle no. 94



JBL ANNOUNCES NEW BI-RADIAL CONSTANT COVERAGE HORNS

The new units are the Models 2360, 2365, 3266 and are said to be ideal for fixed installation applications. The horns provide uniform on and off axis response from below 500 Hz to beyond 16 kHz. In addition, their unique geometry and tall mouth dimensions ensure precise vertical as well as horizontal beamwidth throughout the rated frequency band.

Designed to simplify cluster lay-out and eliminate the need for horn overlapping, the three models are available with nominal coverage angles of 90°x40°, 60°x40°, and 40°x20° (-6dB beamwidth, horizontal and vertical). All three horns feature 31-5/16″ square mouth dispersions, and each is supplied with a cast aluminum throat that will accept JBL 2440, 2441 or 2482 two-inch diameter compression drivers. The horn bells are constructed of heavy duty 5/16″ fiberglass reinforced plastic. Mounting holes are provided on the bell's top walls to facilitate three point hanging.

JAMES B. LANSING SOUND, INC.
8500 BALBOA BLVD.
NORTHRIDGE, CA 91329

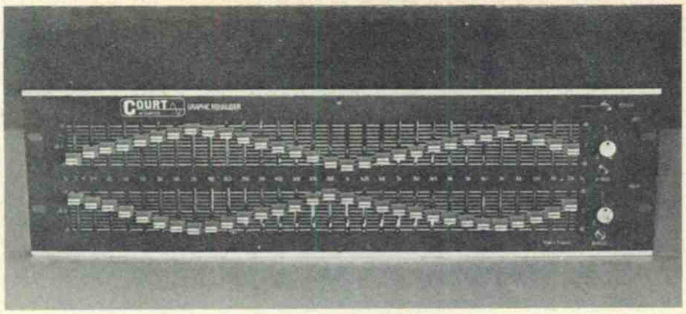
for additional information circle no. 96

NEW INTERFACE SERIES 114 MIXERS

The new compact portable high performance professional 12 in 4 out (12 into 4 into 1) mixer is designated Series 114 and uses modules with bent over panels to accommodate the input and output connectors directly on the module, thus eliminating all wiring to the mother board into which the modules plug.

The standard output module shown in the photo uses flush peak-reading LED VU level indicators just above the slider masters; these new LED VUs made by IE are bright and viewable from any angle, and they make possible a very compact mixer. The module also includes two monitor mixdowns with solo, plus cue masters and a setup oscillator.

The standard Type J input module includes trackselect, phantom power, phase reverse, line/mike switch, solo, four cue/echo sends, odd/even panpot, three equalizers with four-frequency switch at the mid, four-position low roll-off, and Duncan professional conductive plastic slider with dust seal. Input uses transformer for best



interference rejection without adjustments and includes 4-position pad before the transformer and 4-position input gain set. Module also includes an LED preamp overload danger indicator.

According to Interface, performance of the mixer is to the highest professional standards, and the latest high performance ICs are used. Other premium type input modules (with parametric equalizers and/or VCA controls) are also available in this series. Pans are available for 8, 12, or 16 inputs. The 12 input mixer, Model 12x4-114J/LEDVU, measures 16x25x4 inches, weighs 26 pounds, and is priced (1980) at $4,920.00 retail list, quantity one.

INTERFACE ELECTRONICS
3810 Westheimer
Houston, TX 77027
(713) 626-1190

for additional information circle no. 98

please mention . . .
YOU SAW IT IN R-E/P

COURT ACOUSTICS APPOINTS QUINTEK U.S. DISTRIBUTOR FOR NEW GE 60 DUAL 30 BAND GRAPHIC EQ

The GE 60 has been designed for studio and professional sound system users who want the ultimate in tone control for stereo equalization.

Two 30, 1/3rd octave bands, based on standard ISO center frequencies from 25 to 20,000 Hz are provided in a compact, efficient, cost effective package.

The unique Uniloop filter and feedback design affords optimum shaping for curve linearity and phase stability. An important feature of the design is that in the maximum or minimum positions they still exhibit a substantially flat response allowing the GE 60 to be used for 'all boost' or 'cut' applications.

The filters consist of precision inductors and capacitors to minimize noise and distortion found on active equalizers.

All 60 precision faders have center 'clock stops' for accurate alignment and afford 20 dB of Lift or Cut. The level control provides up to 20 dB of gain and the 'By Pass' switch allows A-B comparisons and enables the signal to pass through with the equipment switched off.

The 25 Hz filter acts as a shelving network so it can be used as a rumble filter, a subsonic filter to eliminate room agitation or unwanted low frequency elements from high power speaker systems. Similarly the 20 kHz fader acts as a low pass filter to minimize supersonic interference.

Frequency range is from 18 Hz to 24 kHz. Filters have a gain range of ±10 dB. Maximum output level is +23 dBm. Input impedance is 10 K ohms. Output impedance is 100 ohms, suitable for 600 ohm or 10 K bridging. Noise is better than 85 dBm. THD is better than 0.01%.

The unit is suitable for rack mounting or for use as a free standing unit. It is 5¼" high by 10" deep. A tamperproof cover is available as an option, as is a pink noise test

for additional information circle no. 99

record, 1/3 octave bands 20 - 20,000 Hz. Suggested price is $1,690.00.

