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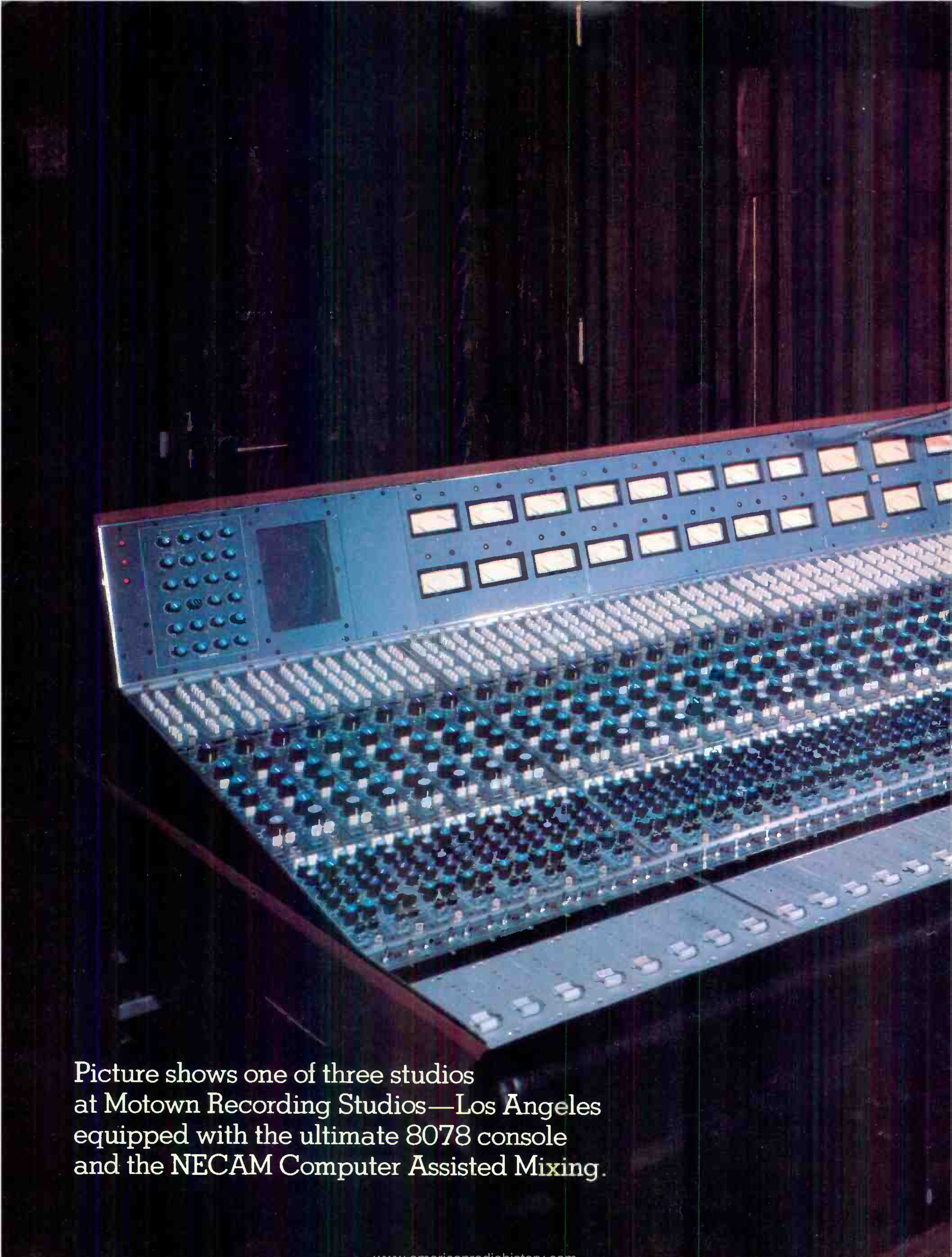

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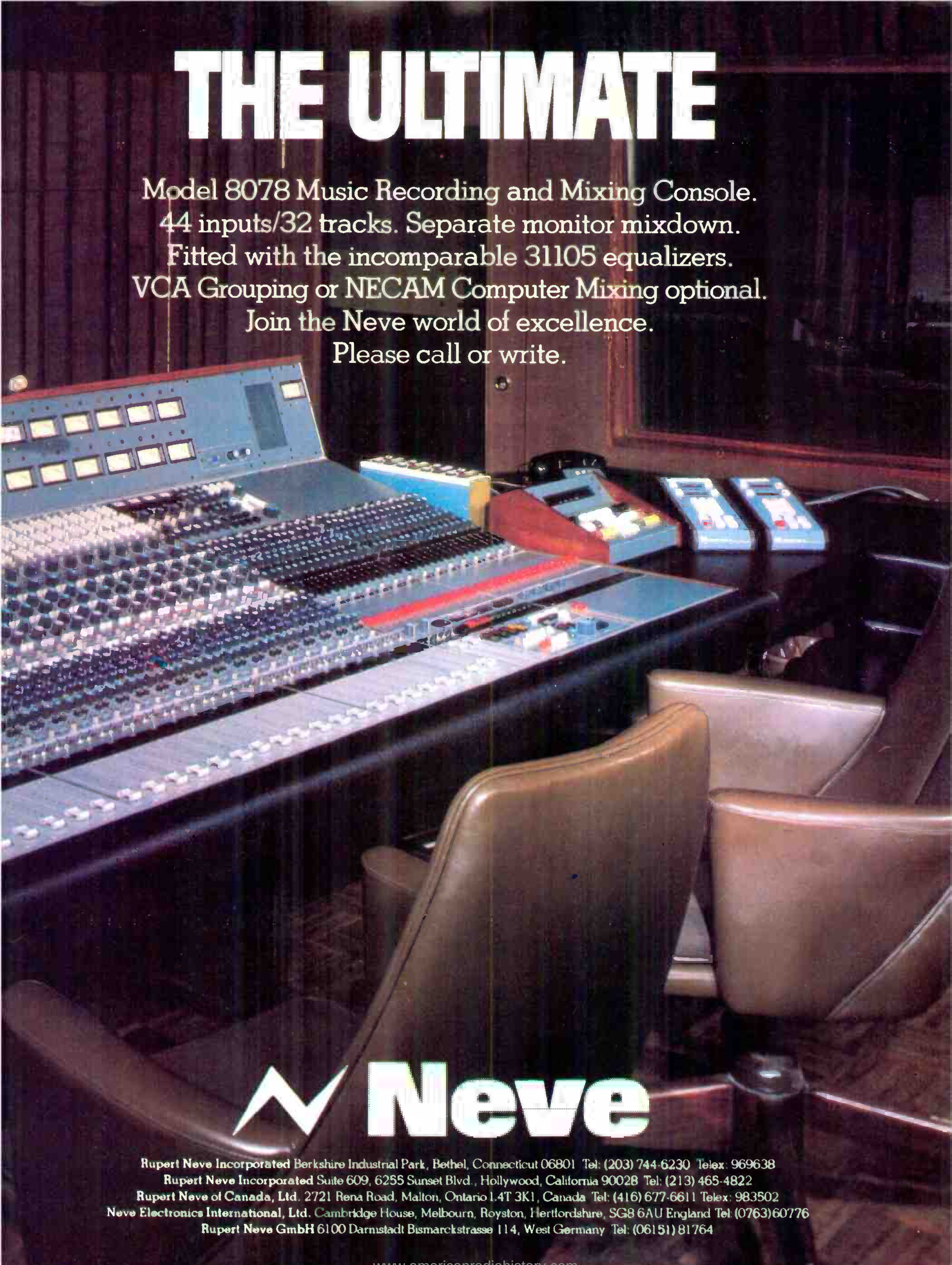
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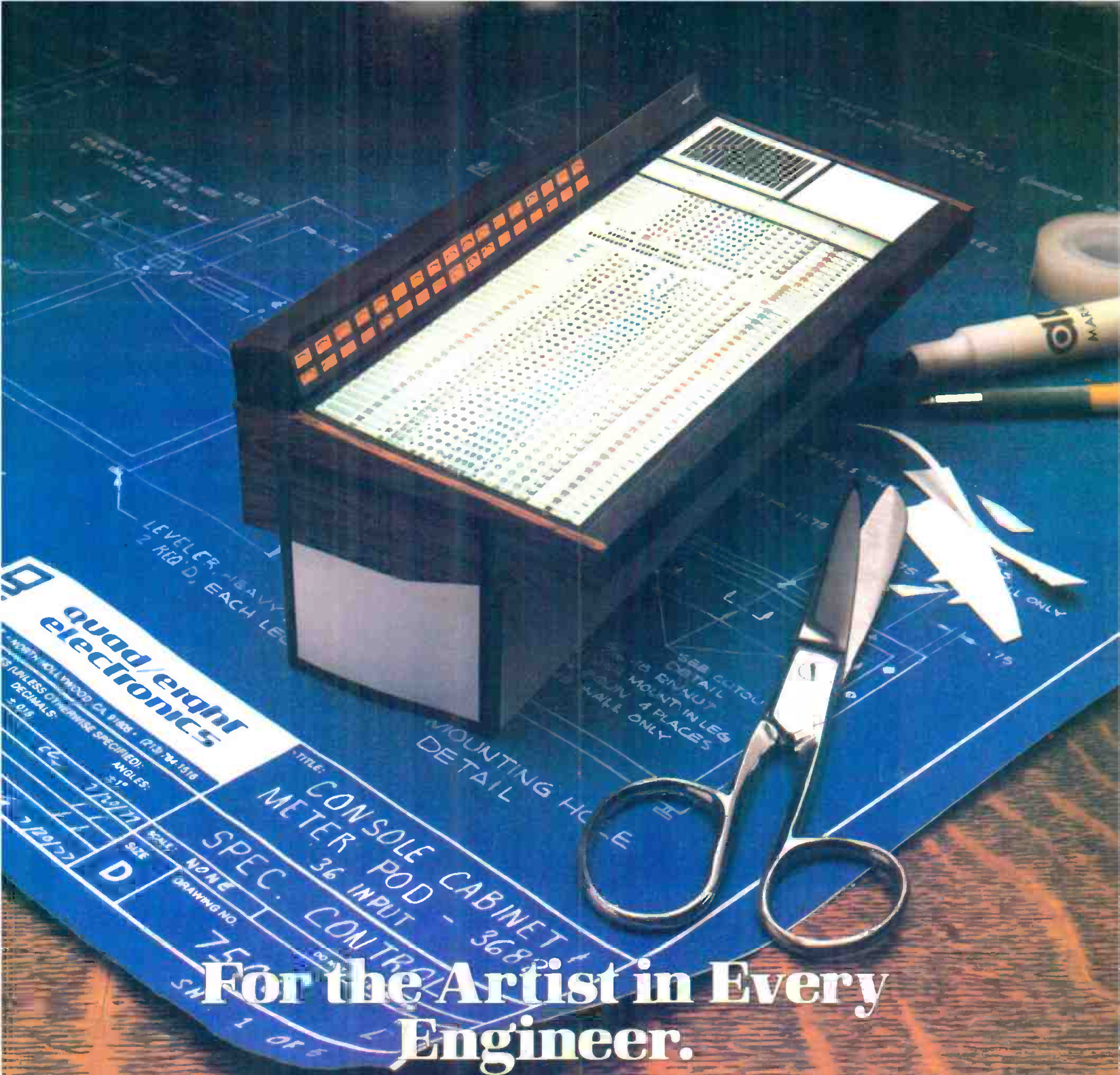

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Letters

from: **Brian Cornfield, President**
Everything Audio
Encino, CA

For lo these many years, we at Everything Audio, as well as others, have been reading of the great number of studios and mind-shattering advances in the area of acoustical design for which the Sierra/Hidley organization have taken full credit.

I feel that it is about time that this statement be corrected. Their public relations job being done on our industry is a good one, granted, and to that end I congratulate the advertising agency, but when public relations clouds the reality of our industry I must protest. I would hope that credit for advances in our industry would be given to those who are deserving, instead of those bold enough to claim advances that are not truly and wholey theirs. This endless supply of self-grandisment has been going on for so long that the Sierra/Hidley team should be getting tired of patting themselves on the back. Their statements reflect a belief that they are the only organization capable of acoustic advances.

What about the years of work by such notables as Mr. George Augspurger, Mr. Don Davis, Mr. Jack Edwards, Mr. Michael Rettinger, Mr. Jeff Cooper, Mr. John Storyk, Mr. Bob Toddrank, and ourselves at Everything Audio. When Mr. Tom Hidley went to Europe and started Eastlake Audio, the entire acoustic design community continually worked on advances. Our own introduction of ear level, direct field monitoring and other designer advances, including Mr. Don Davis' contribution, served to move the industry forward. The concepts are evident in all advanced facilities. Interpretation of PZM™ microphone curves by Mr. Davis gave us an "onion skin" picture of control rooms. Recent time-aligned speakers that UREI have developed have aided the community in co-axial studio monitoring.

In a June, 1979 *R-e/p* letter, Mr. Kent Duncan and Mr. Tom Hidley again got a chance to appear in print and sadly we were treated to more public relations. In this letter, Mr. Duncan is answering a question from Mr. Rettinger regarding a 3 dB or a 6 dB build-up in a "free field." Their points do not warrant re-discussion except to say that Mr. Duncan re-defined "free field" for purposes of defending his statement and Sierra/Hidley control rooms; some of which have a 5 - 6 dB center build-up. This method of argument is invalid in a scientific community. If we are to improve the state of acoustic monitoring, we *all* must abide by certain facts, rules, and laws. The goal of the design community should be to create "real" not modified field acoustics, and it should be stated if a designer is working under modified conditions.

Other than being clinically incorrect, it is an assumption to use the term "exclusive" when describing this 6 dB add condition. Not only is it the creators option that this condition is desirable but it leads the reader to believe that no other control rooms exhibit this phenomenon. Painfully obvious in all Sierra/Hidley writings are the ever present public relations buzz words. In past time, such things as control room equalization and random fields were equated with quality and success by the Sierra/Hidley organization. These concepts, again, were stated to be good by their own creators. Today, however, we see a 180-degree change in opinion. Statements about control room equalization are now referred to as devices that are only incorporated in poorly designed rooms, not Sierra/Hidley designs. In a recent *Recording Engineer/Producer* letter, Mr. Kent Duncan states the Sierra/Hidley opinion that some rooms were too random for good sound reproduction. This remark is directly opposed to earlier Hidley statements and designs. Their experimentation with clients' money resulted in proving themselves wrong. Learning by experimentation is fine, but the client should be made aware that he is funding a research project, not buying ultimately correct information.

When Mr. Hidley left in 1975, his rooms had tier-drop ceilings, high monitor placement, minimum front traps under the windows and other "Westlake" fingerprints. A read-through of recent articles and a look at some recent rooms reveals two basic areas of change. The first change is the pretentious attitude and the second is simiarlity to United States designs. Now, Hidley rooms look more like United States design work except for the ever-present rock walls and speakers.

The hard tier-drop ceiling is now soft as in other designs. Recent Kent Duncan rooms have monitors placed much lower than the earlier designs. These changes indicate a direction influenced by the American industry. Such statements as, "I look forward to discussing some new live control rooms . . . particularly now that American studios and designs are following suit with more control room variation," indicates a disregard for origination credit. Perhaps Mr. Hidley thinks that the rest of the acoustic community either has a short memory or that no one will take the time to dispute his statements. Mr. Hidley goes on to state, "The character of a control room is the sum of direct and random field signal." Work going on indicates that a control room should and can be designed to not have a character and not have random field signals introduced. Direct, coherent sound is what it's all about today. Just consult a contemporary United States designer. All this evidence to the contrary, Mr. Hidley still wants to combine good with bad and come out with better.

One final point, the purpose of this letter is not to take away from the Sierra/Hidley advances. When flush mounted speakers were laughed at and 604s on chains were the most common system, they made great steps in many areas, especially the beautifying of the control room. But please give credit where credit is due. The accomplishments of American-based designs are major, and although we are in a competitive market, we must recognize that the evolution of the acoustic design industry is the net result of a community of professionals and does not owe its creation and improvement to one man.

---

reply from:
Kent R. Duncan
Sierra Audio Corporation
Burbank, CA

I confess to being a bit confused by Brian's letter. He calls for more use of technical terms that we all seek in the analysis of the rooms we build, however, instead of responding in kind to the many technical points made in my article, he coins a new technical term, "onion skin."

I fail to understand Brian's bitterness in his attack on Tom Hidley. Of the eight experts listed by Brian, Messrs. Augsburger and Edwards are, of course, early associates of Mr. Hidley's; Mr. Cooper, Mr. Todrank and Brian Cornfield all were employees of Westlake in the early years.

Research on rooms exhibiting live mid-band decay times which started this year with articles that follow this one, parallel work started by Tom in 1975. And one

continued overleaf . . .

RECORDING engineer/producer

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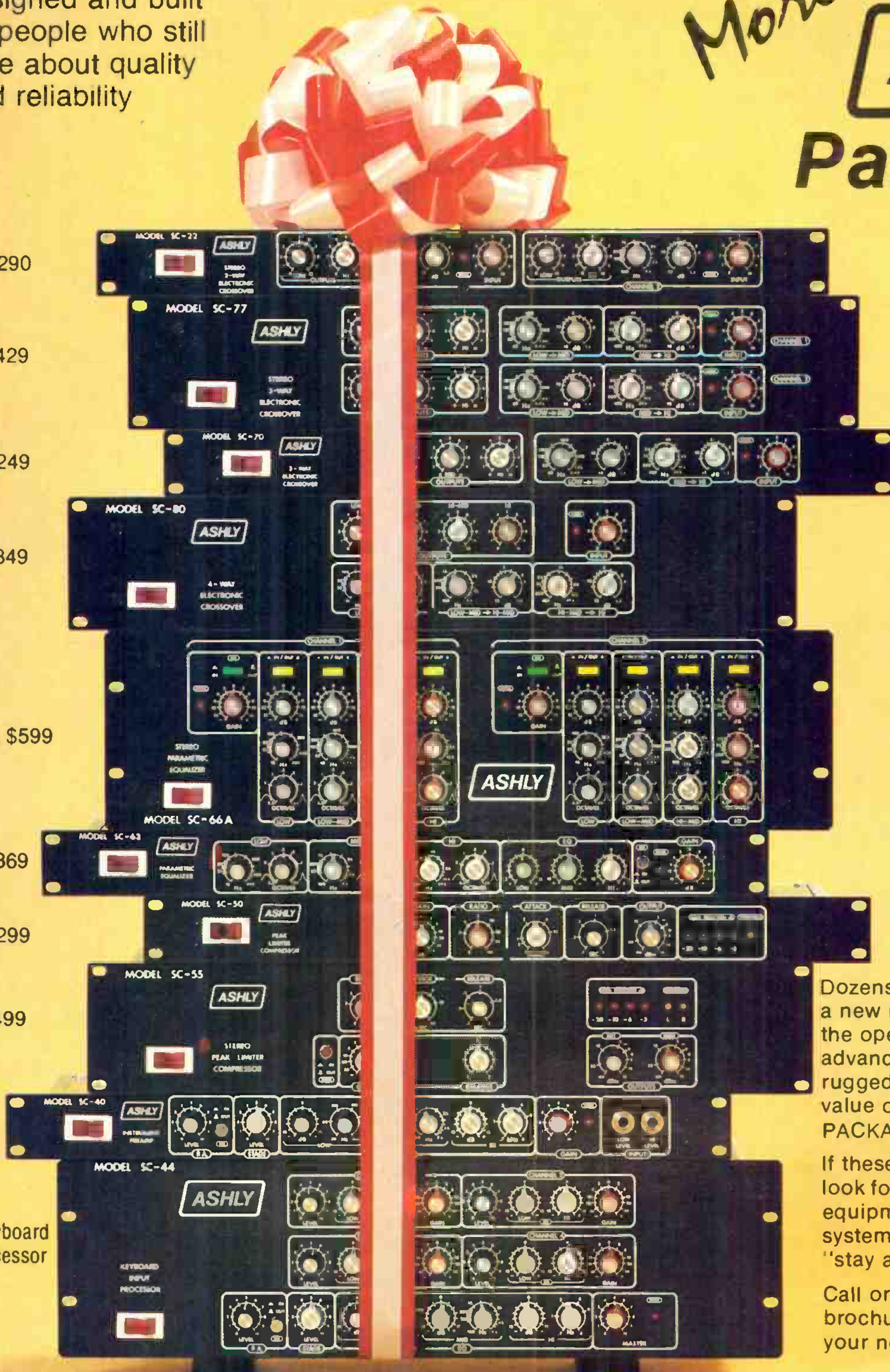

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Kent Duncan
(continued)

contribution I am happy to make is to head a discussion such as you are reading in these pages.

Brian dislikes our advertisements (I don't have an agency, I write them myself), but I can only answer that Sierra/Hidley has built or consulted on about as many projects as all the gentlemen mentioned above, combined.

We are attacked for our 5 - 6 dB center build-up which Brian tells us "... (is present) in some of our rooms ..." In fact, each one of the last 132 studios we have had the privilege to design has been in excess of 5 dB. A very concise letter from John Eargle of JBL follows this letter explaining better what I wanted to relate in my article.

Brian calls us a "scientific community." All the esteemed people mentioned, as well as Sierra, would like to come up with more objective tools to specify these difficult parameters that are control room sound. In our own quest to devote more time to refinement of our consulting work, we became totally disinvolved with recording equipment sales (save our monitor system which is an integral part of our design). I salute Brian for being able to be "everything in audio" and supply gear, design and build rooms, disregard all he learned from Tom Hidley, and turn us around overnight to being a "scientific community."

In attacking us for "changing our views based on experimentation with client's money," I protest. Would Brian prefer to sell cookie cutter designs that will perform identically and never improve? Our entire two-hundred-twenty room history has been one of investigation, refinement, experimentation, improvement and questioning the parameters of our measurements. I am confident we will never stop learning.

While we know we build world class structures that perform acoustically and are profitable for our clients, all of us know there is a good deal going on in the rooms that is not being measured or described. Since 1975 when we departed from our basic design and began trying things that seemed to need experimentation, our performance guarantee has become much tighter in specification and much broader in parameter. Our clients are protected by this guarantee and no other designer we know has attempted to guarantee his work.

Let us close with the idea that more standardization in measurement, in technical terms, and performance specification would benefit us all. We must serve those in our craft who truly set standards, the engineers whose subjective judgement guide us in understanding industry needs.

from: **Michael Rettinger**
Encino, CA

This is in response to Kent R. Duncan's reply in the June, 1979 issue of *Recording Engineer/Producer* to a letter of mine in the May issue in which I took exception to a previously published statement of his, according to which, in a free field, with a single loudspeaker radiating a broadband signal, a 6 dB power increase results theoretically when a second identical monitor is fed an equal amount of power. The first two sentences of his reply are reproduced below:

"It is not my intention to debate the definition of a free field. Suffice it to say that in discussing power additive conditions in our control rooms which we designed to be acoustically two-dimensional by virtue of porting to the trap cavities we have completed five recording studios with a 6 dB add, broadband."

What does this mumbo jumbo mean? What is an acoustically two-dimensional control room? A physical wave has three dimensions and only a shadow can have two. And again he insists on a 6 dB power level increase of two speakers over one. The man is either a magician or he does not know the difference between sound power level and sound pressure level. Hopefully, the discussion below will clarify the situation.