QUINTEK DISTRIBUTION, INC.
Suite 209
4721 Laurel Canyon Boulevard
North Hollywood, CA 91607
(213) 980-5717

for additional information circle no. 100

THREE NEW PRODUCTS FROM CARROTRON

C610B1 Isolator: Is said to be the only piece of equipment available which positively protects musicians from shocks. The signal is translated into a light beam, projected across the package, and converted back into an electrical signal at the other side. There is no electrical connection between input and output, thus there is no path for a shock. The isolator also has a built-in FET preamp and will fully protect the musician even with dead batteries. Price: $119.95

C821B1 Pre-amp: Said to be one of the best floor box preamps available it is claimed to possess a flatter frequency response than other preamps in use. Carrotron uses discreet FET circuitry, claiming that there is no BI-FET IC that will

perform to their standards. Any signal source from low impedance condenser microphones to high impedance guitar or piano can be preamped by the C821B1 with excellent results. Price: $99.95.

C900B1 Noise Filter: The product filters noise from a signal without affecting its sound, by use of a continuously variable logic controlled high cut filter. The bass and midrange (where there is very little noise) are unaffected by the filter's action. The user sets the threshold at a point where white noise and hum become objectionably loud as the note decays. This noise is then attenuated, so the noise does not fill the empty treble band. The result is natual note decay, which no noise gate can accomplish, according to Carrotron. Price: $139.95.

CARROTRON
2934 Shasta Road
Berkeley, CA 94708
(415) 848-2755

for additional information circle no. 101

APHEX ANNOUNCES NEW B&B AUDIO VCA

Aphex Systems, Ltd., has announced the introduction of the next generation of the B&B Voltage Controlled Attenuator (VCA), model 1538, at quantity prices that reflect significatnt decreases compared to the firm's pricing on its earlier VCA model 1537A, according to Marvin Caesar, president.

Caesar said the new unit represents a step forward in VCA technology. "The 1538 is the product of computer optimized design, which integrated previously external circuitry, and a special high voltage, low noise processing."

It has a bandwidth from DC to over 20 MHz, providing a full range of applications from servo motors to video.

Caesar said that samples of the 1538 will be available in May, with delivery on volume quantity orders scheduled for early September. The VCA is being used by Solid State Logic, in Aphex' own Aural Exciter, in the firm's B&B Audio Products line and in retrofit format in over 1,000 channels in studios around the world with the MCI 500 Series Consoles.

APHEX SYSTEMS, LTD.
7801 Melrose Avenue
Los Angeles, CA 90046
(213) 655-1411

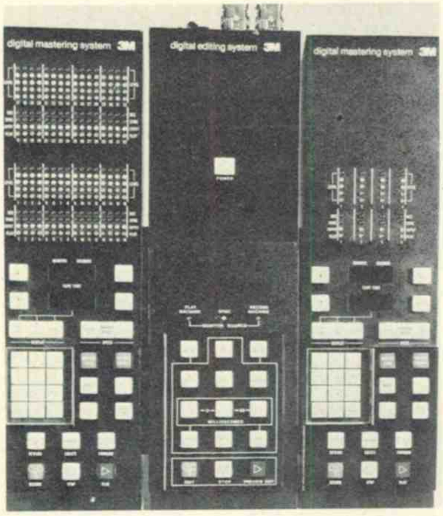
for additional information circle no. 102

3M ANNOUNCES PRICING OF DIGITAL EDITING SYSTEM

Consisting of microprocessor electronics offering extreme precision, risk-free audition or edit preview capability, unaltered originals and splice-free masters, the production version of the system is now available, and is priced at $7,681 and $8,835 depending on interface cabling.

— continued overleaf . . .

The control module, which determines and monitors the tape movement of two 3M recorders, offers special function buttons for determining exact editing points. Refinement can be made by as little as one one-thousandth of a second.



Cross fade capability is also being offered as an option. Cross fade will allow for smoother transitions without abrupt changes, especially during low frequency passages, and virtually eliminates artificial sounds. Added flexibility includes greater ease of selecting a suitable edit point, resulting in considerable time-savings. For a true 10ms cross fade, the old signal is faded-out as the new signal is faded in with error correction. Availability of the cross fade option is projected for late 1980.

3M MINCOM DIVISION
P.O. Box 33600
St. Paul, MN 55133

for additional information circle no. 104

VALENTINO MUSIC LIBRARY SERVICE

The Thomas J. Valentino, Inc., Major Production Music Library and Major Production Sound Effects Library has just announced it has added to its stock of Productiion Music and Sound Effects once again. The firm now has over 155 LPs in its Production Music Library which brings to over 3,000 the number of selections of openings, closing, titles, bridges, and themes that the producer can select from to add as backgrounds to their productions. And the Sound Effects Library has recently been expanded to include 3 new LPs which brings that total up to 21 albums of over 600 different effects ranging from roller coasters, to airplanes, doors, thunder, and crowds and automobiles.

In addition, the Valentino organization just finished with some work in the commercial field as well. The company, in the music area, contributed to the film, "Saturday Night Fever," and its music was used in the soundtrack from the same feature film. In the sound effects area, the company contributed from its stock of effects to the Francis Ford Coppola film, "Apocalypse Now."

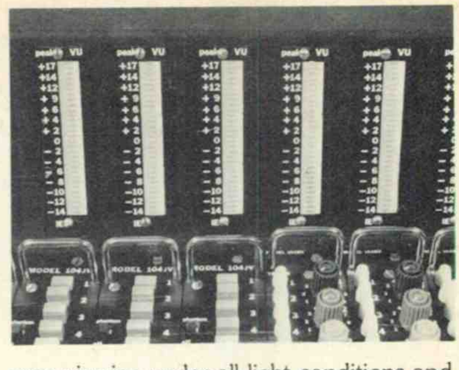
The firm offers the use of its music and effects libraries at very nominal rates to producers in the commercial, industrial, and educational fields, and the extensive catalog can be procured by writing the company at its headquarters in New York.