To begin with, a free field room is an enclosure whose boundaries absorb all the sound incident thereon; in other words, it is an anechoic chamber, representing outdoor conditions.

Sound power is the sound intensity multiplied by the area through which the energy flow takes place; more simply, it is the energy per second. Unfortunately, sound intensity cannot easily be measured for the lack of proper instrumentation, so that recourse is taken to the measurement of the mean-square sound pressure, to which intensity is proportional. Most sound level meters are calibrated in terms of the effective or RMS pressure, while oscilloscopes can measure peak pressures.

A broadband signal is one with many frequency components. In his article, "Studio Design Requirements for the Next Decade," Duncan mentioned pink noise and warble tones for his testing. A pink noise source has many sinusoids with constantly varying amplitudes and phase relationships, and a warble tone has sidebands; this means that both types of signals are complex waves.

When, in the open, a loudspeaker radiates a complex signal and a second unit is made to radiate a like signal in the same direction, so that at the point of observation twice the sound power prevails, then both

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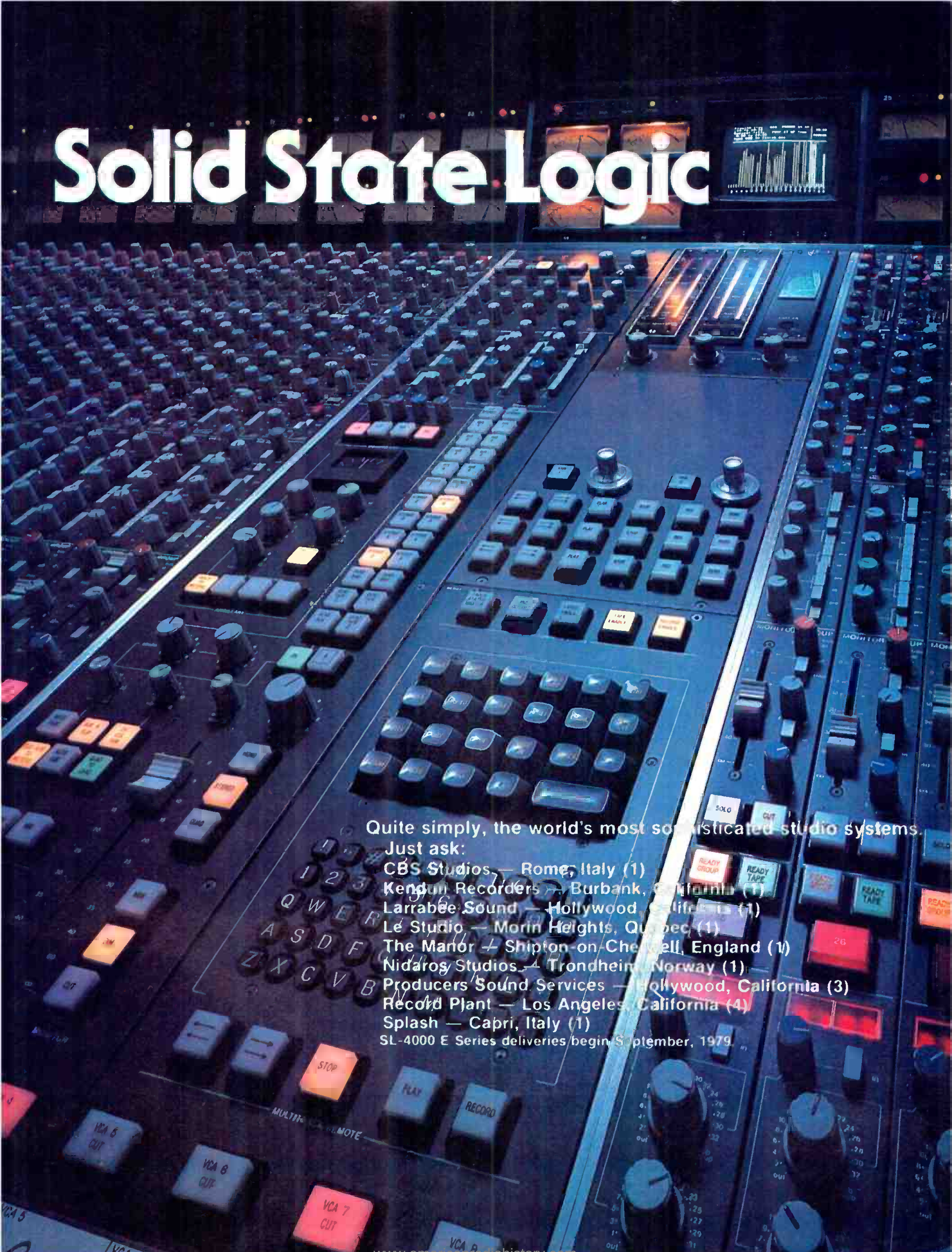
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Michael Rettinger
(continued)

the sound power level and the sound pressure level increase by 3 dB. When we have N like sources, like automobiles or pumps, close enough together so that their distance to the microphone is essentially the same, then the sound pressure level of the noise as well as its sound power level increase by 10log N. Thus, when N is 10, these levels increase by 10 dB. This is for the reason of the constantly changing phase relationships between all the components which make up the noise. The mathematics of the subject is contained in Article 42a in Volume 1 of Lord Rayleigh's "The Theory of Sound."

When, in the open, a loudspeaker radiates a single frequency, and a second identical emitter is placed next to the first and is made to radiate a sinusoid of the same frequency and of equal amplitude to the first, then at a given central point of observation, where again twice the sound power prevails, the sound power level increases by 3 dB but the sound pressure level increases by 6 dB. This is not true when the second speaker radiates a frequency different from the first, even though it may radiate the same amount of power; in this case, both the sound power and the sound pressure level increase 3 dB.

Towards the end of his letter in the June issue, Duncan says further: "While we are still learning the field, I submit that a room measuring a 3 dB power add is far behind the state-of-the-art."

I submit that a promise of a greater than 3 dB power add is a misleading promise.

---

from: **John M. Eargle**
James B. Lansing Sound, Inc.
Northridge, CA

In the June, 1979 issue of *R-e/p*, Messrs. Duncan and Rettinger report positively and negatively, respectively, on sound field additions of 6 dB when going from one monitor system to two systems. The following comments may clarify the situation:

If a monitoring environment is symmetrical, both structurally and electro-acoustically, a monophonic broadband noise source (representative of a mono split signal) fed equally to both monitor systems will produce a sound pressure level summation along the median plane approaching 6 dB. The degree to which the summation approaches 6 dB is in fact a measure of symmetry of both the room and the monitor systems.

This is not a violation of the Law of Conservation of Energy; the *mean energy density* in the room will only have increased 3 dB as a result of going from one monitor to two. But along the median plane we are sampling nearly identical direct sound fields and early reflections produced by each monitor speaker, and the result is a *doubling* of sound pressure level by means of simple in-phase vector addition. If the two monitors were fed signals from two *different* broadband noise generators there would be no such build-up along the median plane; instead we would observe the expected 3 dB summation. Stated another way, the two monitor speakers behave as a two-element array exhibiting a sharpening of the lobe along their major axis.

In my own living room, with about 0.5 seconds reverberation time in the mid-band, a pair of well-balanced symmetrical speakers results in a mono summation of 4.5 dB, and this without any particular attention to architectural symmetry.

The ear is fairly forgiving of this precise median plane 6 dB build-up of pressure levels. Whether in the control room or in the home, the head of the listener is constantly moving about, providing a degree of space integration. The ears also respond to the ensemble of reflections in the room and thus sense basically the 3 dB build-up of mean energy density in the room.

Moving on to pan pots, the "3 dB-down" pan pot in common use today is correct for setting stereo center images at their proper energy level in the integrated listening environment. Its only drawback is a "3 dB-bulge" of the phantom center in mono playback, a consequence we have long ago learned to put up with. The "6 dB-down" pan pot, a rare item, was likely not designed for proper stereo balance but rather to preserve proper balances during mono playback of stereo program material.

---

from: **Steven B. Fuller**
Lecturer in Audio Engineering
School of Engineering and Applied Science
Washington University
St. Louis, MO

Regarding the exchange between Messrs. Rettinger and Duncan in the June (1979) issue; Mr. Rettinger is correct but I am certain that Mr. Duncan was referring to sound *pressure* levels adding and merely slipped in referring to power in an article otherwise dealing with pressure. Pressure is the acoustic equivalent of voltage and intensity is the equivalent of power. Recall that

$$I = (SPL)^2/{}^{\rho}0^{c}$$

where ${}^{\rho}0^{c}$ is the acoustic impedance of air and I is intensity of power per unit area.

Mr. Duncan slightly misses the point when talking about pan pots. When centered, pan pots are generally 3 or 6 dB down on each side so that the aural sum in a "6 dB add" control room, for example, would be +3 dB or 0 dB respectively. Of course, one needs to consider the intended purpose of a pan pot. Is it to move the source from one side to the other without changing aural level or is it to accentuate a signal when it is in the center stage by inserting a 3 dB boost? Here it should be noted that the "6 dB add" room will accurately reflect the mono reproduced signal but will not simulate 90% or more of the stereo listening environments for the final product.

The same dilemma arises in recording with the various coincident stereo techniques. Should you engineer your coincident pair for the "6 dB add" which will insure mono balance or for the "3 dB add" which is typical of listening situations? (Yes, implicit in that is the statement that coincident miking does not insure mono compatibility.)

Regarding the exchange between Messrs. Muncy and Kimber; surely Mr. Muncy is aware that the "shunt leg" only exists if he uses the equivalent current source instead of a voltage source. See the excellent letter from Mr. Tiers in the April issue for a proper treatment of the subject. (Mr. Tiers, I am pleased to say, is a product of Washington University.) *Enough said on speaker wire!* (I don't mean to deny the effects of exotic speaker wires but they can't be treated in such a simplistic way.) As to the third wire to sense the load voltage, it is quite practical and easily done without amplifier redesign, but woe to the loudspeaker if the wire should break or otherwise lose contact. Because of the low impedances involved it should not need to be shielded. Further, Mr. Kimber should not speak so lightly of "only DC levels" since most good DC power supplies have a bandwidth greater than 20 kHz. (This is to insure a high "damping factor" at non-zero frequencies.)

---

reply from:
Kent R. Duncan
Founder and President
Kendun Recorders
Burbank, CA

1 — Firstly, Mr. Fuller and Mr. Rettinger are correct that we are measuring SPL, not power add (see Mr. Eargle's letter). The phenomenon does however exist, (and we invite Mr. Rettinger to experience it with us) that 6 dB SPL is easily achievable using our design.

This is easily measured, as described in my article, by adding two monitors individually balanced to produce the same

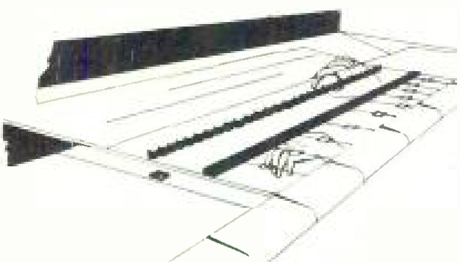
continued overleaf . . .

Kent Duncan
(continued)

SPL with broadband pink noise, together to produce 5-6 dB rise, depending on the geometric symmetry of the room and the electronic symmetry (amplitude and phase) of the monitors.

2 — I feel this is enough hair splitting. We do not propose to participate further in re-definition of definitions in the pages of these letters to the editor by people who wish to debate terms rather than provide music producers with reliable tools.

3 — As far as pan pots are concerned, I'm afraid we're dealing with a philsophical engineering question, rather than a practical one. Unless you are mixing in a situation employing dynamic panning, whether the pan pots are 3, 4.5, or 6 dB down in the center, is unimportant. In most mixes, the pan pots are used for static placement, with the actual levels being adjusted for a proper mix, very often checked for mono compatibility. Again, professional use is an important yardstick for us all.



AUTOMATED MUTING

from: **Michael Tapes**
Sound Workshop Professional Audio Products, Inc.
Hauppauge, NY

I was extremely pleased to read Doug Perry's comments regarding muting during automated mixdown, in the June, 1979 issue of *R-e/p*. Without repeating Doug's observations, it was like I was listening to myself preach the virtues of automation to a prospective purchaser of one of our consoles. Indeed, after one experiments with all that computerized mixdown has to offer, he (or she) is aware that the most important virtue is an independent muting system.

Since Doug's experience was isolated to mixing on an MCI 528 console, he, as well as many other engineers do not realize that not all automation systems allow mutes to be written independently. ARMS (Auto-Recall Mixdown System) Automation offered by Sound Workshop, and MCI's JH-50 system are among the few systems that have a separate MUTE WRITE function on each channel as well as the conventional level write. On systems without MUTE WRITE, a mute cannot be distinguished from a "fader full attenuate." This means that if a section is re-written, both the levels and mutes must be re-done. With MUTE WRITE, levels can be re-written while the processor turns the channel on and off according to previous data; or the muting can be re-done while the processor controls the levels as previously written.

With this type of system, the first pass or few passes can be devoted to all of the track gating necessary for the mix, and from that point on the producer and/or engineer can devote his/her energies to mixing music instead of memorizing when to punch in the horns and punch out the clock (out of frustration). In addition, on the latest MCI and Sound Workshop systems, the mutes are FET or relay actuated, which eliminates the ramp time associated with VCA turn.

I welcome this opportunity to further Doug's remarks and caution potential automation customers to consider their purchase carefully.

from: **Winn Schwartau**
Empiricism
Ossining, NY

Some additional notes on the Multitrack Comparison chart that appeared in the June 1979 issue of *R-e/p:*

1 - Dates may differ from the first prototype, first introduction of machine, and dates of production.

2 - Heads were often supplied by different manufacturers throughout the production life of a machine. If it is an older machine which has had the heads replaced, check that they have been spec'd out and aligned properly. Most heads have variable azimuth, zenith and height.

3 - All larger 2" machines have some form of motion sensing. You may go from any mode to stop without spilling tape. Occasionally, you may go from a fast mode straight to play without having to go through stop.

4 - All 60 cycle motors, may be replaced with 50 cycle motors. Use where 60 cycle power is unavailable. The manufacturer can help you in locating the proper motor. Varispeed for 60 cycle motors can be built by using a 60 cycle reference oscillator which can be varied ±50% or so. The oscillator will drive a power amp, which should be capable of 100 volt or so output. The power amp will then directly drive the capstan motor via external capstan input or equivalent on each machine.

5 - Fifteen ips is almost the standard for most machines, with 30 ips becoming increasingly accepted. Most Ampex and MCI electronics have some interchangeability. (The AG-440 has 160 kHz bias, while the MCI JH-110/114 have 110 kHz bias.) The bias cards and frequencies changed so much throughout the years, that it would be unwise to just swap them without careful examination. Check with the manufacturer for updates, especially concerning relay switching. Great improvements have been made since 1969.

6 - Monitor logic implies that the electronics will enter into "input" when the appropriate channel is placed into "record." Machines differ when they go into fast wind modes or stop. Some switch to "input," others don't.

Early locators, like the Eventide and Selectake I are of the shuttle type when searching. They approach the search number, slow down, and overshoot the numbered location. They will then go in the opposite direction, overshoot, etc., until the fast mode cycle window narrows down to the desired location. Later locators have velocity detectors, and each will "slide" into the desired location postion without overshoot.

from: **Ethan Winer**
Conroy Road
East Norwalk, CT

There is one slight change to the Stereo Synthesizer circuit described in the last issue of *R-e/p*. The capacitor connected between pins 1 and 8 of the NE5534 should be connected between 5 and 8. The circuit will function acceptably as drawn, but the change is a more correct hook-up.

News

AMPEX SELLS CUSTOM FACILITIES TO CBS RECORDS

Ampex Corporation has announced it has reached an agreement to sell the machinery and related inventory of its "custom" duplicating facilities, Elk Grove Village, Illinois, to CBS Records for an undisclosed amount of cash. The sale was effective July 31, 1979.

George Ziadeh, vice-president/general manager of the magnetic tape division, said the sale will permit Ampex to concentrate its resources on more profitable and faster growing segments of the magnetic tape market, such as videocassettes.

The company recently announced it has received a license to manufacture and market VHS* format videocassettes, and expects to begin full scale production at its Alabama tape manufacturing facility in the near future. Ampex currently manufactures and markets Beta** format videocassettes.

In addition, the division has placed on the market a new professional helical video tape and has launched a nationwide promotion and advertising campaign in support of its consumer audio tape product line.

*™ Victor Company of Japan
**™ Sony Corporation

. . . *NEWS continues on page 123* -

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STUDIO UPDATE

Northeast:

☐ **STEVE BRAMBERG,** general manager of **ELECTRIC LADY STUDIOS** in New York, has announced the completion of the four month $1,000,000 renovation project undertaken by **WESTLAKE AUDIO** at Electric Lady, making it the only recording studio on the east coast with the Westlake Four-Way Monitor System. Also, part of the re-do was the installation of a Neve 8078 console in Studio A, also the only east coast studio equipped with one. Studio C, conceived by **JOHN STORYK** and Westlake boasts the Necam Computer, one of two available on the east coast. John Storyk originally designed Electric Lady for **HENDRIX.** "Hendrix wanted Electric Lady to be ahead of its time," says Bramberg, "and I want to continue the tradition." *52 West 8th Street, New York, NY 10011. (212) 677-4700.*

☐ **ALLAN MIRCHIN'S AURA RECORDING** has opened his fourth **SUGARLOAF VIEW** designed studio. The new room, located at Aura's West 52nd Street headquarters, is intended to handle Aura's continually increasing commercial production business. The studio features several new Sugarloaf View concepts specifically designed for high-pressure agency recording, including easy set-up platforms for string and rhythm sections, fully wrap-around control room window, and a separate projection booth. *136 West 52nd Street, New York, NY 10019. (212) JU-2-8105.*

☐ Architect **JOHN STORYK,** of **SUGARLOAF VIEW,** has designed a major addition to the **HOWARD SCHWARTZ** New York City recording studio complex. The two new rooms bring to the Schwartz facility a total of four "state-of-the-art" studios which Schwartz intends to focus towards record label recording dates. The all-MCI equipped facility is highlighted by a fully automated custom designed 56 input console. The $200,000 console is the first of its kind available on the east coast and the costliest ever built by MCI. Occupying some 5,000 square feet on the 19th floor of 420 Lexington Avenue, New York City (at 44th Street), the new Howard Schwartz Studios represent an investment of well over $1 million. *(212) 687-4180.*

☐ **ROSE HILL STUDIOS** announced the appointment of **CLIFF KENT,** formerly of **QUANTUM ELECTRONICS,** as chief engineer. Some of the current projects taking place at Rose Hill are **SPARROW,** recording for **JIMMY IENNER** for **MILLENIUM RECORDS, TAKSIM,** produced by **VINCE TAFT,** and singles have been completed for **THE WHIZ KIDS, THE NEW YORK FLYERS,** and the **TODD HOBIN BAND.** *3929 Seneca Turnpike, Marcellus, New York 13108. (315) 673-1117.*

☐ **QUEEN VILLAGE RECORDING STUDIO** has recently taken delivery of a Neve 44 input 32-track 8088 console. Fitted with Necam, the console is one of the largest available and, according to **WALTER KAHN,** Queen Village president and owner, has provided the flexibility needed by his growing operation. *800 South 4th Street, Philadelphia, PA 19147. (215) 463-2200.*

☐ **KAJEM** has been adding acts, services and facilities this month. In the studio to record was **ARISTA'S BABY GRAND** and **DOUG HENDERSON** for his international release **"JOCKO'S ROCKETSHIP."** As well as offering multitrack recording courses, Kajem now has formed an advertising division titled **ADIO,** headed by **SAM MOSES** and **WAYNE GALLAGHER.** Though Kajem now has a TASCAM 90-16 16-track tape recorder and a Spectra Sonics 1024-24 console, they are still offering 8-track facilities and rates to those who want it. *1400 Mill Creek Road, Gladwyne, PA 19035. (215) GR-3-3277.*

☐ The production team of **ETHAN WINER** and **BOB CARLSON** of **THE RECORDING CENTER, INC.,** recently completed a package of eight radio spots and an audio visual package for **BLUE CROSS OF RHODE ISLAND.** Other recent projects have included an audio visual soundtrack for **THOMPSON CSF** and several radio spots for **STANLEY TOOLS.** *25 Van Zant, East Norwalk, CT 06855. (203) 853-3433.*

☐ **CELEBRATION RECORDING, INC.,** recently completed a digital recording project of a Renaissance Improvisation group called **JAZZantiqua.** Using a Sony PCM 1, the digital master was engineering by **MICHAEL FARROW** for videotape demonstration use at the CES convention in Chicago. *2 West 45th Street, New York, NY 10036. (212) 575-9095.*

have you?
• increased track capacity — gone 24, 16, 8 •
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these are some of the interesting news items that can be announced in the next available issue. Write:
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STUDIO UPDATE


Secret Sound, New York

☐ Using a SMPTE syncronization device, **JERRY LOVE** and **MICHAEL ZAGER** of **LOVE-ZAGER PRODUCTIONS** expanded **SECRET SOUND STUDIOS** 24-track console to 48 tracks to accommodate a remix of "Life's A Party," of **THE MICHAEL ZAGER BAND** on Columbia Records. The original 48-track recording was done at **TRIDENT STUDIOS** in London. Engineering at **SECRET SOUND** were **RICK ROWE** and **DARYL GUSTMACHIO.** *147 West 24th Street, New York, NY 10011. (212) 691-7674.*

☐ **HUGO PERETTI** and **LUIGI CREATORE** have added commercial production facilities to the **H&L MUSIC COMPLEX,** a 24-track studio custom designed by **JOHN STORYK,** of **SUGARLOAF VIEW,** featuring an MCI console, ARP string ensemble, Moog synthesizers and Yamaha piano. *532 Sylvan Avenue, Englewood Cliffs, NJ 07632. (201) 567-8100.*

☐ If you are looking for **RBY RECORDING STUDIOS** and its owner/engineer, **JACK JONES,** try Southbury, Connecticut. After ten years in Roslyn, Long Island, Jack moved to a larger new facility featuring a 32-in Tangent board; MM1000 16-track, MM1200 24-track, JBL 4315 and Century III monitors, Eventide Harmonizer, MXR DDL, Flangers and Phasers. Now working on tracks, RCA recording artists **BENNY DIGGS** and **ARTHUR FREEMAN** with **WILL LEE** on bass. **JOE LOPES** engineering. *RD #1, Main Street, Southbury, CT 06488. (203) 264-3666.*


RBY, Southbury, Connecticut

☐ The re-opening of **STUDIO B** in Boston was celebrated by over 400 people from the Boston music scene. **WAYNE WADHAMS,** who admits to like bringing people together and throwing a party, was the host of this open house, and the studio owner. *419 Boylston Street, Room 209, Boston, MA 02116. (617) COOKING [266-5464].*

Southeast:

☐ **STARTRIP PRODUCTIONS,** an active producer of radio and television commercials, jingles, audio-visual and film soundtracks, has installed a new 16 x 8 Tascam control board, according to owner **BOB BARNES,** with plans to open a four and eight track mixdown and dubbing room within the next 60 days. *2809 Edgewater Drive, Orlando, FL 32804. (305) 422-1549.*


Criteria, Miami, Florida

☐ **MACK EMERMAN,** president of **CRITERIA RECORDING STUDIOS,** in Miami, terms a recent marathon "direct-to-disk" session at the studios "the pinnacle" of his recording career, which spans a quarter of a century. "I felt a certain sense of urgency in capturing the sound intact," says Emerman of the session which involved recording the University of Miami Concert Jazz Band live. Many of the musicians of the band, which won the 1976 Montreux Jazz Festival, were graduating in June. Considered one of the finest jazz groups in the world, the U.M. band is directed by **WHIT SIDENER.** Emerman himself engineered the double album, **"HALCYON DAYS,"** assisted by senior engineer **STEVE KLEIN** and **DENNIS HETZENDORFER,** as well as by **MICHAEL GUERRA.** For parts of the session, the engineers used some new Schoepps transformerless mikes which Emerman purchased in Europe last summer. *1755 N. E. 149th Street, Miami, FL 33181. (305) 947-5611.*

South Central:

☐ The **ACME CARTOON COMPANY** has opened its doors with digital video computer processing allowing for cel animation and computer animation, and a full 24-track audio and video post production facility. **DOUG KAY** and **GEORGE JOBLOVE** are Acme's two system designers. Also on staff as animator is **ROD DAVIS,** who brings to the company a solid background in traditional animation. Contact Diane Barnard. *7141 Envoy Court, Dallas, TX 75247. (214) 688-0303.*

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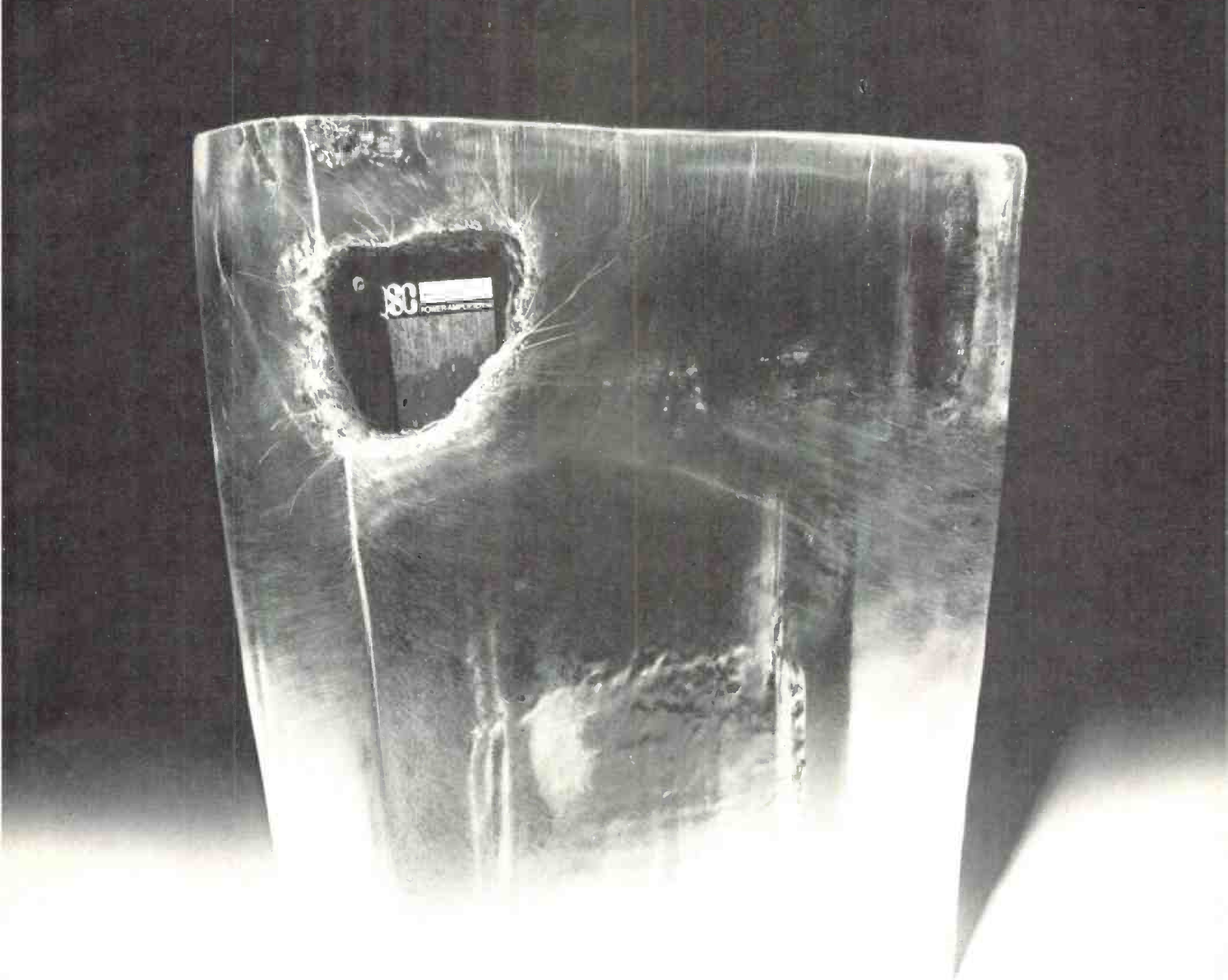
PLAY IT COOL.



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STUDIO UPDATE

☐ **BLOOD, SWEAT AND TEARS** completed laying down tracks at **SUMET-BERNET SOUND STUDIOS** for a forthcoming project for **HITACHI CORPORATION.** Engineering was **CHRIS HUSTON** for producer **JERRY GOLDSTEIN,** of **FAR OUT PRODUCTIONS.** *7027 Twin Hills Avenue, Dallas, TX 75231. (214) 691-0001.*

Midwest:

☐ **KOPPERHEAD PRODUCTIONS** has just completed a major expansion program. In addition to their Tascam 70-H8 with full dbx, they now have a Tascam 90-16 with full dbx, a Tascam Model 15 mixer with 20 inputs, and an AKG BX10-E stereo reverb. Also added are microphones by Neumann, Sennheiser, Sony, Shure and Electro-Voice. Kopperhead also offers a unique synthesizer system made up of an ARP 2600, Minimoog and custom Polymoog. *470 West Mohawk, Malvern, OH 44644. (216) 863-0835.*

☐ **QR RECORDING STUDIOS** has just purchased the new Echoplate II Reverb system and the TEAC Tascam 85-16. **Q. BROWN,** president of QR Studios, says that the addition of plate reverb and the 85-16 will be a positive step for the expansion of QR Studios and QR Productions. **ROSS BROWN,** vice president and general manager, claims that musicians at any level are used to hearing the fine quality that plate reverb delivers, and that they will not settle for less. Chief engineer **JERRY THICHAVA,** who has been with QR since March, added: "We are also installing four JBL 4311 studio monitors, two in each studio." *1307 Ridge Avenue, Evanston, IL 60201. (312) 864-6655.*

☐ **STREETERVILLE STUDIOS, INC.,** located just off Chicago's Magnificent Mile, celebrates its tenth anniversary with an announcement of its $2.75 million dollar expansion. The first room, completed May 15, was dubbed "The Suite" by **JIM DOLAN, JR.** The Suite was designed from the ground up for computer mixing and overdub sweetening by acoustic consultant **GEORGE AUGSPURGER** and architect **JACK EDWARDS.** The Suite features Harrison 4032 console with a Harrison AutoSet computer. 3M supplies the 24-track with an M79 unit, one of three in the complex, while Studer handles all the quarter-inch work. Also installed in The Suite is an extensive array of signal processing equipment, including 26 channels of dbx and Dolby. Second on the list for Phase II is the suite's big brother. Designed by the same two people at **PERCEPTION,** this new 24-track studio is in addition to four other studios at the facility, and offers total privacy. With its own companion rooms (i.e., lounge, kitchen, showers, restrooms) it's designed for block booking or extended projects. The new studio is scheduled to open in July. *161 E. Grand Avenue, Chicago, IL 60611. (312) 644-1666.*

Mountain:

☐ **THE ASPEN STUDIO** will be changing its policy from private to accepting outside projects, according to chief engineer, **RICHIE CICERO.** The studio features a 1,500 square foot facility with a modified Harrison console, 3M 24-track tape machine, EMT 250 digital echo chamber and EMT 140 plate, AKG BX20, Lexicon Prime Time and Cooper Time Cube, limiters including the Orange County Vocal Stresser, two LA3A, three 1176, and two dbx 160s. Outboard equalization includes two Orban parametrics, two Land PEQ-2 and two Pultec 1A3 program equalizers. *P. O. Box 1915, Aspen, CO 81611. (303) 925-8414.*

☐ **COLORADO MUSIC ENTERPRISES** has purchased **MUSIC PLANT STUDIOS,** according to president **DAVID B. SAWYER.** The studio features an MCI multitrack and console, JBL monitors, AKG reverb, and a Kimball 6700 grand piano. Services include record production, copying services, and a musician referral service. **APRIL OSBORN** will be handling the traffic duties for the studio. *4511 East Colfax Avenue, Denver, CO 80220. (303) 3990-4220.*

☐ **SANBORN PRODUCTIONS** has recently upgraded their mobile facilities with the purchase of an Ampex MM1200 24-track and an ATR 102 2-track. The Sanborn remote truck utilizes an acoustically transparent, lightweight design which behaves as a low pass filter eliminating bulky trapping techniques. The control room is equipped with a Sound Workshop 1600 console featuring parametric EQ and VCAs. Automation is planned to accommodate future album projects from various mountain retreats. A newly expanded staff includes **LARRY MARTIN** as studio manager, **SCOTT GETLIN** as marketing director and **CARL FROST** as business manager. *1280 28th Street, Suite 10, Boulder, CO 80303. (303) 443-2372.*

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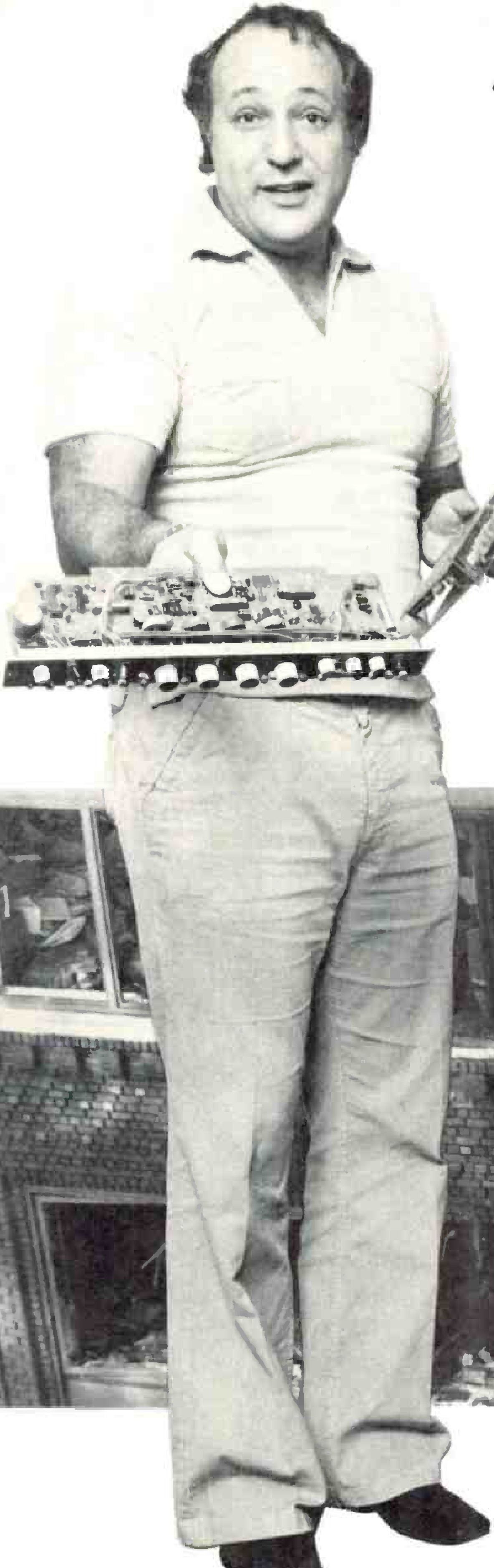

"These Auditronics 501 modules are all that survived the fire...

that totally destroyed our East 46th Street Superdupe and SDC Recording Studio complex last year," says Herb Gordon, studio owner. "Almost before the wreckage cooled and the insurance adjuster completed his work, I was already planning the equipment purchases we would need to get back into business."

"In all the years we had the Auditronics 501 consoles prior to the disaster, I don't remember anything going bad. Our purposes were well served by our old 501s, so I went back to Auditronics the week after the fire to order three more of the latest model 501s."

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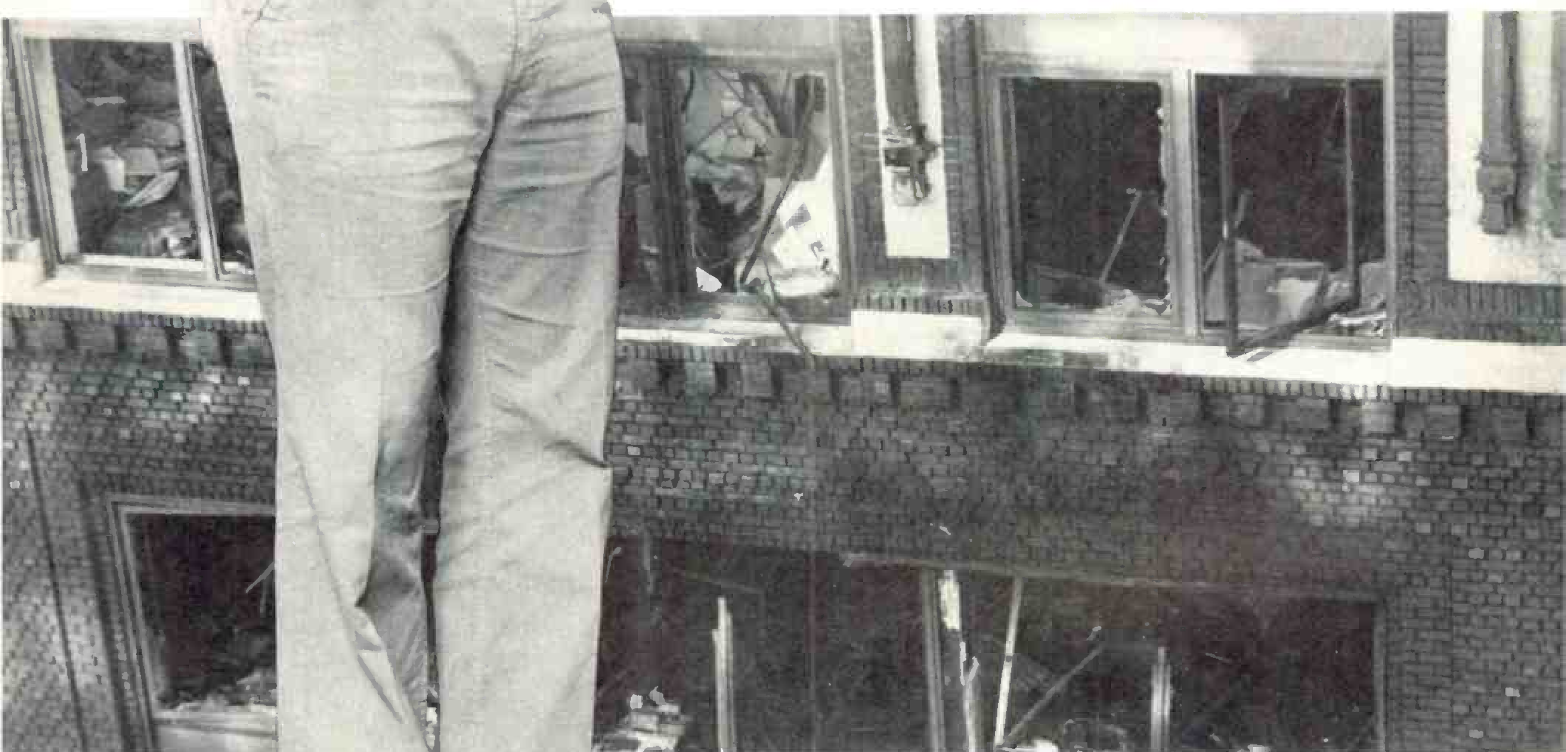

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STUDIO UPDATE

Sea-West, Seattle, Washington

Northwest:

☐ **SEA-WEST STUDIOS** and **BELL & JAMES** were definitely "Livin' It Up (Friday Night)" — at a bash at the studio celebrating the presentation of the Ampex Golden Reel Awards for the "smash" million selling single on **A&M RECORDS.** Pictured are (left to right:) **LE ROY BELL, RICK KEEFER** (owner of Sea-West and engineer of the single) and **CASEY JAMES. "LIVIN' IT UP (FRIDAY NIGHT)"** is the *first* gold single ever recorded in Seattle! Big—Big! Bell & James are presently in the studio working on their second LP for A&M Records. *319 North 85th Street, Seattle, WA 98103. (206) 783-2524.*

☐ **PRODUCERS STUDIO** has opened its new 24-track facility, according to general manager and chief engineer **STEVE DIAMOND.** The room is equipped with an Ampex MM-1200 24-track machine and a Spectra Sonics 26 x 26 console. Recent clients include **MASON WILLIAMS** working on his **BLUEGRASS SYMPHONY** concert tapes. *975 Oak Street, Eugene, OR 97401. (503) 683-1400.*

Southern California:

☐ **DAN KATZ** has been named producer/chief engineer of **THE AUDIO GROUP,** as announced by **SHELDON I. ATFELD,** TAG president. Katz was formerly media coordinator for the **JEWISH FEDERATION COUNCIL OF GREATER LOS ANGELES.** Katz' new responsibilities will include the writing and production of radio spots and also the creation of sound effects for several radio programs. *590 N. Vermont Avenue, Los Angeles, CA (213) 654-6972.*

☐ A part of the Hollywood recording scene for twenty years, **EL DORADO STUDIOS** has recently purchased an MCI 500 series automated console and completed major construction on the entire facility. Studio manager **NADYA BELL** enlisted the services of acoustician **KEN FAUSE** to redesign the control room. Construction was done by **NEW ERA WOODWORKS,** and the new MCI board was installed by **STUDIO MAINTENANCE SERVICES.** *1717 North Vine Street, Hollywood, CA 90028. (213) HO-7-6151.*

El Dorado, Hollywood, California

☐ **UNITED ARTISTS RECORDING STUDIOS** (Hollywood) has brought **HENRY BLAU** on board as chief of design and engineering. A recent arrival to Los Angeles, Blau came to the studio after several years experience as a studio engineer for the **PBS BROADCASTING SYSTEM.** His affiliation was with **WCVE-TV,** in Richmond, Virginia. During past months he has been working as an independent engineer for several local studios.

☐ **AL STEWART** has been in at **DAVLEN SOUND STUDIOS** with engineer **CHRIS BRUNT** assembling material for his new LP. Also working with Brunt is **RONNIE LAWS,** putting together his forthcoming album project. French artist **WILLIAM SHELLER** has completed tracks for his newest with **BRAD HARTMAN** engineering and **LEE HALLYDAY** producer. **JEFF BAXTER** and **AL KOOPER** brought in their group, **FOUR ON THE FLOOR,** as well as **MAURICE WHITE** with the **EMOTIONS.** *4162 Lankershim Boulevard, Universal City, CA 91602. (213) 980-8700.*

☐ **KENDUN RECORDERS** is keeping busy with the likes of the new **JOHN MAYALL** project for **HIBISCUS PRODUCTIONS,** with staff producer/engineer **JOHN STRONACH. JIM SINTETOS** has been occupying mastering with projects for **RSO**'s new act, **MISTRESS. RAY SINGLETON** in to work with **JOHN GOLDEN** on **APOLLO,** to be released by **MOTOWN** at the end of the summer. **GLEN CAMPBELL** working with **JOHN SINTETOS** on his **CAPITOL** release for the fall, with producer **TOM THACKER. FRANK ZAPPA** is in Studio D to mix his new album. **MICHAEL GLOSSOP** shared the engineering dutues with **TOM CUMMINGS** on that project. They were followed in by **KEVIN BEAMISH** and **STEVE WILLIAMS** for the mixes on **REO SPEEDWAGON'S** latest. **JOHN STRONACH** has joined the staff at Kendun after a decade at **RECORD PLANT** as a producer/engineer. **JOHN GOLDEN** and **JIM SINTETOS** have been busy in the mastering room with such projects as **SERGIO MENDES, JOHNNY GUITAR WATSON, IGGY POP, KATY MOFFATT, TOWER OF POWER,** and sharing the mastering task with **DAVID HOLMAN** for the soundtrack of **MEATBALL.** *619 South Glenwood Place, Burbank, CA. (213) 843-8096.*

☐ **DARYL DRAGON** (The Captain of **CAPTAIN & TENILLE**) is launching a new $1.5 million dollar recording complex in August with a nautical theme. Located in the west end of the San Fernando Valley, the studio, built by **RUDI BREUER,** will feature a 52 input computer-controlled Neve Necam mixing console and a below-ground-level isolated drum booth. Complete with waterfall and redwood hot tub, the studio is set in a relaxing atmosphere, with a lounge that resembles the interior of a ship. *Rumbo Recording Studios, 20215 Saticoy Street, Canoga Park, CA. (For information contact Jenny at Moonlight and Magnolias, (213) 342-3193.)*

STUDIO UPDATE continued on page 28 . . .

HUSH MONEY.

The dbx 208 tape noise reduction system is a new product that will impress both your engineering staff and your accountant. The 208 features 8 channels of simultaneous noise reduction on plug-in modules, plus a spare, all in a compact 5¼″ rack mount package.