THOMAS J. VALENTINO, INC.
151 West 46th Street
New York, NY 10036
(212) 246-4675

for additional information circle no. 105

NEW INTERFACE LED VU INDICATORS

Interface Electronics has announced the first in a series of new-technology LED VU level indicators suitable for many applications in audio equipment and contructed as a three-inch bar graph on a 3.5 x 1 inch bezel. A fast attack makes these indicators read peak level, so that headroom can be inferred directly. Slower decay prevents missing a fast peak. Bright wide-angle rectangular LEDs in blocks allow



easy viewing under all light conditions and from any angle. Integrated level-sensors mean small simple reliable circuits. Indicator shows 15 levels from -14 to +17 dB in approximately 2 dB steps using 30 LEDs, with a 1 volt sine wave reading zero dB with a 1,500-ohm input resistor. Calibration can be changed by changing the input resistor, as with an ordinary VU meter. Power requirements is 10 to 15 volts at 250 ma. Depth behind panel is only 2 inches. Small width (less panel space) means more indicators can be used; for example one on every input. Price is $100.00 in single lots.

INTERFACE ELECTRONICS
3810 Westheimer
Houston, TX 77027
(713) 626-1190

for additional information circle no. 107

TWO OPTIONS FOR EVENTIDE H949 HARMONIZER

These new options increase the versatility of the H949 in two important areas — getting rid of "glitches," and computer remote control. The H949 is Eventide's "top-of-the-line" unit, which operates as a pitch changer, digital delay line, flanger, ADT (automatic double tracking) unit with true randomized delay, and all-round special effects device.

The first option is a signal processor/analyzer board, which will eliminate almost completely the "glitching" characteristic common to all pitch change devices which operate by adding or eliminating signal segments. This computerized addition determines the optimum splicing point, thus reducing the potential "glitch" to inaudibility in most cases.

Eventide stressed that this board does not necessarily replace its current splicing methods, which also reduce or eliminate "glitches" in other manners. The company reminds all listeners that *any* pitch change unit will, for theoretical reasons, be less than perfect, and the choice provided by the H949 will allow optimization for all types of program material.

The second option is the Computer-Remote Control board. This enables almost all functions of the unit (including pitch change) to be controlled by an exterenal computer. The unit uses the IEEE-488 standard, which permits multiple units to be controlled individually on a single bus, thus reducing the cost and the complexity of interconnection.

This is the second IEEE-488 product introduced by Eventide: the 1745M digital delay line remote control having preceded it by two years. The unit is completely compatible with the IEEE standard, and can be controlled by low-cost home computers (such as the Commodore PET and the Hewlett-Packard model 85), as well as several automated consoles and industrial computers.

Both the new options for the H949 are designed to fit inside the unit.

Delivery will begin on August 1, 1980, at which time firm pricing will be quoted. At that time, the options will be available both installed in new units, and to all H949 owners for field or factory installation.

EVENTIDE CLOCKWORKS, INC.
265 West 54th Street
New York, NY 10019
(212) 581-9290

for additional information circle no. 108

16 BIT LINEAR, DIGITAL DELAY FOR DISC MASTERING PREVIEW FROM A.M.S.

Advanced Music Systems announces the availability of the new DM-DDS Digital Delay System. This stereo digital delay has been designed to meet and exceed the exacting requirements of disc mastering preview delay. The DM-DDS is available with both analog or digital input/outputs to allow high quality mastering from digital tape machines or superior analog recorders that do not have preview outputs. Compatibility is assured with all future trends in analog or digital formats.

The use of a digital preview delay means ease of setting up and use, independence of tape speed, ability to preview direct-to-disc, and has the added benefit that the cost and complexity of mastering console preview channels are eliminated.

Three preset delays, or keypad entry to the nearest millisecond, program the unit very quickly for different cutting speeds/lathes.

The system is offered in two standard versions:

Frequency Response Version 1 — 10 Hz to 27 kHz +0.5 dB/–2 dB.

Frequency Response Version 2 — 10 Hz to 22 kHz +0.5 dB/–2 dB.

Version 1 Max Delay 1.3 Sec/Ch. Version 2 Max Delay 1.6 Sec/Ch. Distortion — typically less than 0.02% at 1 kHz full output.

External delay modules are available to increase the delay capability to 8 secs/Ch Version 1 and 10 secs/Ch on Version 2.

Version 1 (1 sec/Ch) is priced at $12,400.00; Version 2 (1 sec/Ch) is priced at $10,650.00.

QUINTEK DISTRIBUTION, INC.
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(213) 980-5717

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CERWIN-VEGA SM-12 AND SM-15 PRO STAGE FLOOR MONITORS

The SM-12 two-way vocal spot monitor uses C-V's popular ER124 high-efficienty 12" driver while the higher-output two-way SM-15 incorporates the 3" voice-coil 153V 15" driver. Both systems employ the 120 dB SPL H-25 high-frequency compression driver and horn for dispersion which develops high sound pressure in front of the monitor while allowing off-axis response to slope uniformly.

The monitors can be positioned with the speaker inclined at 20°, 45°, or 55° for long throw to rear stage, high intensity foot monitor, or side fill usage. The low silhouette formats avoid a cluttered stage look and are sized to take a minimum of space in a van or truck.

C-V engineers have selected just the optimum combination of low frequency response, efficiency, and enclosure volume to meet the demanding requirements of professional stage monitoring. Bass response has been specially tailored for accurate vocal reproduction and is essentially flat with a smooth and gradual roll-off (12 dB/octave) below 100 Hz.

Both monitors are capable of producing in excess of 120 dB SPL at 1 meter with their respective inputs of 100 W and 150 W and the tweeters are protected with exclusive auto-resetting circuit breakers.