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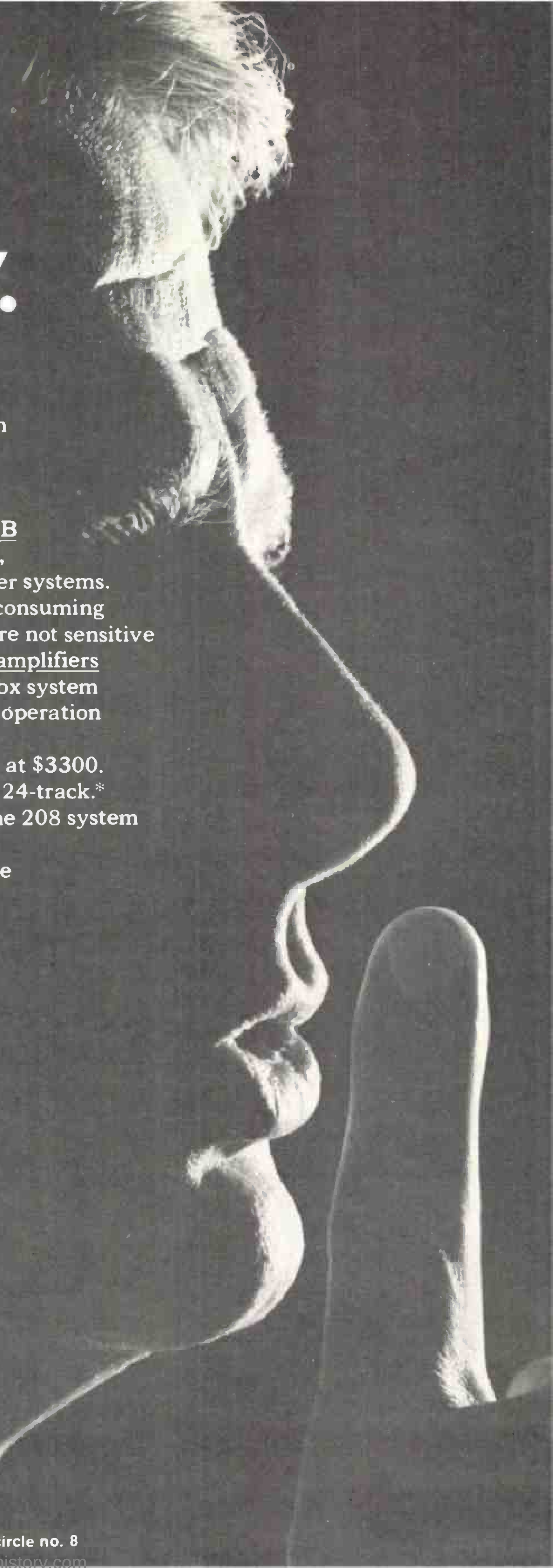

*Nationally advertised value.
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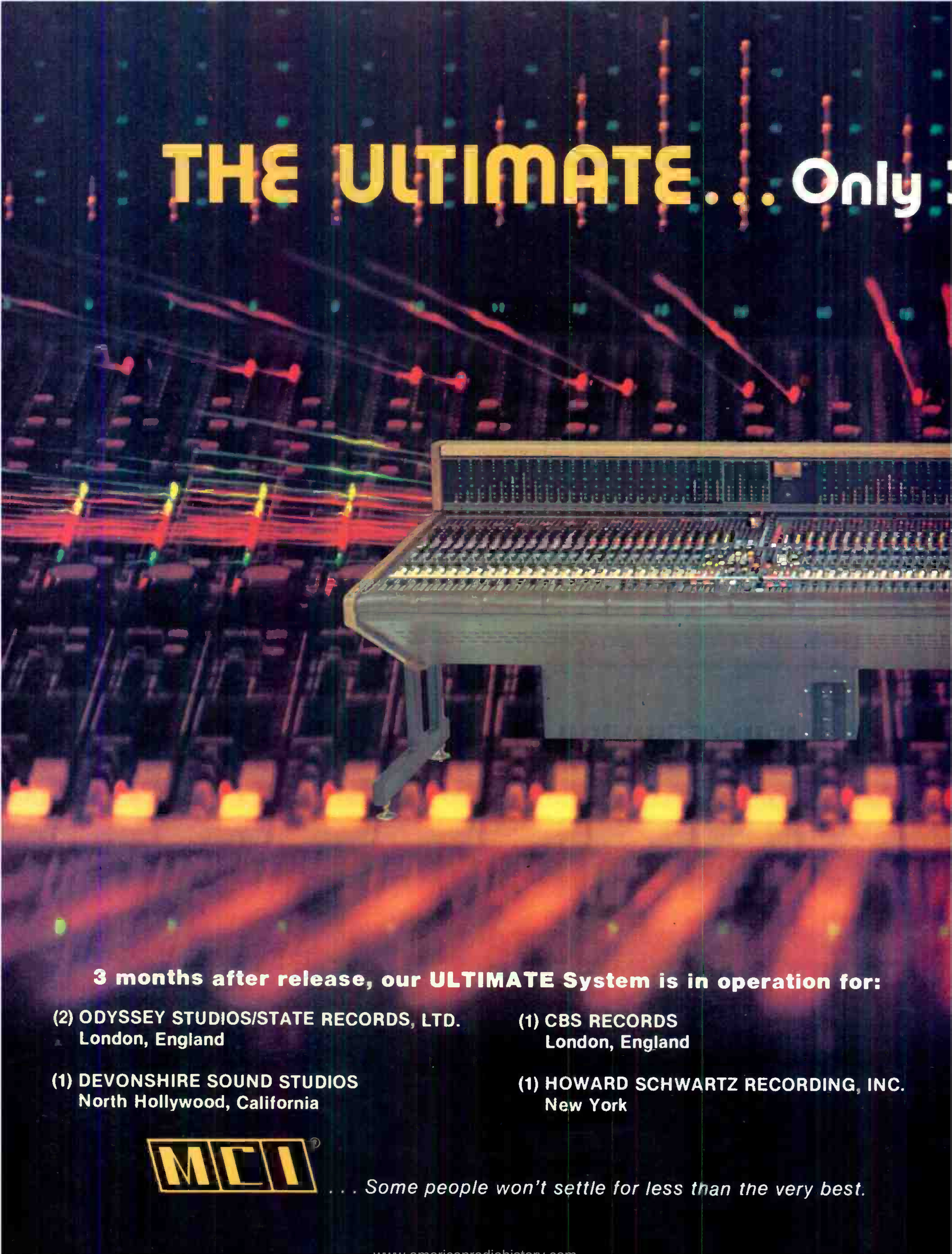
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(1) MIDDLE EAR
North Miami Beach, Florida
(1) ARTISAN SOUND RECORDERS, INC.
Burbank, California

STUDIO UPDATE

☐ **CALIFORNIA RECORDING STUDIO** has announced the opening of its newly remodeled 24-track studio, featuring a new MCI JH-16 24-track tape recorder with the new Autolocator III. Also part of the new gear at California Recording is the Neotek Series III console. *4203 Sunset Boulevard, Hollywood, CA 90027. (213) 666-1244.*

☐ **PRODUCERS SOUND/HOLLYWOOD** is utilizing Sierra's talent to design and build their new seven room facility which will be located at the site of the **HOLLYWOOD RANCH MARKET.** Producers Sound has long been known in Hollywood as one of the leading film scoring studios in the area. This is one of the first facilities that will integrate state-of-the-art film, video and music production.

☐ **SIERRA AUDIO** was commissioned by **VIDTRONICS,** a wholly owned subsidiary of **TECHNICOLOR CORPORATION,** to design a unique audio post production facility in Hollywood. The room was designed to integrate video and audio production requirements while providing living room comfort for clients. Vidtronics has been in the forefront of the industry and an innovator in advancing technology. Vidtronics celebrated the opening of their new studios during July.

☐ **JACK DAUGHERTY,** producer for the **CARPENTERS,** has opened a studio in Glendale, California, boasting a 40-input Sphere Eclipse recording console, as well as an Eventide Flanger and Harmonizer, Lexicon Prime Time Digital Delay, and an AUGSPURGER custom monitor system with JBL components. Daugherty feels the studio design featuring a dome shaped ceiling adds clarity, brightness and flexibility to the studio recording capabilities. The studio opened its doors under the name of **MONTEREY STUDIO** on July 1st. *230 S. Orange Street, Glendale, CA 91204. (213) 240-9046.*

Northern California:

☐ **DUSK RECORDING STUDIO** added new UREI time align monitors and a new MCI 2-track in time to accommodate recording artist **DON OWENS** on a date booked in by producer **MICHAEL THOMAS** and tracking engineer **BOB LANGLIE.** *2217-A The Alameda, Santa Clara, CA 95050. (408) 248-3875.*

☐ **PARISIENNE** new wave band, **SHAKIN STREET,** checked into **FILMWAYS/HEIDER SAN FRANCISCO** Studio A to complete work on their new album, with **SANDY PEARLMAN** producing for CBS International. **GLEN KOLOTKIN** handled the engineering with **JEFFREY NORMAN** doing the assist. *245 Hyde Street, San Francisco, CA 94102. (415) 771-5780.*

☐ **DAVID RUBINSON** working hard at the **AUTOMATT,** producing, engineering and mixing the soundtrack of **APOCALYPSE NOW. SANTANA** sharing the producer and engineering tasks for **SANTANA.** Warner Brothers recording artist **VAN MORRISON** producing and mixing his next album with engineer **MICK GLOSSOP. WEBSTER LEWIS** finishing up an Epic project with **FRED CATERO** mixing and **HERBIE HANCOCK** producing. The Automat has also recently installed automation in Studio A, their largest room, and was also pleased to announce that engineer **JIM GAINES** has joined their staff. *827 Folsom Street, San Francisco, CA 94107. (415) 777-2930.*

for additional information circle no. 10

STUDIO UPDATE

☐ **MYSTIC ISLE RECORDING STUDIO** will be closed all summer to prepare for the production of an LP album with **HOT-L-SACTO.** The album is due to be completed in late August, according to marketing director **JOHN DOYLE,** at which time the studio will resume its normal operations. *P. O. Box 160281, Sacramento, CA 95816.*

Canada:
☐ **SUGARLOAF VIEW** has completed design on **PLUS 4 PRODUCTIONS STUDIO,** located in the Canadian woods outside of Montreal. The MCI 24-track equipped complex will be the first Canadian solar powered studio and will feature an exotic greenhouse and structural design based upon Mayan architecture. The studio and control room are the first Canadian example of the Sugarloaf View room of the '80s, stressing new geometric dimensions in design, new theories of acoustics, and the use of new materials. *R.R. #1, Athelstan, Quebec, Canada. (514) 264-4633.*

Columbia:
☐ **FONOVISION INTERNATIONAL,** located in Bogota, has commissioned **SUGARLOAF VIEW** to design two studios and control rooms as well as support areas for their complex. The principals of the project hope to book not only native artists, but international performers as well. The studio will be equipped with a complete MCI equipment package supplied by **AUDIOTECHNIQUES** of Stamford, Connecticut. *Apartado de Aereo 100920, Bogota, Columbia.*

Nigeria:
☐ **SUGARLOAF VIEW** is currently designing a Nigerian project, called **NATRAL,** which will include two buildings. One building will house studios, a cutting room and various support spaces. The second building will contain a pressing plant. *Lagos, Nigeria.*

to be represented in the next available issue write:
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WHAT IT TAKES TO REALLY HEAR THE TALENT

ATR-100

When you hear recording engineers talk about the "Ampex Sound," they're talking about the sound of transparency. To the limits of technology, Ampex recorders and mastering tape let the production qualities and the sound of the talent come through the recording process intact. Here's a brief review of Ampex professional products for the serious studio...

MM-1200. Lay Down the Ampex Sound

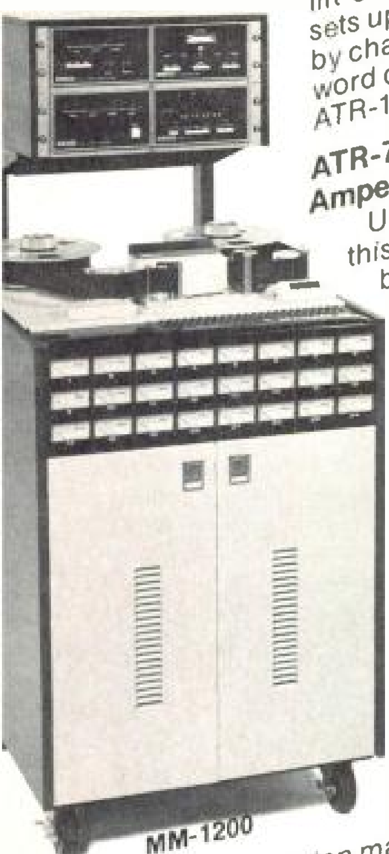
Ampex designed the MM-1200 as a money-making proposition for studio owners. It'll work around the clock as group after group lays down the audio, and then it'll keep right on working as you go back for sweetening, for a vocal, or even for a final mix to a video production (with the addition of a video layback head). Quick-change heads make the MM-1200 even more useful—you can go back and forth between 8, 16 and 24 channels, and switch from one-inch to two-inch tape as easily as a clarinet player changes reeds.

ATR-100. Mix Down the Ampex Sound

This modular one-two-or-four-track machine is the ultimate analog audio tape recorder. In every specification, electrical or mechanical, it **is** the state of the art. Tape handling superiority comes from a unique closed-loop triple servo system that protects your valuable masters. Convenience features begin with the wide-open

Grand M

component placement for easy maintenance, and go all the way to a compact lift-out remote control that sets up all functions, channel by channel. And for the last word on reliability, talk to an ATR-100 user.


MM-1200

ATR-700. Carry Around the Ampex Sound

Unsnap the cover of this reel-to-reel portable, plug in the power, and you're ready to work in monaural or stereo. The ATR-700 has a wider dynamic range than you'll find in most other portables, and it has heavy duty switches and connectors to resist the punishment of normal use over many years of constant service. Switchable equalization makes for fast setup, and full meter monitoring takes the guesswork out of recording and playback. The ATR-700 is a solid moneymaker in the studio, too, when you use it for producing and editing commercials.

Grand Master Audio Tape. The Medium for the Ampex Sound

You'll find a reel of Ampex Grand Master audio tape to fit every one of these professional machines. Every width and every length, packed on reels that fit most professional machines in current use. Best of all, you can use both Grand Master or 406/407 without changing the bias setting on your recorders. Use Grand Master or 406/407 tape for state-of-the-art performance by every measure, from dropouts to edge-to-edge consistency. This is the finest mastering tape you can buy.

Golden Reel Awards. The Reward for the Ampex Sound

A Golden Reel is more than just another award. It's a thousand dollars for the charity of your choice, if you earn it. To qualify for a Golden Reel, you must sell a million singles. Or half a million albums. And you must master your hit on Ampex tape. (Which over 70 top recording artists have done over the past two years.) We're proud of the people who win this Ampex award, and we're even prouder when they give it away to a good cause.

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ATR-700

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the outspoken, irreverent—

Roy Thomas Baker

—JOURNEY—THE CARS—QUEEN—

by
Howard Cummings

Howard Cummings: The first thing I know about you is the Klook's Kleek recording with Ten Year's After in 1968. What about before then?

Roy Thomas Baker: I started in April, 1963 on staff at Decca along with Bill Price and Gus Dudgeon.

Howard Cummings: Then you were at Decca when they installed the first Dolby units in April, 1966.

Roy Thomas Baker: Yes, we were involved with the experimentation. Since there were only 4-tracks at the time, you had to go 4-track to 4-track for multitracking. You needed Dolby in those days with the 4-track, 1-inch Studers, which were experimental as well. Prior to the Studers, we had the Ampex units with the silly gates (head covers) on them.

Going back to the Klook's Kleek thing; we strung cables over the roof and modified the Studer 1-inch. We had a mixing desk called a "S.T.O.R.M." unit, designed by a madman, which looked beautiful. It had these large knobs which looked like chess pieces for the "arch" faders, and the EQ was only treble and bass — a wonderful unit. "S.T.O.R.M." probably stood for something like "Stereo Transistorized Outside Recording Mobile." It wasn't built for rock, but it sounded great for rock. The unit had been designed for the classical mixers, but they didn't like it.

Howard Cummings: What happened after Decca?

Roy Thomas Baker: I went to a small place called Morgan.

Howard Cummings: With Robin Black and Roger Quested?

Roy Thomas Baker: Robin was a "second" then. There was myself, Andy (Johns), and one other. I got the sack (fired) at the place. The place was basically four walls, a lot of padding, everything looking pretty, with dreadful equipment and sound.

Howard Cummings: This must have been around 1969 also because Jethro Tull used Morgan for some of their stuff.

Roy Thomas Baker: It was around the time Mott The Hoople did their first album. Then I moved over to Trident with Ken Scott and Robin Cable. There, I basically handled all the stuff no one else could cope with — like the Dr. John stuff with thousands of people running around the studio, trying to record at the same time. The Stones did the film soundtrack overdubs for *Gimme Shelter*. As a matter of fact, I'm in one film segment.

I also worked with Frank Zappa & The Mothers, which is where I re-met Aynsley Dunbar (drummer with Journey and Starship) and worked on some Santana tracks, which is when I met Neal Schon and Greg Rollie (Journey).

Trident was also one of the first 8-track studios in London. It was basically an Ampex machine, *but it was a 60-cycle machine*, since there was no such thing as a 50-cycle machine! So it would run at 12½ ips until Ampex could supply a 50-cycle motor! Then they went over to the Ampex 16-track which got booted out the back door because it didn't work.

HC: I think they had the 3M also.

RTB: The 3M machine was wonderful. I haven't worked with any of them for ages. I have my own 40-track 2-inch which I take around with me.

HC: Which brand?

RTB: I have a Stephens custom-built and computerized — the only one of its kind. It's wonderful, and unlike other Stephens, it doesn't break down.

The original units have only one sensor, where mine has three.

My auto-locate has 10 memories which is good for mixing when you want to play back only the choruses, verses, or intros, plus a dump memory to remember your positions.

It also has a scan control to go to 60 ips in any mode, which is good for effects.

HC: I hear with the other Stephens they have problems with the transport, even though they might sound good.

RTB: My wow and flutter figures are something stupid like .001; better than MCI.

HC: So this is something you take with you in the States or anywhere?

RTB: Anywhere. It's the only one to record on three continents. It's in two flight cases. Everything is made of aluminum. It's very small and light.

HC: Why did you decide to go with 40 tracks and why with Stephens?

RTB: Stephens is the only one with the 40 tracks and even then I run out of tracks.

HC: What about interlocking two machines, or is that a joke for you?

RTB: It's silly because I can't go from studio to studio with it.

Cummings:
ROY THOMAS BAKER . . . *continued*

HC: What about the signal-to-noise and the cross-talk of the 40-track.

RTB: Signal-to-noise on the Stephens is 4 dB better than the MCI 24-track, per track. The crosstalk is a ¼ dB difference, but I can't remember whether it's up or down. With the 40-track, I've got a better "toy" to work with.

Depending on the time of year, my favorite tape is Scotch 250, but I can't use it during the summer because of static in the studio! Then we use Ampex.

I also work with very high levels. We run 9 dB over Ampex zero, plus we always bash the meters. I can't remember the last time we looked at the meters on the board. If I can hear the distortion, I back off. But I like the sound of tape saturation. And no Dolbys. They're horrible things.

HC: And no dbx?

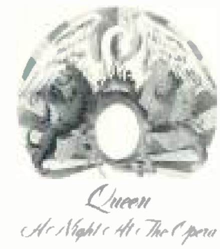
RTB: God no. They're even worse. If I want to use dbx, I might as well use a compressor. (Laughs)

HC: Thirty ips or 15 ips on the 2-inch?

RTB: Thirty ips, 3M 250.

HC: And the two-track mix?

RTB: Fifteen ips on 3M 250. When I make the production master, I use the Agfa. It's very, very clean.



HC: The PEM 468?

RTB: Yes, the blue box for the "cut." Then that goes around the world for all the other cutting and cassettes. When I cut, I usually use the EMT compressor plus ITI parametrics. I compress in the cutting room instead of the mix so I can get my tape saturation sound. That, in itself, is a good sound.

HC: What about other cutting procedures?

RTB: I cut everything in New York with George Marino at Sterling. With him if you ask for more level he'll say, "Yes, you can have lots more level." George will move it up all the way till it distorts, then back it off 1 dB at a time *instead of moving up 1 dB at a time.*

I haven't found anyone as good as George for the level, sound, or quality. I also like his Braun speakers.

HC: When did you start with the Stephens? Cherokee, London, Switzerland?

RTB: San Francisco, Cherokee snd then London. I never took it to Switzerland because we were moving around so much and sometimes we had to split up, which we were afraid of.

HC: What do you mean?

RTB: Sometimes we were in two studios at once. When I did *Queen*, it was a wonderful situation; we were using four studios simultaneously and I was floating from studio to studio.

HC: Within the one complex?

RTB: No, no, no! In London, we would have roadies with fast cars transport the tapes from studio to studio. I was doing guitar over-dubs at Lansdowne, vocal over-dubs at Scorpio, bass over-dubs at Roundhouse. We'd put the bass on, then "OK, goodbye," off in the car . . .

HC: (Laughter)

RTB: . . . to the next studio, etc., til we ended up at Sarm for the mixing.

HC: Great! Why did you decide to go that route?

RTB: We picked studios we liked for various "sounds and effects," but we started running out of time and overlapping. Brian would be at Lansdowne, Freddie at Scorpio, etc.

HC: And this was for?

RTB: *A Night At The Opera.* The basic tracks were done at Rockfield (Wales), as well as the album before that.

HC: There's a thunderstorm effect on one of the Jazz cuts.

RTB: That was live! We were up in the Alps making the album and there was a thunderstorm that knocked out all the power. We walked outside and saw the clouds rolling over the lake so Brian ran in and grabbed his portable cassette recorder. The thunderstorm went right past our faces and hit the studio transformers about three feet away! If you listen to the record, you hear that big "click," which we could have gotten fried from!

HC: I wanted to ask how you put together Bohemian Rhapsody, *very opera-influenced.*

RTB: Aha! It all started at Freddie's house. He was sitting at the piano and he told me he had this tune. "But here, we've got to have an opera section," and I laughed. When we started recording it was very funny with all

AT LAST

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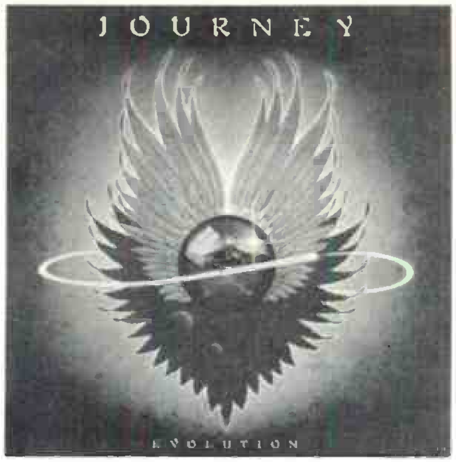
the "Galileo, Galileo's" in it. Even the band started laughing at it all. We enjoyed every aspect of it. It took a long time to do, with all the voices, and by the time we finished, we were on our third roll of 24-track because the tape wore out.

The actual "opera" backing-vocals for *Bohemian Rhapsody* were done at Scorpio, after running from studio to studio. We had a Telefunken machine over there which broke down frequently. As we were working on it, we noticed a crosstalk problem and found that one of the erase heads was broken. We looked at the head and noticed that the outer tracks were larger than the inner tracks, which is a logical idea because of edge problems. But this also made it non-compatible with any other non-Telefunken! So we phoned Telefunken and asked why they did that and they said, "We invented the tape machine. We can design it any way we want!" — a wonderful answer! (Laughs) So we never used the 24-track Telefunken again because of the incompatibility. We kicked it out the door, as we did the MCI gear that broke down. We ended up with a good 3M. By that time we were on our fourth reel of tape.

Once it was finished, we decided, "Wouldn't it be nice if we could release it as a single?" We contacted EMI and told them we had a seven-minute release but they told us "NO, no, no. We couldn't get any airplay



with it." We tried to make some edits in it, but no way would it work.

We're friends with Kenny Everett, an English DJ, and we played it for him and he said he's got to have it. So we accidentally sneaked him a copy of the disc and he decided to play it on the air. Kenny said, "I've got a copy of the new Queen record but I'm not allowed to play it because it hasn't been decided if it's going to be a single. Hang on, my finger slipped," he played it. "Sorry, I wasn't allowed to play that. Hang on, my finger slipped," and he played it again. In the course of that weekend, he played it 14 times! The following Monday, there was a complete uproar with EMI. "Do you know you undermined our promotion department?" "But you said we couldn't get the airplay, and we got the airplay!" So by that time, there were kids running to the record stores asking for it. Then it was released in England. Since it went out in England, it was released in the States, and then gathered even more momentum in England, then also became a hit over here.

HC: And it went on to become the biggest-selling single in the U.K. up to that time.

HC: What do you look for in an engineer?
RTB: Competence, being able to think for himself, a sense of humor, and being able to drink a lot! (Laughs)

HC: What if he comes up with an idea, do you expect him to bring up his viewpoint?
RTB: I encourage it!

HC: Who do you admire or who has influenced you?
RTB: I don't think I've had any influences. At least I don't know of any.

HC: So you're really a loner.
RTB: Yes. I haven't really listened to singles since Motown-Beatles-Stones stuff. If I've copied anyone, it's purely coincidental. Basically I like classical.

I don't even sit at the board anymore. I sit on one of the couches in front of the board. Then I can listen for the performances instead of the sound. Then we get the sound together when we mix. So there's less to worry about technically. I leave that to the engineer. And once I finish a record, I never listen to it again. It's past history.

HC: What appealed to you about Foreigner?
RTB: I've known of them for years.

HC: From Crimson? (King Crimson, 1969)
RTB: Yes, and Spooky Tooth. I was working on some album in England when I got a call from Foreigner's manager asking me to do their first album. Physically, I couldn't do it because of my workload, but I was impressed with them. So he asked, "Who else can get your sound?" And I told them my engineer at that time, Gary Lyons, could. So he did it along with his cousin, John Sinclair. Since I cut back on my workload this year, we've agreed that I'd do their next album.

HC: Keith Olsen did the second album, so I'll ask how the sound will change from the first two albums.
RTB: I don't know. I don't listen to records. I've only heard their singles on the radio. Obviously I'd like to go for a more "open" sound instead of a "closed, up-front" sound, but we're in a bit of a dilemma because they

THE STEPHENS 40-TRACK RECORDERS

Roy Thomas Baker's 40-Track Stephens recorder is the fourth to be delivered since their introduction. The first was delivered to Leon Russell in 1973, the second to Premore in 1975, and another to Leon Russell in that same year. RTB's machine was delivered in January of 1978, by John Stephens, president of Stephens Electronics, who carried it across the Atlantic as baggage to prove its portability.

RTB's machine incorporates a number of improvements which will be described later in this artical.

Transports

All Stephens machines are designed to eliminate the need for capstans and pinch rollers. This is accomplished by two specially designed reel motors which operate in a "tight" servo loop with a precision tape speed sensor phase-locked to a crystal-controlled frequency standard. Since there is no capstan or pressure roller, the complexity of the system is reduced. Additionally wear and tear on the tape is also reduced. The most important part of a multi-track recorder, mechanically speaking, is the deck plate, for if this is not stable, azimuth and tracking errors can result. Therefore, in this design the system uses one inch thick aluminum tooling plate which is milled, then surface ground, seasoned, and then surface ground again. Because of this precision foundation, azimuth adjustments are significantlty reduced; the heads are optically aligned and mounted on a precision headplate. Due to the absence of a capstan and pinch roller as well as the nature of the tape path, tracking error is said to be reduced to a point where it is more than adequate even with the tight spacing of a 40 track format.

Audio Electronics

All components are modular in construction and unplug from each other. Over two years were spent in the development of the 40 track heads to achieve adequate crosstalk, frequency response, and signal-to-noise ratio. The playback

continued overleaf . . .

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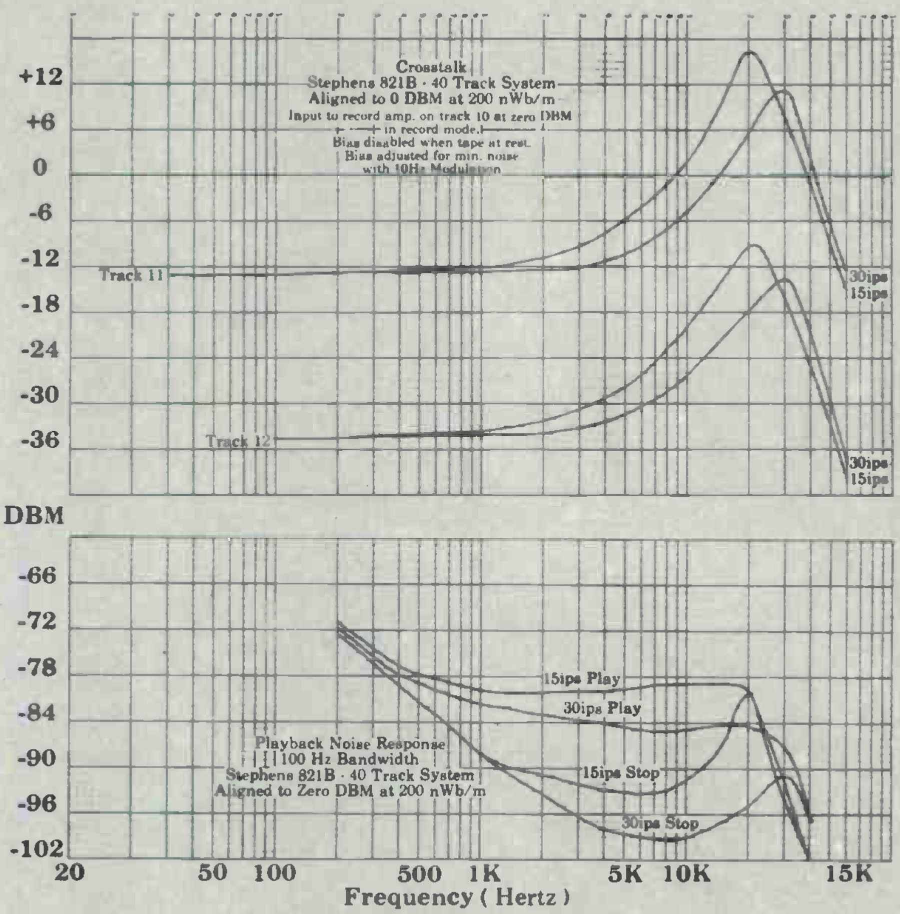
THE STEPHENS 40-TRACK RECORDERS *(continued)*

circuitry is designed so that the playback head is involved in a closed loop with the playback pre-amplifier. The record head is connected in the same way. This results in several major advantages. For instance, in the playback system there is a more complete match in impedance between the head and the pre-amplifier. This improves the transfer of energy and therefore reduces the intrinsic noise of the pre-amp. This feedback loop also reduces the crosstalk that would normally capacitively couple to the circuitry. Greater linearity and phase response are only two of the benefits derived from this type of system. The record circuitry, of course, has been designed with the same objectives in mind. That is, better linearity, lower crosstalk, flatter frequency response. For instance, the



frequencies that are applied to the tape are stated to be flat within 1 dB from 20 Hz to 60 kHz. To further improve quality another objective has been to get the bias frequency as high as possible. Many machines' bias frequencies are as low as 100 kHz. At 30 ips this is a cause of a large residue of noise on the tape. Stephens machines operate at 200 kHz. Another aspect of the system is the amplifiers. The original 600 series amplifiers were designed to look like radio tubes to reduce the reluctance of the industry towards transistors. The 618 and the 620 have therefore been packaged in a nine-pin tube type configuration. Wide bandwidth, low noise with high output and a low output impedance were the major design goals. Maximum peak-to-peak output voltage can be as high as 48 volts. Output impedance is normally around 15 ohms. With the 618 terminated in a 600 ohm load, output is within ±2 dB at +24 dBm to 8 MHz with an open loop gain of 60 dB.

DESIGN IMPROVEMENTS

RTB's was the fourth 40 track machine built by Stephens Electronics and the first 40 track to be built in 2½ years. In that 2½ year period, Stephens transports and audio electronics have gone through some major redesigns. The earlier 40 tracks incorporated the 811D electronics system, which had a rotary switch for each channel to select record, source, and mute assigns. The 811D system was non-remotable. The new system (821B) is multiplexed to reduce the number of

continued overleaf . . .

want to stick with Atlantic Studios in New York since it's been lucky. And I can respect them for that.

HC: You recorded Journey's Infinity *album at Elliott Mazer's His Master's Wheels, in San Francisco.*

RTB: His Master's Wheels was not a studio. It was a converted warehouse. Though we virtually did everything there, it wasn't really a studio. You could feed-back through the window that separated the studio from the control room since it was just a piece of glass. The wires ran from the control room into the studio, so the door was always open. If any trucks went past, you had to stop. One night it was windy and the roof started flapping, so they put bricks on it. Some of the group refused to show up because the place had fleas. It was a horrible situation, but the group wanted to record there since they had recorded their previous album there and liked the sound. The fact that their prior album didn't sell was totally irrelevant. I would have been a lot happier coming down to Cherokee, which we later did.

Wheels had an old Neve desk, one of the saving graces, *but it was an 8-track Neve* and a 16-track Ampex — a monster, so we got a 40-track. I found I could rent it for the same price as a 24-track. It might have even been Leon Russell's machine.

HC: How did you get together with Queen?

RTB: Myself and Robin Cable decided to take a look at the new DeLane Lea studio complex in Wembley and Queen happened to be there recording some demos at the studios. Since they were just opening, they needed someone to record to see how the studios would sound, and Queen agreed in exchange for the tapes. I ended up producing their first four albums up until *Day At The Races* and *News*, then I returned for *Jazz*.

HC: These guys seem to be very influenced by opera — the soaring harmonies, the structuring of the songs.

RTB: Who isn't? I've recently acquired my first record player, and my records outside of those given to me, consist of Verdi, Gilbert & Sullivan, etc.

HC: But how much of that influence comes from you and how much from them when you produce with them?

RTB: The end product is a "corporate" thing.

HC: But the injection of ideas.

RTB: Depends a lot on the song and whose it is. They write the songs and we take it from there.

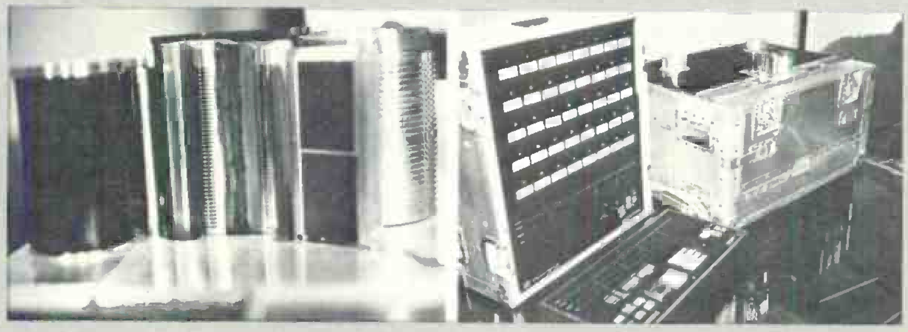
HC: Do they usually come in prepared with material; rehearsed?

THE STEPHENS 40-TRACK RECORDERS *(continued)*

circuits needed for each audio channel as well as reducing the number of wires in the cable to the remote. It incoporates a master rotary switch to select a channel which is then assigned through the use of five pushbuttons. LED readouts display the functions assigned to each channel.

Changes in the servo system were incorporated to improve the tape handling capabilitites as well as to adequately interface to a high response autolocator. To reduce the wow and flutter a more accurate encoding disk was developed as well as an improved encoding disk sensor interface. A third sensor was added for redundant dynamic braking information and direction information for our (then) newly developed Q-II autolocator.

To reduce power dissapation in "play" mode, the power supply was redesigned with a two level voltage source for the reel drive motors (approximately 36 volts for shuttle and 19 volts for play). A current limiter was added to improve torque control during shuttle mode as well as to limit maximum curent from the beefed-up power source with a meter to indicate proper operation of the motor drive system. A second power supply was added for the Q-II computer.



RTB's Specifications

1 - Bias and Erasure: Due to improved tapes newly arriving on the market, RTB requested improvements be made on the erasure and range of bias adjustment. There were problems. The first problem was that the output of the bias supply was inadequate for erasure of tapes such as the (then) new 3M 250 and Agfa 468. Increasing the output generated another problem, what to do with the heat dissipated from the erase stack when all 40 channels are in record mode. A heatsink had to be added to the erase head which then conducted much of the heat to the deck plate. The other problem was the change in alignment procedures which now incorporated the use of 10 kHz instead of 1 kHz. In the earlier system (811D) one could not reduce the bias level to see the low side of the peak when using 10 kHz as a reference for bias alignment. A simple resistor change solved that problem. Most of these changes have been incorporated in all later machines.

2 - Record and Playback Electronics: Since the record electronics have more headroom than most, RTB discovered he could lay on a much higher recording level (+10) with the 40 track than he could on any other machine that he had used. To his dismay, however, he found he could not reduce the playback level adequately during mixdowns. This problem was solved by increasing the resistance of the playback level pot, thereby increasing the range of adjustment.

3 - Portability: RTB's plan included the 40 track being used to record in London and mix in Los Angeles. It was therefore imperative that the 40 track be able to withstand adverse traveling conditions, and potentially rough handling, as well as easy to move about. The complete 40 track in its portable cases weighs about 150 lbs., which is a good selling point for its use as a portable machine. The machine fits into three cases: the deck requires one, and its case includes a spacer to shield the relays and amplifiers from smacking against the side. The sync panel requires another case which has a removable back and front, so that the cables can be plugged into the back while the front is visable. The power supply gets the third case, and operates from inside its case. The heat sinks were put in a "chimney" formation, for dissipate of heat. The cases are made of brushed aluminum, and have large rubber wheels for mobility and handles. The only modification necessary to the deck was adding hold-down springs to the relays to "anchor" them in place.

RTB: Half and half. They're all conceived outside of the studio, then rehearsing normally takes place before the session or in the evening time afterwards.

HC: There seems to be a lot of vocal phasing on a lot of their tracks. Is this a pet thing with you?

RTB: I've always enjoyed phasing. Back in '63, classical records were mostly done in stereo. Machines and tape weren't the best in quality, so we ran two machines as back-ups for the drop-outs. Since we were able to monitor either independently or together, and because of the wow and flutter, it would add phasing. At the time I was always trying to turn-on producers to phasing but none of them wanted to know, especially the Decca staff producers, who were pretty useless anyway. They were more concerned about making sure that their product was liked by their bosses instead of the public. So they'd spend 90% of their time trying to *keep* their job and 10% of their time *doing* their job. None of them wanted to take the "chance." All of a sudden you have the Beatles and Toni Fischer using it, and it becomes "We must have phasing, whoopee! What's all these rockets taking off, etc." And the technical staff was trying to cut back on wow and flutter which *causes* the phasing, the natural phenomenon. So phasing has always been a part of my life. I've never been turned-on by phasing *because it's always been there.* It always existed.

Obviously, now there's easier and easier ways of doing it with all the little boxes but none of them have replaced tape phasing. I love the sound of the Eventide Phaser. It doesn't *sound* like tape phasing, but it has its own unique sound.

HC: For you, is there a difference between phasing and flanging?

RTB: Yes, the sound is different. I don't know about the technicalities, but if you were to play one of each, I could tell you which is which.

HC: When you record with Queen, do you usually record the "basic" rhythm tracks first?

RTB: Yes. If Freddie has written the song, it's been composed around piano. Therefore, piano, drums, bass, rhythm guitar. If Brian wrote it, it's usually been written on guitar, therefore drums, guitar, bass.

HC: And a guide vocal?

RTB: Freddie on his track. Brian would just give semi-guide vocals.

HC: Jazz was recorded in Montreux for the most part. What about the first albums?

RTB: The first was recorded at Trident on "down-time" as an experiment. Since I was an engineer there, I wanted to get into production but the people at Trident didn't

continued on page 44 . . .

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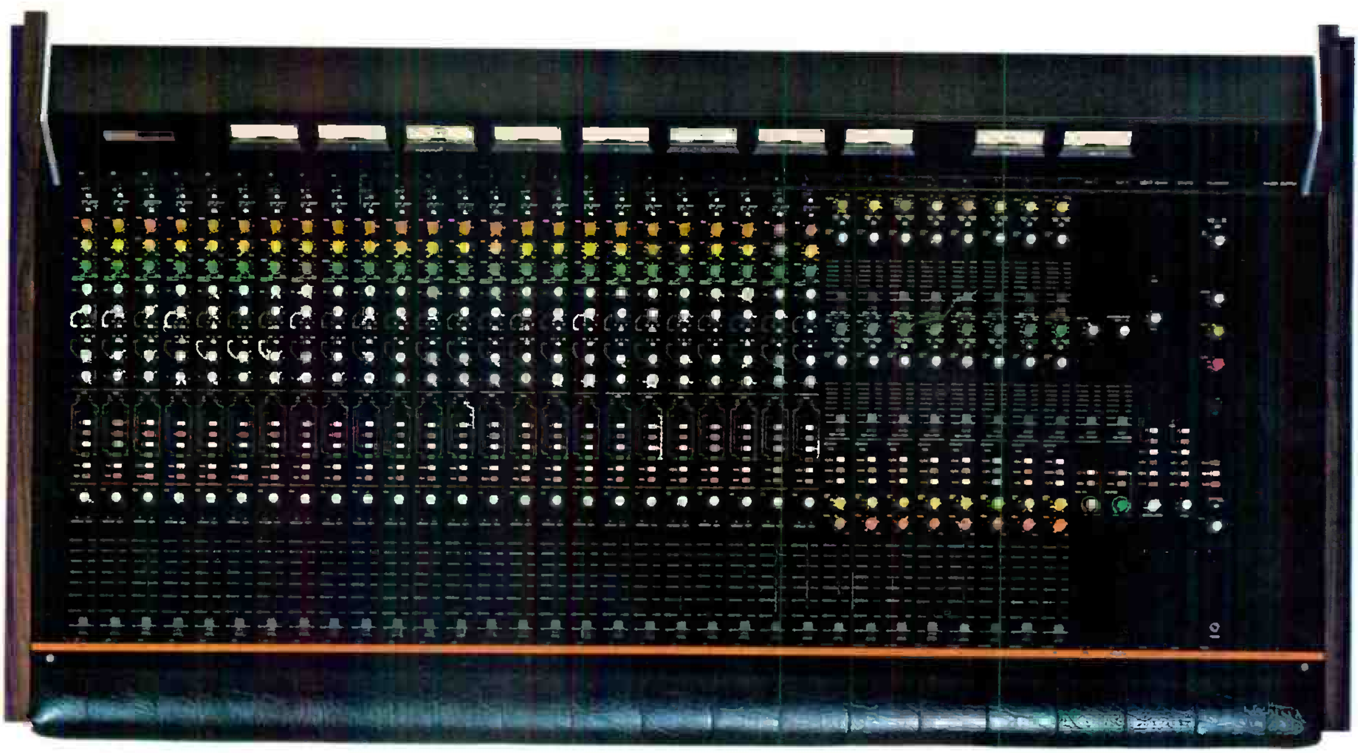

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like that. They complained that I spent $12,000 on the first Queen album.

HC: Well, the first album made the charts and recovered their money, didn't it?
RTB: It recovered it *for them*, but not for us. After our third album was released, the first two went into the charts in many of the territories around the world. But we were still living in sort of slum conditions while the Trident people had the Rolls-Royces.

HC: It sounds like the production contract was prejudiced.
RTB: That's the way it was. Another thing, Ken, Robin, and myself made suggestions for the mixer which they later started selling — and we didn't get anything out of that either.

HC: What do you look for in a studio when you record a group like Cars vs. Queen vs. Journey?
RTB: The same thing; I don't care for MCI equipment or Eastlake-Westlake rooms. Sometimes I fall into a situation where I have to use them.

HC: Like Montreux?
RTB: At least Montreux had Neve, which was good. *(Jazz)*

HC: The Anita Kerr studio?
RTB: That's it, yeah! But we didn't use the studio cause it's too small, so we used the audience area of the Jazz Festival with the nice concrete floor.

HC: What's the capacity? Like the Albert Hall? (3,000-5,000)
RTB: Yes, huge. Nice and big.

HC: You also recorded part of Jazz in Nice. What studio is down there?
RTB: Bear-Le-Alps with MCI and Eastlake. But we brought in Cadac speakers. Anyone who can get used to Cadacs will get on well with them. Everyone over here likes JBL, but I don't know why.

My favorite speakers are Braun. I also have a pair I take with me, which have built-in bi-amps. They're wonderful!

HC: Any other outboard gear? Graphics, filters, phasers?
RTB: No. Anything like that we rent. It depends on the project. If you're working in a studio with good equipment, you don't need those.

HC: Let's go back to what you look for in a studio.
RTB: In the 50's and early 60's, you could listen to a record and say, "That's the 'Phil Spector sound' or that's the 'Motown Sound' or that was recorded at Goldstar or Clover." You didn't have to look at the record. When was the last time you could actually "hear" a studio in a record?

HC: Take the Beatle material for example. 90% of their stuff was done at Abbey Road, but if someone wouldn't have told me that Hey Jude *was done at Trident or* Baby You're A Rich Man *at Olympic, I wouldn't have known.*
RTB: That's the point I'm making. You have to look to see where it was recorded.

HC: These days, it's such a "traveling" situation.
RTB: Yes, and as far as I'm concerned, I like a studio with its own "character" of sound.

HC: Do you like a "live" studio or a "dead" studio?
RTB: Live.

HC: Large or small?
RTB: Large for more ambient response. I feel better in a bigger room because I like the space. I like drums to be in the middle of the studio with the other guys around them. Obviously I don't worry about the leakage.

HC: What about acoustics and set-up?
RTB: I recorded Ronnie Wood's album (*Gimme Some Neck*) in a castle in France . . . parquet flooring, concrete, absolutely no acoustical baffles at all.

HC: Which studio do you use at Cherokee?
RTB: The big one. I like the desk, obviously (Trident), the room itself with the kitchen floor tiles, and the wooden walls for the natural reverberation.

HC: And baffles?
RTB: Never use them.

HC: So you like the "creative leakage?"
RTB: To me, it doesn't matter if you get a little leak-through. In the old days, half of the drum sound was the leakage from everything else.

HC: And the piano?
RTB: Sometimes we have to use a cover because it can get a bit tricky. You have to stick the mikes in the holes, but I don't like that. I like the open top.

HC: What about putting it in another studio and linking the studios?
RTB: No. It's not that important. What is important is the "feel." You lose feel when you do that.

HC: How did you meet the Cars?
RTB: I was asked by Elektra to see the unsigned band in Boston. Since I was going to be cutting the Journey album in New York, I said OK. I went up in a freezing storm and saw the group at a gig along with seven other people in the audience. I liked the songs, the feel, and their approach.

HC: The songs that appeared on their first LP?
RTB: Yes. I had heard snippets of their demo which a lot of companies had "passed" on — they being too "modern." I got on very well with the guys and said, "Let's do it." "When?" "February." "Where?" "London." I virtually committed myself on the spot. I was trying to think of an ideal place for them — AIR-London. I love it — a very "clinical" place, but "nice clinical." It's now a bit dusty and dirty and it sounds a little different too, better than it used to, even though George Martin claims the acoustics haven't been changed. Anyway, we sat there in the middle of winter and did the album in 20 days — the whole lot.