CERWIN-VEGA
12250 Montague Street
Arleta, CA 91331
(213) 896-0777

for additional information circle no. 113

UREI ENTERS THE AMPLIFIER MARKET

UREI, North Hollywood, California, is announcing its first power amplifier. UREI is noted for its signal processing equipment and its recently introduced line of studio monitors.

According to a company spokesman, UREI decided to enter the amplifier market because this was an area where the company could make a significant contribution to the industry.

The UREI Model 6500 amplifier offers totally modular construction which allows the user to remove either channel for inspection or exchange while the amplifier is mounted in the rack. Each channel is totally independent with its own power supply and even its own continuously variable cooling fan.

With the 6500, UREI is introducing a totally new system of regulating feedback in the amplifier. Through a new system called Conductor Compensation (patent pending), the feedback loop is extended to the speaker's terminals, rather than the speaker outputs on the amplifier. UREI claims the result is extremely high damping factor and near-perfect transient response.

The Model 6500 provides 275 watts RMS per channel at 8 ohms with .1% THD and up to 1,200 watts RMS in mono mode at 4 ohms.

UREI
8460 San Fernando Road
Sun Valley, CA 91352
(213) 767-1000

for additional information circle no. 115

news

FIRST OTARI MTR-90s ARRIVE IN THE FIELD; ARE DUBBED 'THE NEW WORKHORSE'

Otari Corporation has started shipping their new 24-track; 2" recorder, the MTR-90. According to Otari marketing manager, Steve Krampf, "The first units have been very well accepted by the studios receiving them, in fact, from the feedback we've received, we've decided to dub the recorder 'The New Workhorse' since we anticipate that it will become a new production standard in the industry. All of our literature and national advertising will use this theme."

Krampf noted that since Otari products are sold throughout the world, the MTR-90 will be sold in the U.S. in limited numbers this year and will only be available at selected professional dealers.

BGW ACQUIRES TANNOY MARKETING RIGHTS IN U.S.

Brian Wachner, president of BGW Systems, has announced that Tannoy Products Limited has given BGW exclusive U.S. distribution rights to all Tannoy Professional Products.

"We have taken on the Tannoy line to

RECORD PLANT TO BUILD NEW SCORING STAGE AND RECORDING STUDIO

Chris Stone, president of Los Angeles-based Record Plant, (standing in the rubble) has announced that demolition of Studio D has been completed to make way for a new multi-faceted Studio D which will consist of a scoring stage with options for complete television, video, and motion picture scoring — equipped with 35 mm projection.

"Recession is the most opportune time to expand facilities if you are certain that the lull is only temporary," says Stone. "It is mandatory at this time to offer a large facility designed and equipped to service the marriage of the audio and visual arts. In this room we can do a movie score during the day and rock and roll recording at night."

The new studio, designed by Tom Hidley with consultation by Lee DeCarlo (chief engineer at Record Plant), will have the same basic dimensions in terms of ceiling height at Studio C, which most recently recorded "Chicago," "Rod Stewart," and Eddie Money's up-coming albums.

Hidley's design includes a 53 x 48 foot studio with a 22-foot ceiling. The new room houses three iso-booths with a private lounge. It will be technically equipped with a 3M, 32-track digital mastering system and a 48 x 32 SSL Series E console. Lighting director Chip Monck will have installed a fly system with counterweights for easy conversion of any visual lighting requirements.

The grand opening of the new Studio D is planned for Halloween 1980.

further round-out our product line to the professional market," Wachner said. "We have grown to be one of the prime suppliers of power amplifiers to the professional market in just five years, and have expanded our professional line with two electronic crossovers. The addition of Tannoy will give us a line of monitors the quality of which will compliment our strong line of amplifiers."

BGW will distribute the Tannoy Buckingham Monitor, a three-way system in a bass reflex enclosure using a 10" dual concentric driver and two large woofers. The Buckingham provides a single point sound source capable of a 124 dB peak SPL.

Two smaller monitors, the Super Red* and the Classic, utilize a new 15" dual concentric driver, providing the same single point source capability of the Buckingham with SPLs of 121 dB.

A new monitor, the Little Red*, was introduced at the spring AES Convention in Los Angeles.

In addition to the above monitors, BGW will distribute the Tannoy XO-5000 electronics, a stereo crossover with built-in dual channel adjustable time delay, parametric equalizer, subsonic filters and plug-in speaker modules. Plug-ins are available for all Tannoy monitors, plus Altec 604Gs, JBL 4333, 4343, and 4350.

Tannoy 'Red' products are not to be confused with other monitor speakers using a similar color identification originating from Audiotechniques, Ltd., of Stamford, Connecticut.

PYRAMID AUDIO COMPLETES MOVE TO NEW LOCATION

Pyramid Audio, of South Holland, Illinois, has recently opened the doors of their new facility. The new store has taken a step forward from the conventional displays seen in most stores.

According to the owners, Rob Vukelich and Skip Groeneveld, "the store was designed and built with the professional in mind. We've created a working control room to serve as our display area. We feel that the people of today want more than a picture and a price. By building what we did, the customer can sit at the board and patch in any of the equipment we carry. We feel this is the most unique and comfortable showroom in the midwest."

Pyramid Audio is geared to fill the needs of studio owners and broadcast facilities, as well as studio design and consultation work.