HC: What did you and Geoff Workman (engineer) go through to get the results you got?
RTB: My usual things that I like, which you can hear — big vocals, etc. I don't even remember which mikes, I don't use the same mikes all the time anyway, but I do know they were gray! (Laughs)

HC: What about bass; direct or combination?
RTB: I normally put it on three tracks; direct, top amp, and bass amp. I always have a split-feed. The treble comes out of one stack and bass frequencies out of another. There's a variable cross-over switch that can be used.

HC: Guitars . . . anything raucous?
RTB: The "usual" thing. But never direct for guitars.

HC: "Usual?" Are you saying you have a "standard" recording approach for every group you do?
RTB: To the contrary. The way it sounds is the way it was recorded. It's all ad-lib as we go along. I never have one set thing with the same person.

HC: You were talking about "track-building" with 4 or 5 guitars per track. There seems to be more of that type of recording with Queen than the first Cars LP.
RTB: *Good Times Roll* has about 50 voices in one part. But the thing is, it vanishes after one line. It consists of a four-part harmony quadruple-tracked. Three voices multiplied by four is twelve, multiplied by four again, is 48 voices singing that part. The three guys were singing the same part starting out on one track, then that was triple-tracked, and so-forth for first, second, and third-part harmonies.

HC: What about EQ?
RTB: That changes *all the time.*

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ROY THOMAS BAKER *(continued)*

HC: How do you feel about EQ?
RTB: The end justifies the means.

HC: Do you feel you could have gotten the same sounds for the Cars at Cherokee or Scorpio compared with AIR?
RTB: Not the same "sounds." *Candy-O* sounds different from their first LP. It's much "harder" because Cherokee sounds "harder" than AIR. That's what we wanted and it worked.

HC: Now you say there's a certain "concept" or "flavor" to each album you do. How is Candy-O *different from the first?*
RTB: The concept on the first is the "open" approach: to songs, simplicity, and sound. No stifling or over-production techniques "for the sake of doing it." We left the "holes" open. With *Candy-O*, we "filled" some of the holes.

HC: What about dealing with the vocals of the guys since this was their first album? I know they had been in groups before and had recorded.
RTB: They had recorded before, but none of them were aware of the way I worked. They quickly caught on and everuthing went perfect.

HC: David has a big drum set. Are you of the school that you prefer to use fewer mikes on the set or of using 20 mikes for 20 pieces?
RTB: We mike everything there is! In fact, we use two mikes on some drums — top and bottom — and change the phase discrepancy.

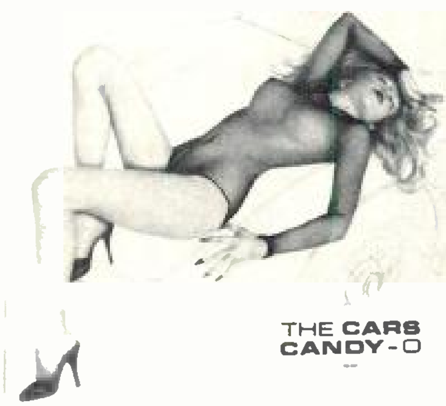
The main point is — you can put as many mikes around the drums as you want. Whether you actually switch them on and use them is a different matter. This would vary a lot on what's being played. Sometimes if a drummer is playing only bass drum, snare, and hi-hat, it might be advantageous to switch the extra mikes off. Then again, it might be advantageous to leave the mikes on for the leakage.

HC: How about mike philosophy? You don't seem to be locked in to any one approach.
RTB: It varies with what I'm doing. I've never gotten locked in to one set of mikes or placements. If you listen to the *Jazz* album, you'll hear on *Mustafa* that we changed the drum sound *within* the same song! The drum sound for the first verse was done on a small kit in the upstairs studio with close-miking. Then we recorded the big bits in the downstairs studio. Then we spliced the backing track and the verses were done upstairs and the choruses were done downstairs. One part is semi-disco and the other is semi-big-and-open. You notice the opening starts with a ticky-ticky then it goes into a big boom-boom. I do things like that which is far from being locked-in.

HC: Ric writes most of the songs. Is there any direction you pointed him in?
RTB: No. The songs were "there." Obviously they had to be "customized." We had to add a few choruses, make some things longer, change the speed, things like that.

HC: What kind of shape were the songs in when they first brought them to you?
RTB: They were in a demo with the whole group playing. All the alterations we did were done in the studio, not in advance. Then we added the choruses, tempo changes, etc.

HC: So you also deal with arrangements?
RTB: You've got to. That's the job of the producer.



HC: Some producers like to bring in an outside arranger and pick from two or three versions.
RTB: I need to be involved in every aspect. Producing, to me, isn't just getting a pretty drum sound. It also has to do with whether the songs are presentable.

Candy-O, because it was more "produced," took 4½ weeks instead of 20 days (laughs). I know people who sit in the studio for three months. If they don't know what they're doing, they shouldn't be in the studio in the first place, unless they're doing something technically extra-brilliant. Most of the time they can't make up their mind. A lot of people say they've been in the studio for three months, which translates to six hours a day, five days a week. I work from 10 a.m. until midnight, seven days a week.

HC: Who decides on "45" releases? Like Good Times Roll *was the third single.*
RTB: We conceived that from day one: that the first three songs on the album would be the singles, but in reverse order; i.e., the first song was the third single, the second song was the second single, the third song was the first single. We did that because most people play the first single, as the first song on the album, but forget about the rest of the album.

HC: You hooked up with an engineer named Geoff Workman.
RTB: He worked at Wessex originally. He joined me in San Francisco to do the Journey album *(Infinity)*, then the Cars first lp, Queen's *(Jazz)*, Woody's, the new Journey, new Cars, and now the new Foreigner. He also can play keyboard instruments.

HC: Since you're working with Geoff, have you gotten out of engineering completely?
RTB: The day I started producing was the day I gave up engineering. I don't like engineering. I liked it when I was a kid, but it became boring.

HC: Because of the "mechanics?"
RTB: I needed a vehicle to get into production. I left Decca because it was a bore. I wanted to become a producer but it was hard *because I was an engineer.* I knew people who designed jackets and wrote liner notes at Decca who became producers, and I was a better producer in my lunch hour than they were. Engineering was the only way I could carve a "niche."

Nowadays, a lot of the engineer-producers only offer their engineering services as a "producer," and to me, producing goes much deeper than that. Producing runs from artist-development onwards. People don't buy records because it has a good drum sound. They like the song, the group, and the "aura" that goes around it. So even though I knew I was a good engineer, I had awards so I must have been good (laughs), it was basically a "vehicle."

I engineered the first Nazareth album, then I produced the second album, which was *not* a hit, partially because the songs weren't there, and partially because I was selling my engineering services as a producer, which is wrong. Producing is a whole different story. I don't see why I should sit there watching meters while I'm trying to get everything else together.

There's a lot of production apects which others don't work at. I delve into artist development — things like working out what we should do on this album or the next album — not cram everything onto the first.

HC: What about the avenue of approach on Journey's songs compared with the Cars?
RTB: Journey didn't know enough about multitrack guitaring and vocals. I normally use four tracks per guitar part, so if it's a three guitar part, I use 12 tracks of guitars.

HC: That leads to a lot of bouncing, even

with the 40-track?
RTB: I have to. I *always* run out of tracks, so I just bounce internally.

HC: What's the concept on the Infinity *album?*
RTB: Basically to keep the "rawness." The guys at Columbia played me their early records and I wasn't too overly impressed. Then they took me to one of their gigs and they were actually *better* onstage than they were on record, which is normally a turnabout for bands. One of the first things we wanted to do was to make sure that the enthusiasm on stage went down on record. Secondly, the songs had to be more commercialized; i.e., no million bar-long, self-indulgent guitar solos, which bore everyone silly. Things had to be more to the point, like nice, big choruses of vocals and big guitars.

HC: How about digital?
RTB: So far, I think the results sound slightly "restrictive" — duller and you can't edit. Obviously it's going to happen. But the worst link in the chain is the disc — the concept of a round piece of plastic being dragged past a needle which scrapes across it, is hardly the most sophisticated method. It worked in its day, but they should go digital across the board to the point you can go out and buy an album the size of a credit card for really good quality and fewer problems.

Everyone is trying to make discs sound better but it is quite a waste of time.

HC: What about this direct-to-disc product, Teldec and Toshiba-EMI releases?
RTB: They're great, but the material is still the big selling point, not the actual quality. It's just another way of making a bad thing better instead of a complete re-think on the way to go. You also hear the gripes about the record companies not using virgin plastics. What it comes down to is the record companies say that only .00001% of the buyers complain. That percentage has to be a lot higher before they'll do anything.

HC: I know you have a Roy Thomas Baker Audio-Visual company. Are you into visuals now or are you planning it?
RTB: I have more video equipment than audio. TV cameras, Bolex film cameras — I've always been into that. I've always played an active part in pushing people towards video, such as the Queen promo films.

HC: Are you into the visual aspects strictly for visuals or for audio-visual?
RTB: Audio in conjunction with visuals, *if it can work*. One of the pitfalls so far is a lack of stereo television, and trying to get the artists to do something different from what

they do on stage. No one has really come up with a concept that does anything different. What they do on stage vs. what they do on video should be different. They have to make that transition.

HC: But isn't a "Queen show" a "Queen show?"
RTB: That's a classic example. When they're on stage, they do their stage show, but when they do their films, it's the four of them standing around the microphones or the concept of "the four faces" in the *Bohemian Rhapsody* promo. That idea came from the cover of the *Queen II* album.

HC: To finish — do you feel there's such a thing as "your" sound?
RTB: I don't really know because I don't listen to my stuff. I didn't think there was but people tell me, "It sounds like you did it — nice and big and loud." □□□

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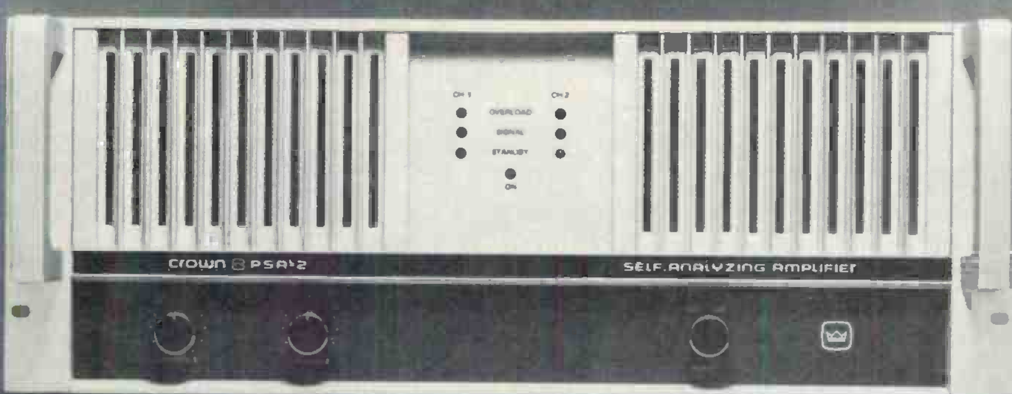

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SPECIAL EMPHASIS:

REVERBERATION

. . . for the comprehensive discussion of this most important subject R-e/p was most fortunate to enlist the views and talents of several outstanding authors, each noted for excellent works in the field. The section begins (on this page) with ***Jim Cunningham's*** *'overview' article, followed by* ***F. Alton Everest's*** *investigation of Comb Filter Effects (page 62).* ***Scott Putnam*** *then further refines the practicalities of building an Acoustic Chamber (page 73).* ***Michael Rettinger*** *comments on several factors affecting the studio's performance (page 82). Control Room Reverberation Time is the subject of* ***Alan Fierstein's*** *article on page 87, followed by* ***David Griesinger's*** *description of the role that Programmable Digital is to play in the future (page 92).*

... an overview

by
JAMES CUNNINGHAM

Artificial reverberation, or echo as it is often called, is one of the most useful effects available to the recording engineer. Recently there has been an explosion of reverb devices available in the marketplace, and to compete for your dollars, they all must claim to be the best if not the ultimate machine of all time. Actually, each has strong and weak points, so the purpose of this article will be to investigate the four basic types and see how they perform.

The four types are: **1. Acoustic, 2, Plate, 3. Spring,** and **4. Digital.** Each of these occasionally has been compared to natural reverb, so a look at some of the natural characteristics should be worthwhile.

Natural

Music is most often performed in a room large enough to accommodate musicians and an audience, so assume our space is 240,000 cubic feet or 100' x 40' x 60'. The recommended reverb time for such a space is 1.6 seconds at 500 Hz. Under average humidity conditions, the reverb time at 10,000 Hz, however, is likely to be no greater than .8 sec.[1] One important aspect of reverberation is the room modes or number of resonances per Hz the room exhibits, as this influences the coloration of the sound. For the reverb decay to be smooth, the room boundaries must be designed so that all resonances decay at the same rate. Although this room exhibits the staggering number of ten thousand resonances per Hz in the critical mid-frequencies, only about three are necessary as far as human hearing is concerned. Another characteristic of this room is the density of reflections. There are many thousands of reflections in the first second, although once again, only about 1,000 are necessary. This is shown in Figure 1, each line representing one reflection from a boundary surface; notice how they decrease in intensity as they increase in number. Figure 1 shows another interesting aspect of natural reverberation; there are very few reflections in the early period — just enough to add some "air" around the

— the author —

Jim Cunningham has been technical director of United Western Studios in Hollywood, California, for the past three years. For six years before that he was with the Sound Market, in Chicago. While in Chicago he did extensive research into the acoustic properties of echo. This research eventually lead him to design the Ecoplate reverberation device which was recently introduced to the recording community.

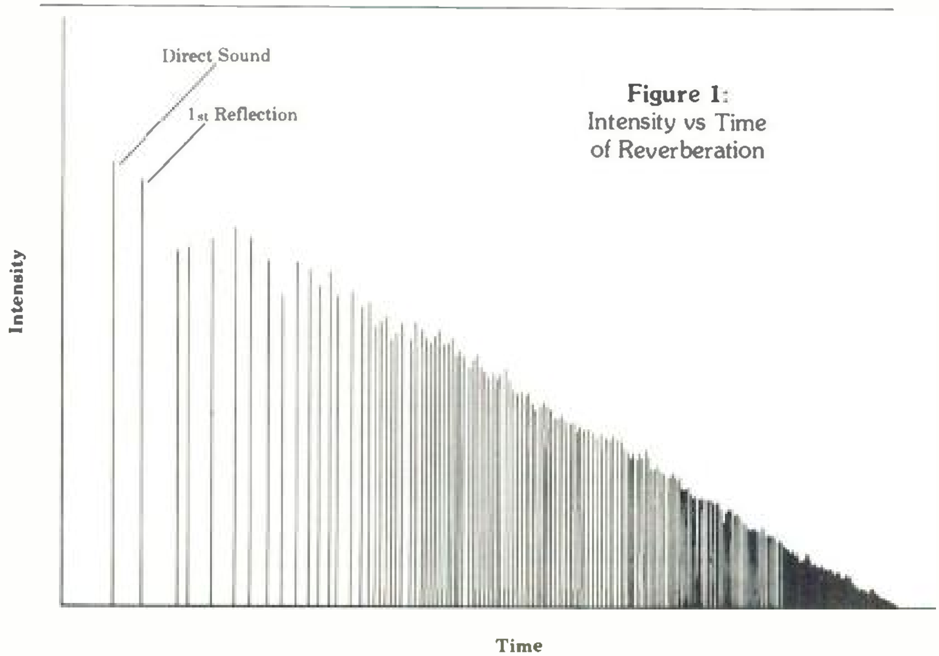

Figure 1: Intensity vs Time of Reverberation

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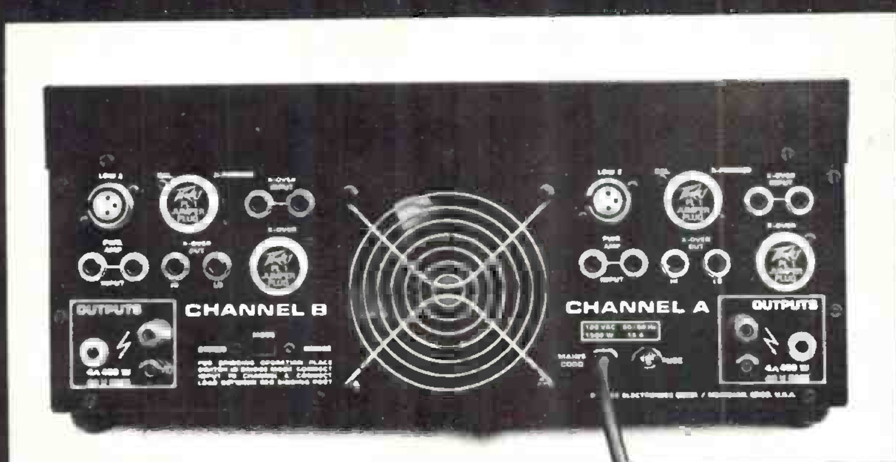

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direct sound but not enough to clutter it. Note that the reflections are random with respect to time and intensity and the time difference between reflections is usually not more than 20 milliseconds. One more characteristic: each reflection may bounce around as many as 50 times before becoming inaudible, traveling an average distance of 38.7'. In doing this the higher frequencies will suffer more loss than the lows which accounts for the shorter reverberation time at high frequencies.

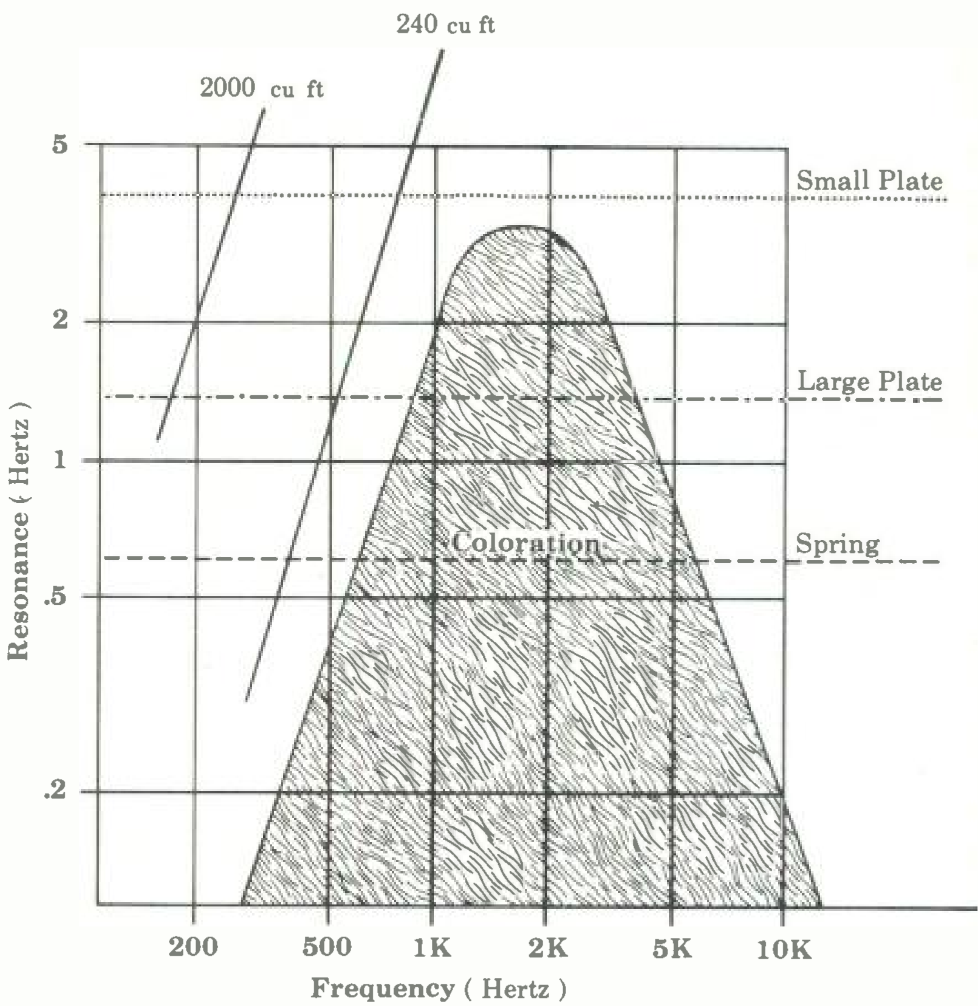
Before anyone gets the impression that all this is a recommendation for live "in concert" recording — or giant echo chambers — it is not. But perhaps the preceeding will serve as a base of reality from which we can assess the various artificial reverberation systems.

The Acoustic Chamber

The acoustic chamber is the oldest type of artificial reverb; ideally, it is a room of at least 2,000 cubic feet whose walls are non-parallel to avoid flutter echoes and "hard" or reflective enough to have an absorption characteristic of .03 Sabines, easily obtainable with a few coats of verathane on plaster. the ratio of length, width, and height should not be integral multiples or the frequency response of the room will be unnecessarily irregular. All this should give a reverb time of about 3.5 seconds at 500 Hz. With an average temperature and humidity condition, the maximum reverb time at 10 kHz will be 1.25 seconds. For slightly longer times at 10 kHz, water vapor could be inserted. The placement of transducers in an acoustic chamber is largely a matter of individual preference, although starting with the loudspeaker(s) in a corner and the microphones somewhere off center is a good place to begin. An interesting variation of it is to use a Dipole speaker somewhat off center with two PRP™* microphones located in the dead zones of the speaker. A dipolar reproducer can be built by mounting most any speaker on a baffle board without enclosure so both sides of the speaker are open to the chamber. At a point 90° off axis from the cone the sound waves generated at the front and rear surfaces will meet. Because these waves are approximately equal in amplitude but opposite in phase they will tend to acoustically cancel. At the places where this interaction occurs there

*PRP™ (Pressure Recording Process) synonymous with PZM™ (Pressure Zone Microphone).

Figure 2: Audible Coloration of Reverberation *vs.* Frequency *vs.* Volume



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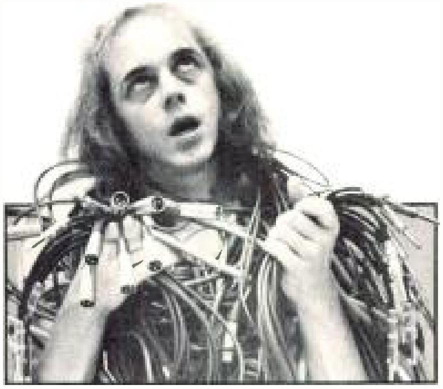

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. . . It means you can send all the guitars and keyboards from the stage to the mixer on one line of your existing microphone snake. MAINLINE eliminates costly balancing transformers and tons of additional cable. You save money, time, and hassle. It's simpler, more reliable, and most important, the music sounds much better!

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will be little direct radiated sound. This allows only indirect sound to reach the microphone and according to PRP principles, no coloration will be added due to comb effects of early reflections. *(Comb effects are discussed in F. Alton Everest's article on page 60.)* Speaking of coloration, Figure 2 shows the zone of coloration of various reverb devices. Note the 2,000 cubic foot minimum of acoustic chambers. This fact, together with the extreme difficulty in isolating the chamber adequately, has influenced many studios to use one or more of the other reverb devices. *(Chamber construction is discussed by Scott Putnam beginning on page 71).*

The Plate

Probably the most popular type of artificial reverberation device currently being used to make "hit" records is the steel plate. Most likely this is due to some unique characteristics these devices have. One of these is the long reverberation times at very high frequencies, over 2 seconds at 10 kHz, another is the frequency response as shown in Figure 3. These two qualities contribute to the crisp, bright decay for which plates are famous. As has been shown, natural reverb cannot duplicate this due mostly to molecular absorption of the high frequencies by the air. Obviously, what is desired here is an effect; a "brighter" sound without having to turn up the equalizer. In other words, if the reverb that is added to the instrument has lots of high frequencies, it seems to give the impression that the direct sound is brighter.

One aspect of a steel plate that has been considered a drawback is the increase in reverb time at low frequencies. Although this occurs in natural reverb, it can be extreme in plates, which operate in the "bending mode." Although it can't be seen, the sound ripples down the plate at a speed much less than that in air. This speed is directly proportional to the square root of the frequency, therefore low frequencies move much more slowly than high frequencies and so will take longer to decay. When these devices first appeared, many engineers rolled off the low frequencies being sent to the plate. While this helps, it also reduces the "warmth" of the reverberant sound. A better way is to increase tension on the plate and select damping material which has exactly the right flow resistance to produce a flat reverberation characteristic.

The Spring

The spring reverb has the widest range of cost and quality of any artificial device. The simplest spring reverb has a driver-pickup coil mounted at one end of a short coil spring. Torsional vibration of the spring produces reverberation, but there is a finite delay as the sound travels up and down the spring, so the early part of the reverb is a series of short evently spaced reflections. Excited by a short percussive type sound, an audible flutter echo is heard. A number of ingenious solutions have been found to this problem such as multiple springs, deforming, etching, etc., to produce a high degree of diffusion. A well-designed spring reverberator, although not inexpensive, can sound more like natural reverb than any

Figure 3A: Warble Tone Frequency Response of LIVE CHAMBER

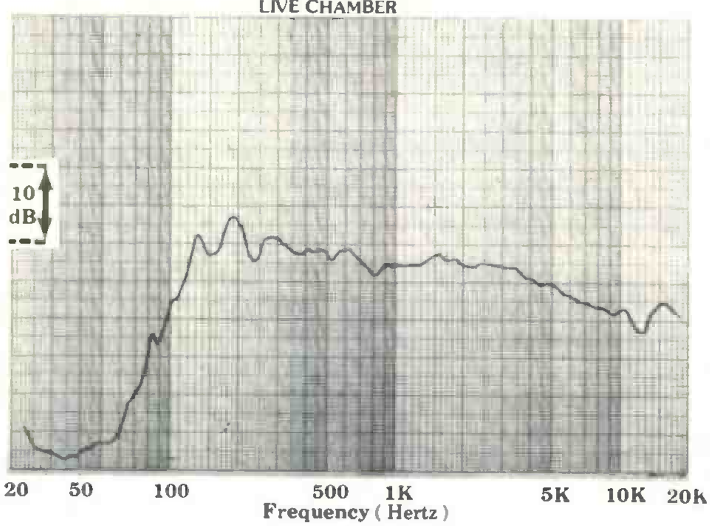


Figure 3B: Warble Tone Frequency Response of PLATE REVERB

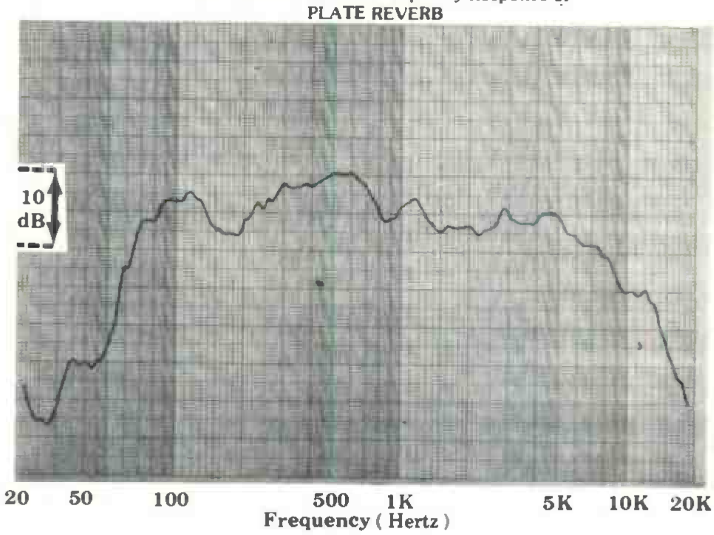


Figure 3C: Warble Tone Frequency Response of SPRING REVERB

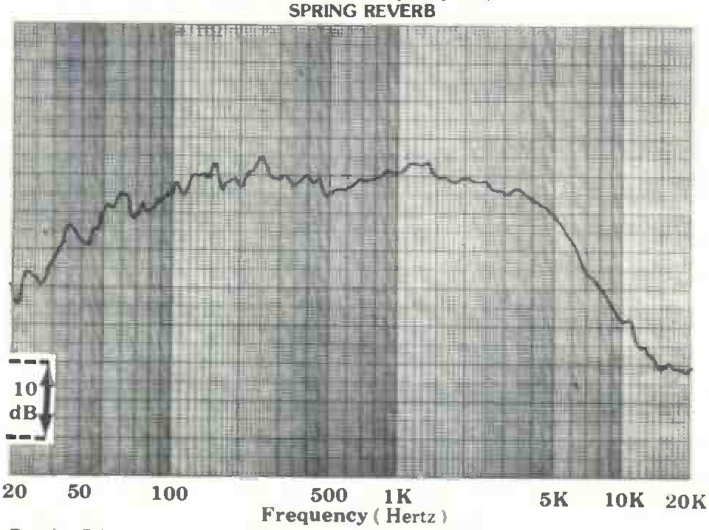
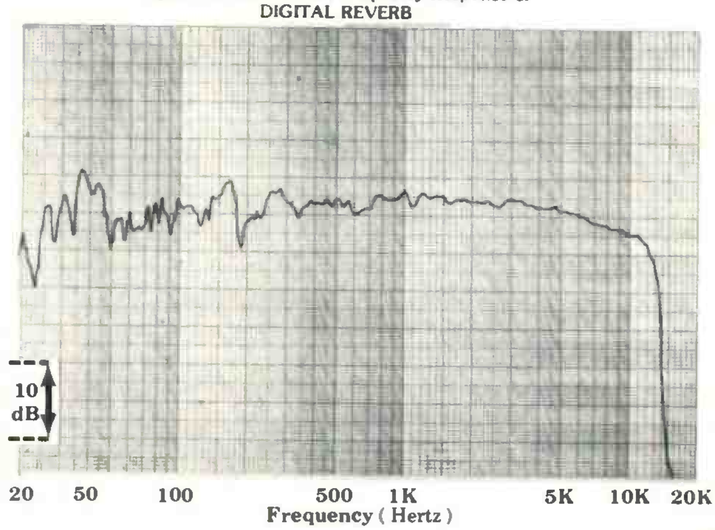


Figure 3D: Warble Tone Frequency Response of DIGITAL REVERB



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other artificial device, except for the moderately high degree of coloration inherent in these devices. (Figure 2)

Digital Reverb

The fourth type of artificial reverb, digital, also has a wide range of cost and quality. They vary from devices with interesting and useful effects but marginal reverb, to all the bells and whistles included in a very high quality reverberation devices. To produce the latter requires an extremely high speed computer as part of the circuitry, not exactly an off-the-shelf processor or inexpensive item.

The buyer of any reverb unit should listen to it with as many different types of music as possible, preferably some short brass passages so that the decay characteristics can be heard in the open. Sustained music like strings will show if there is coloration which will cause "clutter," and some short percussive sounds will expose periodicity.

For the reader who would like to go into more detail on these and other characteristics of artificial reverb devices, the remainder of this article will be devoted to how each of the four types of reverb performs with regard to the various characteristics. Although natural reverb will be factored into this discussion it will become increasingly obvious that reality is not necessarily an achievable or even, at this time, a desirable goal for most popular music.

Perception Of Reverberation

Purists may tend to scoff at this, but the truth is that the present state of the recording art simply cannot recreate music in its existing natural acoustic field, so it is left to that artist, the recording engineer, using close mike techniques and adding reverb, to recreate it as best he can. If he tries to capture natural acoustics in a recording he often instead gets that dreaded "off mike" sound, caused by the comb filter effects of direct and early reflected sound; yet this problem does not occur with two good ears at the same location as the mike (not to be confused with the loss of intelligibility which occurs when the ears are presented with a low ratio of direct to reflected sound). Next time you go to a live concert notice how you are not aware of any reverberation — yet if you listen to a recording of that music made at that same concert, you will be very conscious of it. Neither stereo nor quadraphonic has brought us the solution to this problem. Recent research in vision has brought to light that the brain has a processing channel strictly for depth perception. Because of two slightly different images seen by each eye, the stereoptician can recreate visual depth, but the closest we can come for audio to simulating a depth perception channel in the brain is "binaural sound" or earphone reproduction of a dummy head recording. Some experimental work is being done in "holographic stereo" by cross feeding a specially processed signal to cancel information from the right speaker which is diffracted into the left ear and vice-versa. Until the scientific community gets away from its present preoccupation with "direction sense" hearing theory and grapples with depth perception, it will remain for recording engineers and their producers to stimulate that depth perception channel with any means at their disposal.

Comparison

Table I shows how the four artificial reverb devices perform with regard to a dozen important characteristics common to all of them. The first, high frequency decay is important because, as explained previously, a long (2 seconds or more) high frequency decay can be a useful effect in pop music. The acoustic chamber and spring reverb are about the same as natural reverb. Most of the digital units are variable, a very useful feature if the control allows the 8 to 10 kHz range to be varied without overcompensating the 4 to 5 kHz range. Most plates are outstanding in regard to high frequency decay time, but the thickness is a critical factor and unless special methods are used to obtain a high thermal conductivity, a small, thin plate will have its high frequencies damped by atmospheric pressure more than a thicker plate.

Frequency response is a good way to predict the usefulness of a reverb unit for your application. Figure 3 shows the warble tone response of each of the four types. The plate has the widest frequency response and the digital has the flattest. Although the acoustic and the spring appear to be inferior, this type of curve may be preferable for large orchestra recording. Of course, equalization may be used to alter, to a limited extent, these curves.

Dynamic range is the difference, expressed in decibels between the overload point and the noise floor. The overload point usually begins in the higher frequencies and is mostly a problem associated with electronic or mechanical reverb systems. Plates and springs usually employ some high frequency pre-emphasis thus reducing the power handling of the driver amp at these frequencies. Digital involves a similar design trade-off so if you're big on lots of echo on castinets, you might hear some distortion.

Most digital reverb devices offer the widest selection of decay times, from more than a minute to less than half a second, while plates can be varied from one to five seconds. Springs are not normally variable, although some of the higher cost units have a 2 to 4.5 second variation. Most of the variation in decay time with the last two types will be more in the middle and low frequencies rather than the high frequencies. Acoustic chambers are not generally variable.

All rooms exhibit coloration or resonances due to cancellation or addition as a periodic signal and its reflections meet in space. These are called room modes and the number of them that occur per Hertz are given by the expression

$$(4\ V/C^3)f^2.$$

V equals the volume, C the speed of sound and f the frequency. An interesting exercise will be to see how this works out for the previous "natural" and artificial reverb examples. For the 240,000 cubic foot space at 100 Hz there are about 20 resonances per Hz and at 1,000 Hz there are about 2,000. The 2,000 cubic foot chamber has .16 resonances per Hz at 100 Hz and 16 at 1,000 Hz. The human hearing system, strange as it may seem, cannot detect any coloration in either room.[2]

The shaded area under the curve in Figure 2 shows the frequencies where coloration will be audible. For example, a 240 cubic foot acoustic chamber would exhibit some coloration. In plates the

Table 1: Twelve Point Comparison of Artificial Reverberation Systems

	Hi-Freq Decay	Freq. Response	Dynamic Range	Variable Decay	Coloration	Decay Curve	Early Sound	Pre-Delay	Echo Density	Periodicity	Decay Profile	Stereo Spread
Acoustic	Short	Rolled Off	Fair to Good	No	None	Smooth	None	None	High	None	Low Freq.	Poor to Good
Plate	Long	Wide	Good	Good	Some	Smooth	None	None	High	None	Mid Freq.	Good
Spring	Short	Rolled Off	Fair to Good	Fair	Much	Rough to Smooth	Some	Some	Medium to High	Much to Little	Low Freq.	Good
Digital	Variable	Cut Off to Wide	Fair to Good	Best	Much to None	Rough to Smooth	None	Variable	Medium to High	Much to None	Variable	Poor to Good

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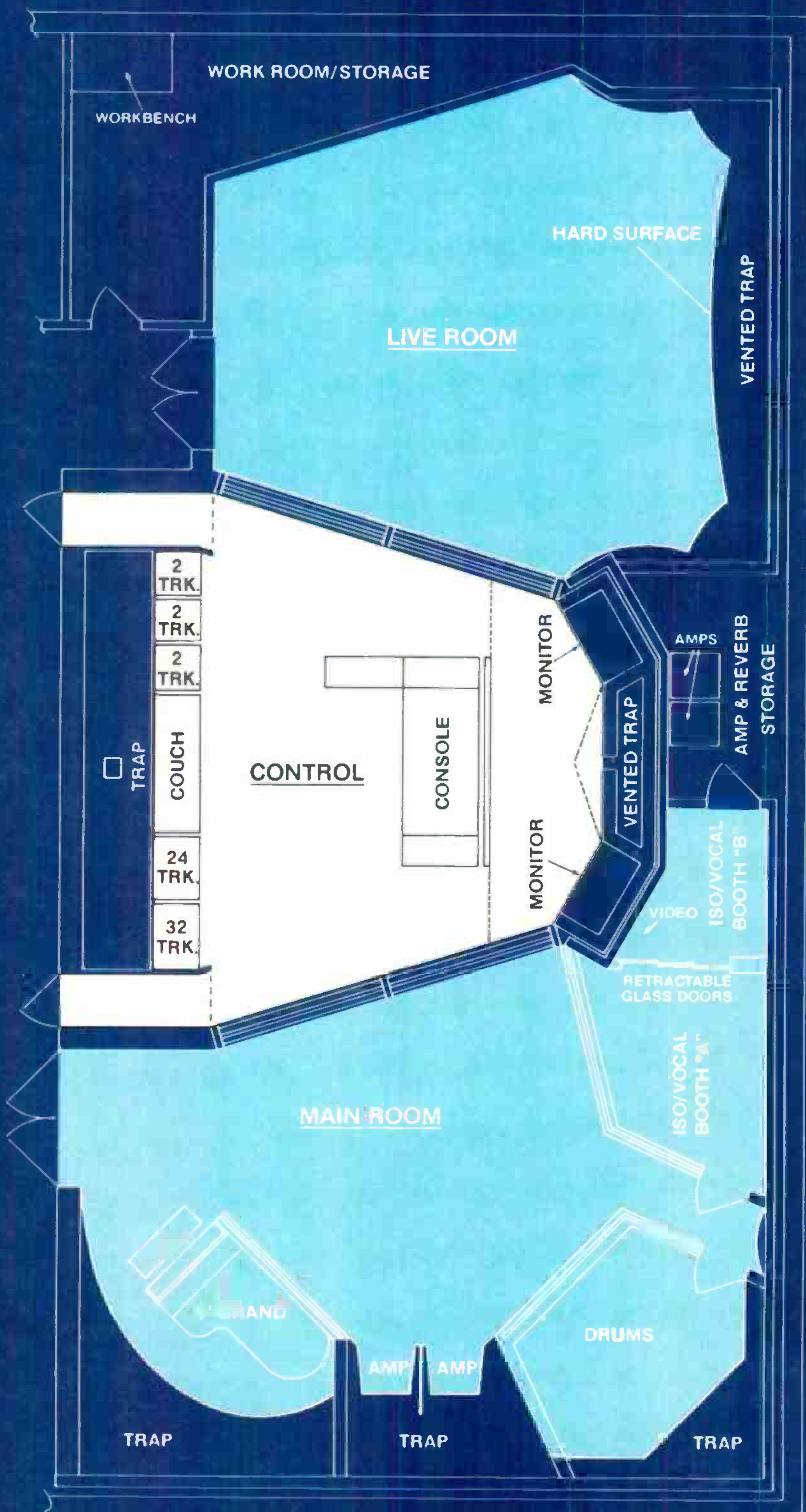

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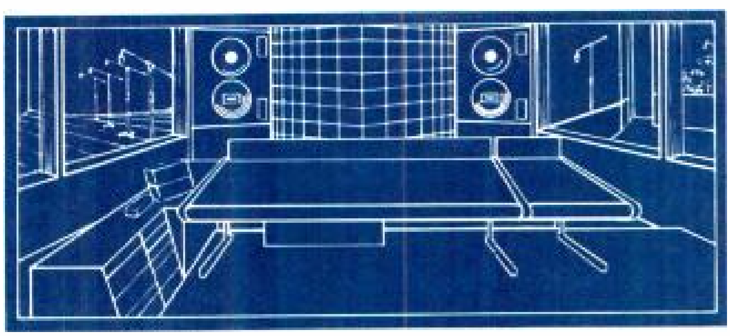

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number of resonances per Hz is equal to A/d where A is the area and d is the thickness of the plate. Figure 2 also shows how the coloration can be reduced in a plate if the thickness is reduced very much more than the area. Coloration in a spring is a function of its length or delay time. This is a complex function of the wire and the number of turns but in a practical case a 47" spring will have a delay time of 300 milliseconds and .6 resonances per Hz, clearly in the audible range. It should be pointed out at this time that most of the characteristics listed in Table I are quantity rather than quality judgements. For instance, for certain kinds of music an engineer might prefer some coloration in his reverberant signal because that gives the kind of effect he is looking for.

A smooth decay curve is one in which the reflected energy dies away evenly at all frequencies. Few artifical systems will do this, however, and if you were to look at the curve on a storage scope you would see dips, plateaus, and peaks in the curve. Acoustic chambers constructed as suggested earlier, will be very smooth. Decay smoothness in plates is mostly a matter of even tension on all edges, which is user-adjustable. Decay smoothness in springs and digital units is a function of the design so it's best to listen to them before you buy.

Evaluation

A good way to evaluate most of the characteristics listed in Table I is to use a small music synthesizer. Set the envelope generator for fast attack and medium decay and use the sine output of the oscillator. Feed it into the reverb device and listen to reverb only at the output. Also try it with the envelope generator set at the shortest decay time.

At the present time, no artificial reverb system can duplicate the low density of early reflections, which is a characteristic of natural reverb and is illustrated in Figure 1. This period supports the sound structure or as some would say, "fattens" it. It is the period during which the ear detemines room size and in stereo gets an "auditory perspective" impression. A little math will illustrate this. Echo density is equal to $(4\ C^3/V)t^2$ and the time period for the early sound is

$$5 \times 10^{-5} V/\Delta t$$

where V is M^3 and Δt is the exciting pulse width.

If we use a 1 millisecond pulse width our 240,000 cubic foot room has a 128 millisecond early period and the 2,000 cubic foot acoustic chamber has a 12 millisecond early period. At the end of both early periods the echo density is about 1,000 reflections per second. This is, of course, why most people use a pre-delay of 100 milliseconds or so in the send to a chamber: to avoid the masking effect of high density reflections on the direct sound. This long delay, however, has the unfortunate effect of making the onset of the reverb sound like a slap echo. If any sound is followed, at the same level, by a repetition of itself the ear will detect two sounds if the interval between the two is long enough. On the other hand, if it is short enough, the two sounds will fuse and be heard as one continuous sound. The same is true of each succeeding repetition. Thus, the sound seems to gain "body" and "fullness" as more repetitions are added until we finally perceive the effect of reverberation. This is what happens during the early period of natural reverb. Spring reverbs can come close to replicating the natural early sound but because of frequency dependent delay effect, they cannot duplicate it. Experimental devices which can duplicate it, as well as proper mike techniques in large studios, have confirmed the need to "fill in" the pre-delay period mentioned above.

In regard to echo density, it has been determined[3] that an echo density of 1,000 per second, *providing that the spacing of the reflections are random*, is sufficient to simulate flutter-free reverberation. In other words, a 1 millisecond delay line with feedback can produce 1,000 evenly spaced repetitions per second but will not produce convincing reverberation — an interesting tunnel-like effect perhaps — but not reverb. On the other hand, if you had 1,000 delay lines, each one greater than the next but incommensurate, you could get one second of good, noisy, reverb.

Most digital reverb units employ delay lines with feedback as part of their circuitry and thus fall somewhere between the extremes mentioned above. The lower cost units will exhibit some periodicity or flutter echo because not enough processing time is available to insure density and randomness yet still perform all the other tasks. The range of effects these units offer, however, should make them worth their cost.

If any reverberation device is excited with a half-second burst of pink noise the "decay profile" will be revealed, that is there will be a group of frequencies which will last longer than the rest. Natural reverb and springs will decay around low frequencies and thus will add "warmth." Some plates will decay around mid-high frequencies and thus will add "brilliance" while some of the digital units have control over this characteristic and will add the full range between the two.

Reverberation: It's Contribution

Probably the most important contribution any artificial reverb device can make to today's music, which is close-miked in a

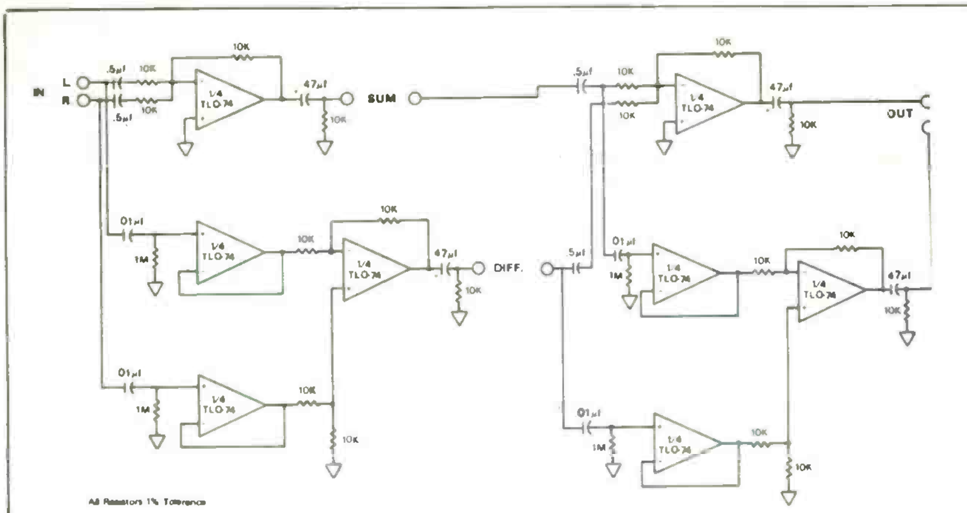

FIGURE 4

Parts list for an amplifier which generates a sum and difference output. This device can be placed between a reverb's output and the console's echo returns. The sum and difference outputs can then be brought up through additional board inputs.