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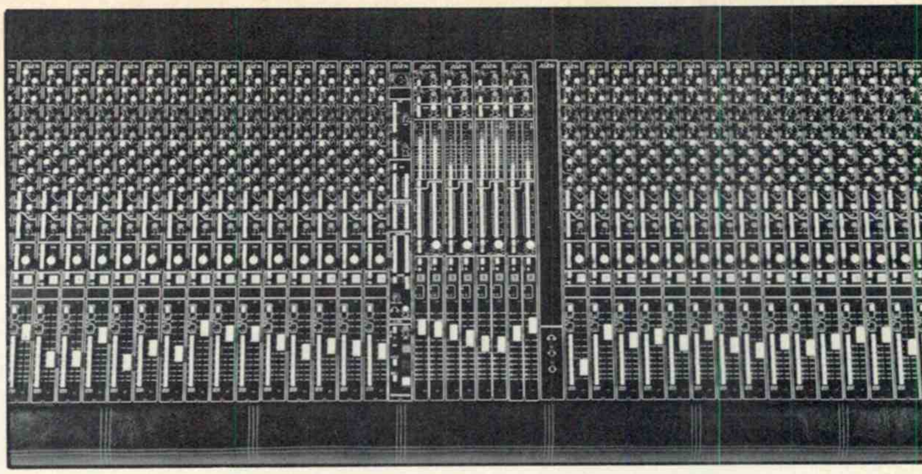
CRYSTAL CLEAR RECORDS IN dbx® FORMAT TO BE INTRODUCED

dbx, Inc., of Newton, Massachusetts, and Crystal Clear Records, of San Francisco, have announced plans to issue albums from the Crystal Clear catalog in the dbx® Encoded Disc format.

According to Ed Wodenjak, president of Crystal Clear Records, "The dbx Encoded Disc is an extraordinary development in recording technology. We are pleased that



HARRISON DESIGNS LEASING PROGRAM FOR THEIR 'ALIVE' SERIES PERFORMANCE CONSOLES

Harrison Leasing Corporation has announced the formation of a short-term rental program for its Harrison 'ALIVE' Series of live performance consoles. The leasing company is a subsidiary of Harrison Systems, Inc., the Nashville based console manufacturer.

Offered as either a 24-input, or a 32-input model, the ALIVE features four main stereo output pairs and eight auxiliary send busses. The ALIVE extender frame, which can be fitted with up to 32 additional input modules, is also available through the rental program.

Rental rates are computed using an initial base fee, plus a per-diem charge. Minimum rental time and security deposit are negotiable.

Further information on this new rental plan may be obtained by contacting Bill Owen at Harrison Leasing, P.O. Box 111013, Nashville, TN 37211 — (615) 832-7431.

dbx has elected to issue several of our titles as Encoded Discs."

The initial offering of dbx Encoded Discs will include Volumes I and II of "Sonic Fireworks," (a collection of music for organ, brass, and percussion), "New Directions" by Laurindo Almeida, and "Taj Mahal Live."

"The best musical performances available will be included in the new releases since master tapes recorded during the original direct-to-disc sessions are being used," advises Jerome E. Ruzicka, dbx vice president and director of the dbx Encoded Disc Program. "Also, each album has been expanded to include previously unreleased selections by the artists featured on each album."

More than twenty record companies are participating in the dbx Encoded Disc Program. The Crystal Clear Records now being re-mastered represent a move to produce in the dbx format, previously released direct-to-disc "audiophile" recordings.

The dbx process virtually eliminates record surface noise while reproducing the full dynamic range captured on the master tape. dbx recently introduced its new Model 21 Decoder which enables the dbx Encoded Discs to be played through a stereo system. The Decoder retails for $109.

URSA MAJOR EXPLAINS REVERB WITH THE SPACE STATION SST-282 DEMONSTRATION CASSETTE

A cassette recording explaining and demonstrating reverberation and related effects has been introduced by URSA Major, Inc., of Belmont, Massachusetts, at a price of $2.00 per cassette. The tape's listeners will have a chance to hear reverberation, delay, echo, space repeats, combs, doubling, flattening, and more, while the effects are being explained. The URSA Major Space Station SST-282, a versatile digital reverberation system, produced all the effects on the cassette, and the unit sells for a low professional net price of $1,995. The recording demonstrates the Space Station's wide variety of uses in the recording studio and is available from URSA Major, Box 18, Belmont, MA 02178.

Another cassette focusing on broadcast applications is available at the same price.

SOUND WORKSHOP ANNOUNCES EXPANSION

Sound Workshop Professional Audio Products, Inc., a leading manufacturer of state-of-the-art recording consoles, recently completed major expansion of its Hauppauge, New York, facility, according to Michael Tapes, president of Sound Workshop. The extensive physical expansion of this facility follows recent enlargement of the company's research and design division in Melville, New York, and illustrates Sound Workshop's spiralling growth. The Hauppauge plant, located at 1324 Motor Parkway, has doubled in size all phases of facilities including its marketing, sales, manufacturing, technical services, and shipping departments. In addition, a newly enlarged computer system, featuring expanded hardware and software, has just been put on line to further increase computerization of production and material control, as well as provide more concise controls for sales and marketing.

Newly-appointed general manager Gil Colquitt, formerly a regional executive with the Columbia Records Division of CBS, will oversee the company's on-going operations expansion.

Describing expansion of the five-year-old company's Hauppauge facility plant, president Tapes said: "In 1979, we saw a sales increase exceeding 60 per cent. In 1980, the introduction of the new Series 30, which is now in total production, plus continually strong sales in our 1280 and

. . . continued on page 134 —

STUDIO UPDATE

. . . continued from page 28 —

■ **THE MUSIC WORKS** (North London, England) is co-owned by producer **JO JULIAN** and studio manager **AL WILLIAMS,** and features an L-shaped studio floor which can accommodate up to 20 musicians. The control room is centered around an Allen & Heath Syncon 28 x 24 console linked to a Lyrec TR 53 24-track recorder, and offers such outboards as Roger-Meyer noise gates, a homemade echo plate, a Grampion spring reverb unit, and an AMS DM2-20 flanger/phaser. *Highbury, North London, England.*

■ **DOWNTOWN RADIO** (Belfast, Northern Ireland) has opened its production facility to outside bookings. The studios house a Raindirk 26 x 8 Series III console linked to a Lyrec TR 532 24-track recorder, as well as outboard gear featuring an URSA Major Space Station, an Audio & Design Scamp Rack, and an AKG BX-20 reverb unit. Montiors are Chartwell loudspeakers powered by Quad 405 amps. The control room is for two studios, one with 1,350 square feet of floor area, and another with 500 square feet of space.. *Kiltonga Radio Centre, Belfast, Northern Ireland.*

West Germany:

■ **RUSSL TONSTUDIO GMBH** (Hamburg, West Germany) has purchased a new MCI JH-538-38-LM console with MCI automation. It will feed an MCI 24-track recorder and MCI and Studer 2-tracks for mixdown. Monitors in the studio are by Eastlake and Auratone, powered by AMCROWN DC 300A amplifiers. Other gear includes Allison Gain Brain limiters, Rebis parametric EQ, Allison Kepexes, RM noise gates, and Eventide digital delay. Mikes are by Neumann, Sennheiser, Shure, AKG, and Electro-Voice. The studio is an Eastlake Audio **TOM HIDLEY** design with instruments including pianos by Steinway, a Hammond B-3 organ with Leslie, and a number of synthesizers. **HANS-JURGEN STEFFEN** is the studio manager. Russl also maintains a mobile recording van. *Lenffohrdener Weg 21, 2000 Hamburg 54, West Germany. Telephone: (040) 578-113.*

Austria:

■ **E.C.S. STUDIOS** (Austria) has purchased a new Trident Series 80 recording console. The studio owner is **GERHARDT ENGLISCH.** *Grundung, Muhlberg 2, 4043 Lichtenberg, Austria.*

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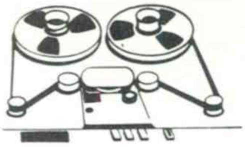

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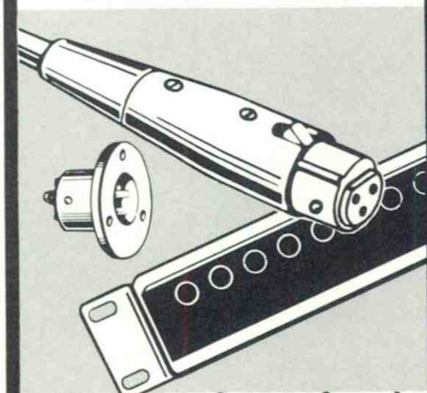

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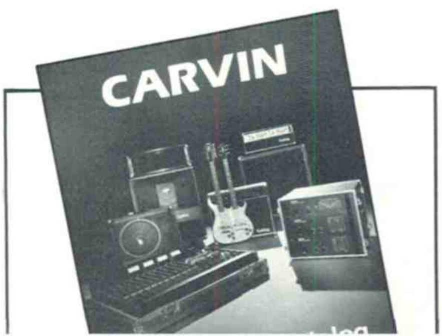

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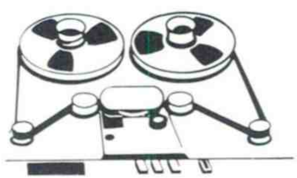

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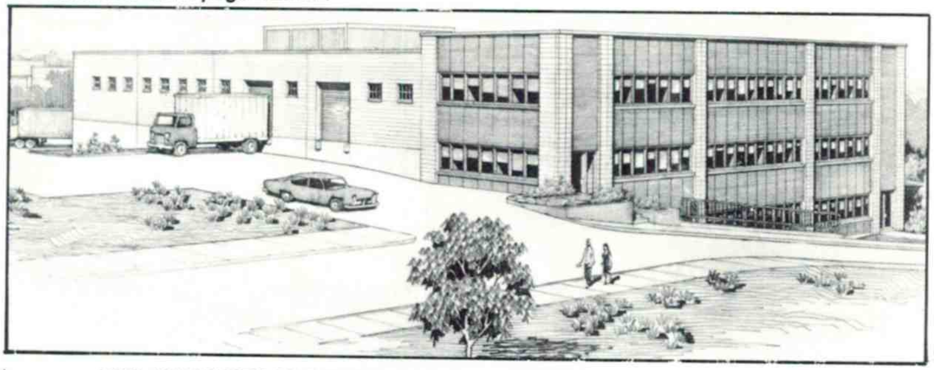


AKG CHANGES NAME TO AKG ACOUSTICS, ANNOUNCES MOVE

The AKG Division of Philips Audio Video Systems Corporation, 91 McKee Drive, Mahwah, New Jersey 07430, will become AKG Acoustics, Inc., effective immediately.

"Our sales and executive offices located in Mahwah will be consolidated with our order administration, warehouse, and service departments, which are presently located in South Norwalk, Connecticut," Andrew A. Brakhan, vice president and general manager, stated in announcing the physical move and organization name change.

"It is going to be much simpler to get results, obtain answers to questions and follow things through to completion. We are entering an era in which client service and attention to detail will be even more helpful to our dealers, and we are determined to stay on top of any situation in which our attitude, promptness, and service can be of help to our customers. Relocation under a single umbrella assures me that we will succeed in our objectives," Brakhan concluded.

As of August 1, 1980, the new address for AKG Acoustics, Inc., will be 77 Selleck Street, Stamford, CT 06902. Telephone: (203) 348-2121.

Series 1600 console line truly necessitated this major expansion.

"The Series 30, an abbreviated version of the top-of-the-line Series 1600, represents the most cost-effective console in the market today. For the first time, the expanding or budget-restricted studio can afford a console that is sonically equal to the finest available, at a price that is within the boundaries set by today's economy. Tremendous interest in the Series 30 was generated at the Audio Engineering Society Show held in Los Angeles last May."

CLASSICAL MUSIC RECORDING WORKSHOP

A workshop on the recording of classical music will be September 24 and 25 at the University of Wisconsin — Eau Claire.

Instructor for the workshop will be Russell Borud, an independent audio engineer and author of numerous articles on recording techniques.

The workshop will emphasize natural stereo recording techniques and is open to anyone interested or involved in the recording of classical music. Participants will learn to assess recording and playback environments and the selection and placement of microphones.

"Most recording workshops deal with multitrack studio techniques, and, consequently, are of little value to persons who record mostly live performances," according to Burton Spangler, UW-EC audio coordinator. "This workshop will help recordists increase their knowledge and skills to produce more natural-sounding recordings."

The workshop format will feature lectures, discussions, critiques, and recording and editing demonstrations.

Participants who complete the course will earn two Continuing Education Units (CEUs). Fee for the workshop is $45 and includes printed instructional materials, two lunches, dinner on Wednesday, and coffee breaks.

More information is available from Spangler at (715) 836-2651.

The workshop is sponsored by UW-Extension in cooperation with the University of Wisconsin — Eau Clair Media Development Center.

. . . continued from page 21 —

What The
AUDIO/VIDEO fusion
Can Mean To The
RECORDING STUDIO INDUSTRY

— part 3 in the series —

. . . for those brave enough to try —

PRODUCTION VIDEO EQUIPMENT PACKAGES

by Steve Barnett

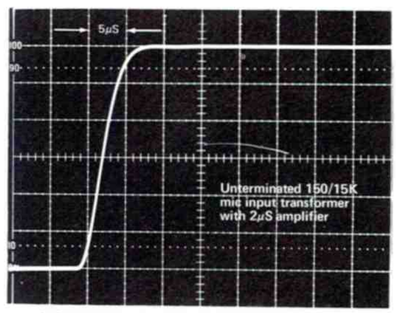
Monitors from JVC that fit into this general quality area are their TM 41 AU monitors. Suttle recommends three at $395 apiece.

A package of this equipment, using the RM 88 editor would cost approximately $23,085. This does not include an audio system or accessories, such as cables, tripods, or tape. Almost any audio signal can be fed into the tape machines, and its source depends on your audio needs and situation.

For a portable system, Suttle offers JVC's battery operated U-matic machines. "There's the CR 4400 U, which is a ¾-inch battery powered recorded, and then there's the CR 4400 LU. That unit is capable of performing edits in the field. The CR 4400 U is $2,850 and the LU is $4,410."

Using JVC for discussion of isolation systems stems from the absence in their line of a switcher for performing cuts during production. Suttle adds, however, that these systems are compatible with most of the switchers on the market in this price range.

— continued overleaf . . .



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VIDEO PRODUCTION PACKAGES for the RECORDING STUDIO

For assembling a package of components from various manufacturers to best suit one's needs, a video dealer can be approached. Like those in any other field, the video dealer can offer a broad selection of professional video gear along with consultation on which of these manufacturer's products can best suit individual needs. An additional advantage is being able to purchase the video accessories in the same location.

Hoffman Video in Los Angeles, as an example, deals primarily in broadcast quality equipment, but also is involved with so called "industrial" video. Hoffman sales representative Doug Jeffs, and head of technical services Bill Wallace, put together a package based on two cameras feeding a ¾-inch U-matic cassette recorder using a switcher to shuttle back and forth between cameras during the actual taping. Decisions in this sort of shooting situation are made on the spot by the director.

Cameras offered in this package are by Hitachi, the GP 7-A with Gen-lock, and are priced (retail) for $4,250 apiece. Video sync for accurate swithing would be handled by a Tektronics 1474 Sync Generator and a Sigma Video Distribution Amplifier, Model 100 A, respectively costing $1,555 and $305.

"For the switcher," says Wallace, "I'd probably go with a 3M 3100, which is a relatively low priced unit ($5,000), but it's a quality switcher. For the monitors, you'd need two black and white units, one for each camera, and two color monitors, one to give you the preview and the other the program (the first to check an effect before recording it, the second to monitor what's actually being placed on the tape)."

For the camera monitors, Wallace suggests the Panasonic WV 5302 dual 9-inch black and white unit, which runs $565, and for the color models the Video Tech VM 8 PRD dual 8-inch monitor configuration for $2,095. For a more accurate picture, Wallace suggests the incorporation of a waveform monitor into the package.

"You need some way of balancing the cameras," he explains, "because you've got the overall gain of the camera, what the outputs are, plus the pedestal, which is the black level for matching to, to give you contrast control. You can't really just set that up on a normal monitor. You don't know which one is right, you need some way of measuring the outputs of the cameras and then matching them so that they are the same. For that purpose I'd look at the Tektronics 528 which runs $1,995."

With regard to recorders, continues Wallace, "If you're recording something, and you know you're not going to edit it, but you'll use it as is, you could get away with a lower priced deck without any problem, but if you want to go out and edit it later by renting editing time I would suggest paying more for the recorders."

"If you're serious about editing," suggested Jeffs, "the Sony VO 2860 A is extremely good. The only way you could improve it would be to leave ¾-inch for a one-inch broadcast deck. It's $6,625."

"Keep in mind that these are all list prices," adds Jeffs, "and with a package like this, we'd sit down and work from cost. So you're looking at $27,000 retail but it would be in the $23,000 to $25,000 price range working from cost and including the accessories."

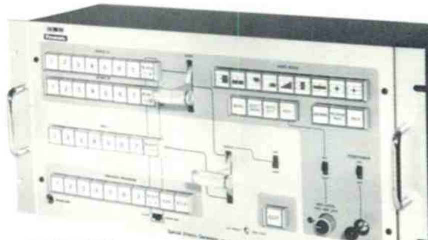
Of the systems discussed in this article, the one from Hoffman was priced about the highest, mostly because of their advice to go for more high-end equipment within the boundaries set out, as well as their addition of the high quality testing equipment to maintain the gear.

One of the manufacturers offering equipment in all the formats which have been mentioned thus far is Panasonic. Mike Murray is the West Coast Regional Sales Manager for the company's industrial video division.

After running down the Panasonic equipment available in this general price range, Murray added, "I think this is starting out quite heavily. My first impression would be to recommend that you spend quite a bit of money on a good color camera and a porta-pack, and then rent editing time to see if it's what you want to do. It all depends on just how heavily committed they are about entering this field."

"We have a good camera which is a step up from th rest of our line. The WV 3900 sells for around $6,000 and is a portable version of our Nuvicon camera. It's good in low light level, and it's very difficult to burn this camera with high intensity lights."

If he had to recommend a ¾-inch cassette system in this range, Murray suggests that one begin with two of these cameras, a WJ-5500-A production



Model WJ-5500

switcher that retails for $3,950 and allows for special effects between cameras, a 9240 ¾ U-matic video cassette recorder-player, which lists for $4,500 and can be later integrated into a Panasonic editing system, a WV 5203 triple five-inch black and white monitor configuration, at $825, and a CT 1910 19-inch color monitor for the line position (what's being recorded on tape) at $625. The sync lock mechanisms and the sync generator are built into the cameras and switcher respectively.

This brings the total package less accessories, to $21,900. Post-production editing, he suggests, should be rented out, as investment in an editor controller and an editing cassette recorder would cost another $10,000-plus. Murray isn't really certain if this large an investment is advisable for anyone new to video, suggesting that ¾-inch might be too heavy an investment for someone who's just exploring the field. They might be better off going with a ½-inch machine and taking it to a post-production house and editing it up to ¾-inch. As a starter this would make the hardware costs very reasonable."

1/2" Format

Panasonic manufactures an industrial line of ½-inch VHS video and equipment, which Murray is careful to differentiate from the home VHS gear.

"With this gear they could experiment

Sony ½" Betamax editor recently introduced

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VIDEO PRODUCTION PACKAGES for the RECORDING STUDIO

continued

(though without switching) to see if they liked it. I would recommend that someone who wants to dabble in video to see how they like it should consider a WV 3900 which is our high quality Nuvicon camera and the NV 8410 ½-inch VHS porta-pack. For that you're talking under $10,000. Then you can take the tape to one of the ever-increasing number of production houses that can transfer ½-inch up to ¾-inch. With regard to the audio," continues Murray, "we're assuming that the stage is already miked properly and we're just picking up an out-line program feed."

Murray also felt that a "single camera set up without distracting cuts may be more effective video demonstration for a band than a highly produced presentation. Hence, the less expensive ½-inch system would suffice."

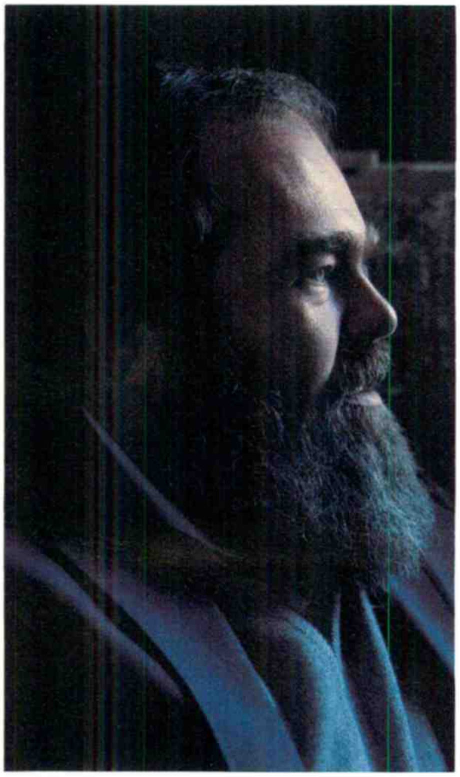
One should keep in mind, that nearly every manufacturer spoken to made mention of developments currently underway in the area of ½-inch editing systems, in an attempt to make its quality on a par with the ¾-inch formats of a few years ago. It may even be wise to take a wait-and-see attitude since industry insiders are predicting the availability for both VHS and Beta format editing systems within the year. If their expectations are realized the savings in hardware costs could be enormous, and, with the recent addition of stereo audio in these formats, the advantages of ¾-inch may prove negligible.

Murray sees the ½-inch tape systems as quite suitable for club circuit demos even today, and he mentions his reservations about ¾-inch demos for record companies. "These are people

The very recently introduced Panasonic 1/2 inch production system

who spend time at the networks and the big production houses. I think that what you might end up doing with a less than professional ¾-inch demo tape is to focus attention more on the minimal editing or production techniques rather than how good the performers are." In further consideration of the ¾-inch format for this use, Murray questions the value of a mid-range product which may find itself as neither fish nor foul.

for additional information circle no. 131



The music industry is now beginning to recover from one of the most negative economic periods in its history. A time such as this can be a very positive thing if we use it as a learning experience.

The most important lesson learned by all of us is that *there is a limit* to the economic return that can be achieved from our daily business activities.

The record labels have reacted. They have instituted more prudent pressing and distribution practices, trimmed artist rosters, and limited budgets for new products.

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