- 4 - .5 µf
- 4 - .01 µf
- 4 - 47 µf
- 4 - 1 M
- 18 - 10 K
- 2 - TLO-74 or equivalent

dead studio, isolated, overdubbed, pan potted, etc., is to add the only *real* stereo effect, outside of stereo mikes, available to the recording engineer. A pan pot places an instrumentalist or vocalist between two speakers by intensity differences — no spatial or depth effect is possible. As noted earlier, the human hearing system can utilize its depth perception sense when each ear gets a slightly different image of the same sound field, such as two mikes in an acoustic chamber. Using two mono reverb units with widely different characteristics could result in very poor stereo reverberation. Also, widely spaced speakers and mikes in an acoustic chamber may give too much separation. The old trick of putting the direct sound on one channel and mono reverberation on the other is, of course, not stereo.

The exact correlation or cross-correlation which will produce the best stereo effect between the two channels of a reverb device are not known at this time. Those engineers who like to experiment with every conceivable device between the reverb and console might try them in the sum and difference channels of the circuit shown in Figure 4 which is best inserted in the reverb returns. Some "super stereo" effects are possible. □□□

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1 - Rettinger, M., "*Acoustic Design and Noise Control*," Chemical Publishing Company.

2 - Kuhl, W., "*Eigentone density and coloration of Reverberant Sound*," 6th International Congress on Acoustics E-2 — 5 (in German).

3 - Schroeder, W.R., "*Natural Sounding Artificial Reverberation*," JAES 10:3, p. 219.

emphasis: REVERBERATION

Acoustical **COMB FILTER** Effects

(. . . how to keep them out of your hair . . .)

by

F. Alton Everest

Phasing and flanging are certainly well known today and those using such effects generally associate them with the term **"comb filter."** Less widely apprciated is the fact that in many common recording and reproduction situations comb filter effects mess up our desired flat frequency response. This is a form of amplitude distortion which is inherent in practically every listening and monitoring setup, every single or multiple microphone mono pickup, as well as many mixdowns from stereo to mono. The magnitude of this distortion depends primarily upon the geometry of the setup, although other factors enter in.

Delay is the key word[1]. In dealing with sound as an acoustical phenomenon, delay is a direct result of the finite velocity of sound. For normal temperatures and near sea level sound travels about 1,130 feet per second, or 1.13 feet per millisecond. In evaluating practical problems, a very convenient thing to remember is that sound travels *about one foot per millisecond.*

A microphone is a rather blind sort of instrument. Its diaphragm responds to whatever fluctuations in air pressure occur at its surface. If the rate of such fluctuations (frequency) falls within its operating band it obliges with an output voltage proportional to the magnitude of the pressure involved. If a 100 Hz tone from a loudspeaker actuates the diaphragm of a microphone in free space, a 100 Hz voltage appears at the microphone terminals. If a second loudspeaker lays down a second 100 Hz signal at the diaphragm of the microphone identical in pressure, but 180° out of phase with the first signal, one cancels the other and the microphone voltage falls to zero. If an adjustment is made so that the two identical 100 Hz acoustical signals are in phase, the microphone delivers twice the output voltage, an increase of 6.02 dB. The microphone slavishly responds to resultant pressures acting on its diaphragm. Little did it know (excuse the anthropomorphism) that when the two identical 100 Hz acoustical signals were in phase opposition that air molecules a short distance away from the diaphragm were obediently doing their violent 100 Hz dance. In short, the microphone responds to the vector sum of air pressure fluctuations impinging upon it. We must remember this characteristic of the microphone as we dive into a consideration of acoustical comb filter effects.

— the author —

F. Alton Everest has been involved in sound and acoustics since the mid-thirties. He has been involved in the research of acoustic problems as well as the practical applications of their solutions. He has authored numerous books including the "Handbook of Multichannel Recording" (Tab), "Handbook of Public Address Sound Systems" (Tab), and "Acoustic Techniques for Home and Studio." His achievements over the years are numerous and include co-founding the Moody Institute of Science, Whittier, California, where he was director of Science and Production for twenty-five years.

A Description

Now, as an astounding revelation to those who are not sure just what a comb filter is, and as a review to those patient ones who do, we shall examine this ubiquitious effect in detail. We have seen that when two different airborne acoustical waves, A and B of Figure 1, arrive at a given point in space, such as our microphone diaphragm, they combine vectorially, that is, with due regard to amplitude and phase. If A and B are identical sine waves of approximately the same amplitudes we have a highly simplified situation. With the 100 Hz example, combining in phase doubles the amplitude, combining in phase opposition results in cancellation. The same is also true in combining identical, but highly complex, signals.

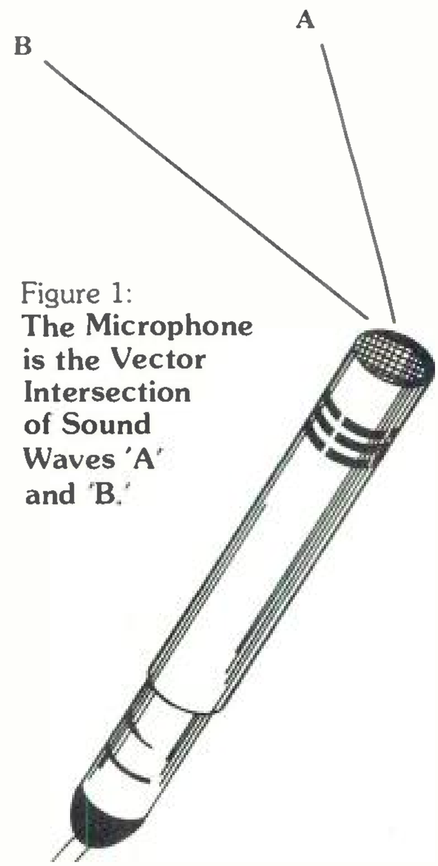

Figure 1: **The Microphone is the Vector Intersection of Sound Waves 'A' and 'B.'**

It is helpful to consider how this interference effect acts down through the audio spectrum. Comb filter interference can radically affect the overall frequency response even though the system components are flat. Let us assume that a microphone diaphragm is actuated by the combination of two signals, a signal direct from the mouth of one talking, and the same signal delayed 0.1 millisecond. Without the

continued overleaf . . .

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delayed signal let us say that the system response is flat and represented by the straight line at 0 dB in Figure 2(A). Adding to a given signal the same signal delayed 0.1 ms, the response undergoes some surprising changes. At those frequencies at which constructive interference takes place the response is boosted 6 dB. Midway between the 6 dB peaks, destructive interference creates dips infinitely deep, theoretically, 20 or 30 dB deep in practical situations. Significant energy is removed from the

Figure 2a

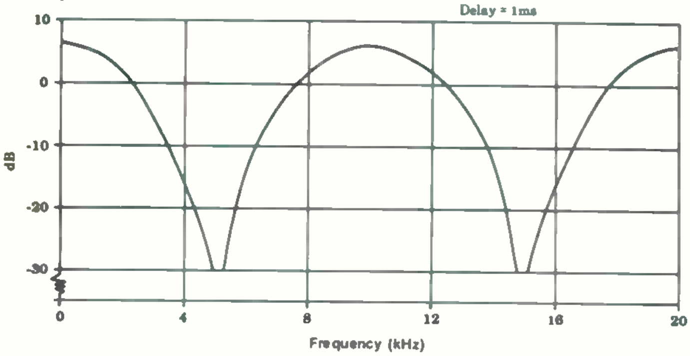


Figure 2b

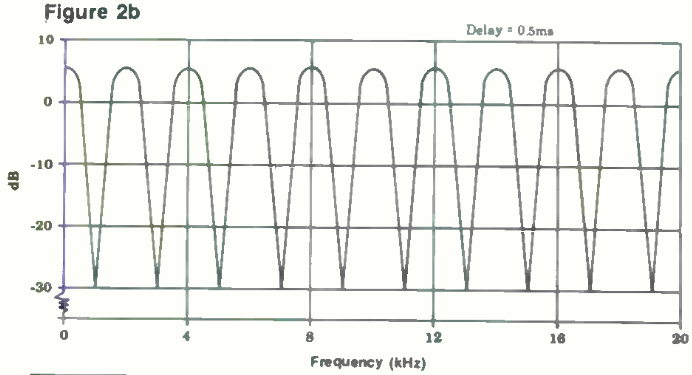


Figure 2c

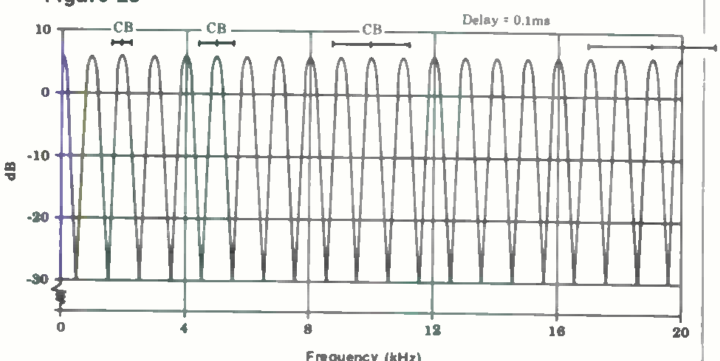


Figure 3a

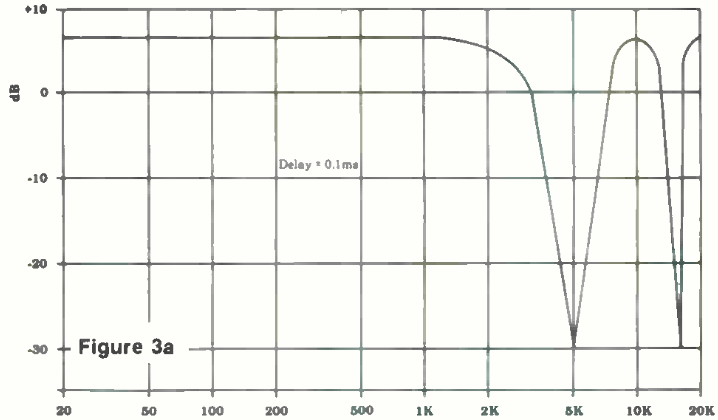


Figure 3b

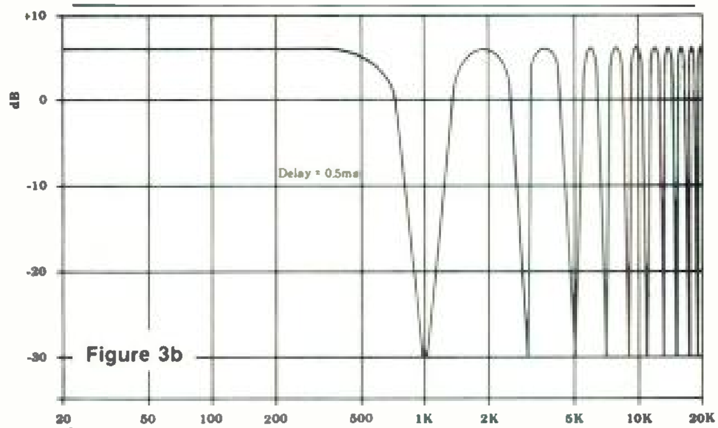


Figure 3c

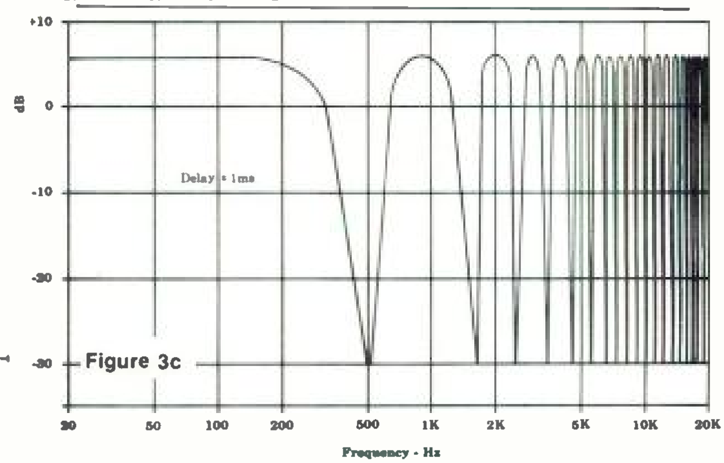
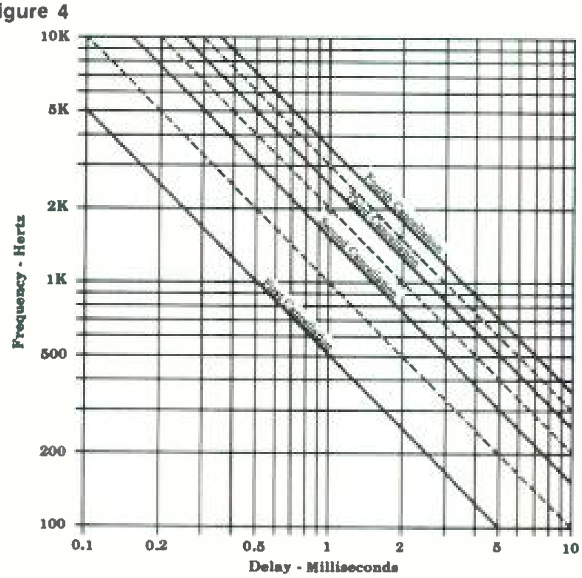


Figure 4



signal spectrum in the vicinity of 5 kHz and 15 kHz and unnatural peaks introduced below 2 kHz, above 18 kHz, and in the 9 - 12 kHz region. Note that a linear frequency scale is used to show the symmetry of the peaks and dips.

If the delay is increased to 0.5 ms the peaks and dips are much closer together as shown in Figure 2(B). It is now apparent why the comb filter name was applied to this effect. Peaks now occur at 2, 4, 6, and 8 kHz and at every other 2 kHz interval up through the spectrum. Between each pair of peaks is the accompanying dip.

Figure 2(C) illustrates the comb filter effect when a signal is combined with itself delayed 1 ms. Peaks are now separated 1 kHz, as are the dips.

Now that it has served its purpose of showing the inherent symmetry of the comb filter response, let us abandon the linear frequency scale for the more familiar logarithmic scale. Figure 3(A) shows the 0.1 ms delay case plotted in conventional semilog form. This gives a much better "feel" as to the effect of 0.1 ms delay on signal quality. The notches at 5 and 15 kHz would significantly color both speech and music. The 6 dB increase in level as a widening of the effective width of the dips.

The 0.5 ms case is presented in Figure 3(B) on semilog coordinates. The dips appear very narrow in this plot, especially at the higher frequencies. Readers who have had experience with controlling feedback frequencies in sound reinforcement systems by applying numberous narrow notch filters might say that the dips of Figure 3(B) might be tolerable, but not welcome. No matter how it is viewed, it is a significant deviation from a flat response.

Figure 3(C) illustrates the 1 ms delay example on a lot frequency scale. Dips at 500, 1,500, 2,500 Hz, etc., are interspersed with peaks at 1, 2, 3 kHz, etc. Looking at all three parts of Figure 3 we note the general principle that the longer the delay, the more the dips extend toward the low frequencies. Table I tabulates the location of the first null and spacings between adjacent nulls and adjacent peaks for delays from 0.1 ms to 50 ms. The same information in graphical form

TABLE 1
COMB FILTER PEAKS AND NULLS

Delay ms	*Frequency of Lowest Null Hz*	*Spacing Between Nulls Spacing Between Peaks Hz*
0.1	5,000	10,000
0.5	1,000	2,000
1.	500	1,000
5.	100	200
10.	50	100
50.	10	20

is shown in Figure 4. The broken lines between the cancellation lines of Figure 4 show, of course, the location of the peaks.

Effect of Relative Amplitudes

In Figures 1, 2, and 3 it has been assumed

— continued on page 68 . . .

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. . . continued from page 66 —

that both the direct and the delayed signals are of equal amplitudes. In practical situations in which the delayed signal is the result of a less than perfect reflection, or arrives at the microphone at an angle at which response is down (e.g., off axis on cardioid mike), the delayed signal may be reduced in amplitude. Further, the inverse square law hasn't been repealed, in fact it hasn't even been called out of committee. A component travelling farther arrives at lower amplitude than the component of signal travelling a direct path.

The boost will be less than 6 dB above normal and the nulls will be less than minus infinity if the delayed signal is less than the direct. Considering these two amplitudes as a ratio of unity or less, Figure 5 enables one to determine theoretical peak height and null depth. If the delayed component is 80% of the direct, the peak height is above 5 dB and the null depth about 14 dB.

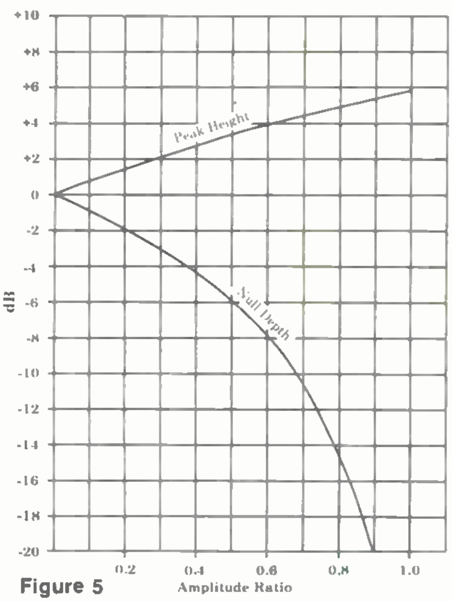


Figure 5

Comb Filters In Practice

There are many ways to generate comb filter effects. An easily manipulated method is that of Figure 6 in which a signal and a

Figure 6

delayed version of itself are combined in a linear network. Applying a repetitive swept sine wave to the input and observing the output on a cathode ray oscilloscope for different delays, the responses of Figures 2 and 3 are readily reproduced.

Example #1

Getting out of the laboratory and into the real world, consider the podium microphone arrangement of Figure 7. Believe it or not, such arrangements with both mikes feeding into the same amplifier can still be found.

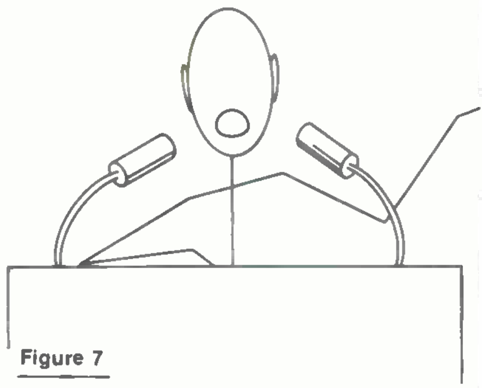


Figure 7

The excuse for using two mikes this way is that it gives the talker greater freedom of movement, but how about interference effects? Assuming the microphones are connected in phase (and they might not be if this practice is typical of the local personnel), and if the talker is dead center, there would be a helpful level boost of 6 dB. What happens if he is 3 inches off the center line? Let us assume further that the microphones are 24 inches apart and that the talking lips are 18 inches from a line drawn through the two microphones and on a level with the mikes. If the talker is centered, the sound travels the same distance (21.63") to each microphone. If the talker moves laterally 3", he comes closer to one microphone (20.12") and increases his distance to the other (23.43"). The difference between these two distances (3.31") results in the sound arriving at one microphone about 0.2 milliseconds behind the other. Result? A comb filter with a nice null gouging out important speech frequencies. If the talker were clamped in this position, the speech quality would not be good, but it would be unchanging. Normal talker movement results in very noticeable changes in quality as the nulls and peaks shift up and down the frequency scale.

Welcome evidence that such effects are real and not just theoretical scare tactics has come out of the Electro-Voice anechoic chamber. Lou Burroughs gives numerous examples of wildly distorted responses due to what he calls "acoustic phase cancellation," or comb filter effect, measured in simulated setups[2].

Example #2

Another situation much more common than Example #1 but similar in principle is

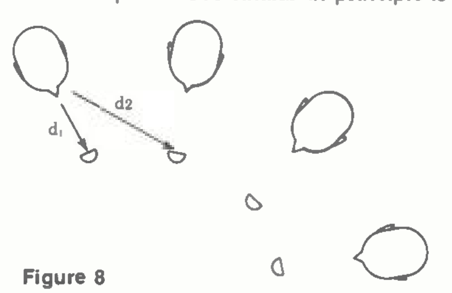


Figure 8

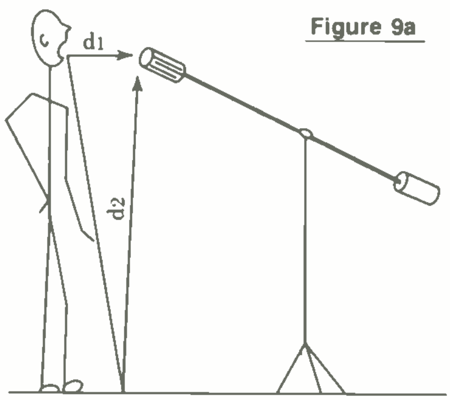


Figure 9a

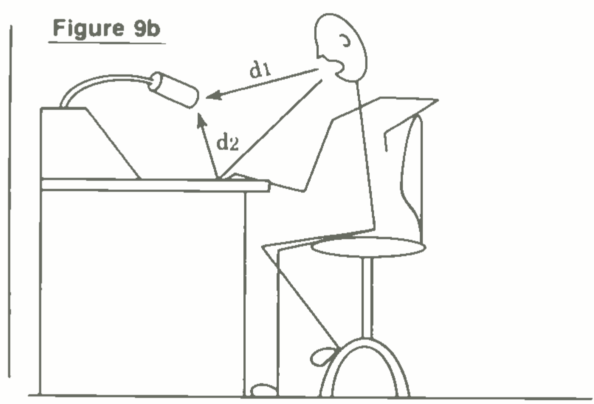


Figure 9b

the case of multiple sources close together, each source associated with its own microphone. Let us consider the singing group of Figure 8 with each of the four singers having his or her individual microphone. Singer A is d_1 inches from his or her own mike and d_2 inches from the next mike. The voice of A, picked up by both mikes, is mixed in the mixer with all the comb filter effects resulting from the path difference between d_1 and d_2. As each singer's voice is picked up, one degree or another, by all microphones, the situation gets more and more complex. Fortunately, sounds picked up by the more distant microphones are weaker and the comb filter boosts and dips are correspondingly reduced as per Figure 5. The experiments reported by Burroughs[2] indicate that if singer A's mouth is at least three times farther from the neighboring mike than from his own, the comb filter effects are negligible. If all singers place their microphones inside their mouths (some come close to this), the comb filter effects are submerged by other problems. If the "proximity effect" of the microphone boosts the bass, perhaps this amount of proximity will completely eliminate the highs as well as comb filter effects.

Example #3

Single microphones, like single men and women, can have their problems. Concentrating on the microphone side of the analogy, reflecting surfaces result in signals arriving at the microphone somewhat later than the direct signal. Figures 9(A) and 9(B) illustrate the case of the talker or singer standing before a microphone on a floor stand, or even hand held. In this case the floor reflected component (d_2) is much weaker than the direct (d_1) because (a) the distance of travel is greater, (b) the angle of arrival at the microphone is off the main axis, and (c) there is energy lost at the floor reflecting surface. Let us consider two simplified, specific cases, one in which d_1 = 10" (Figure 9(A)), and another with d_1 = 50", both having a microphone height and soloist mouth height equal to 56", and a floor reflection coefficient of 0.95. We shall also assume a cardioid microphone which would give a response at 90° about 3 dB less than on axis. With the source 10" from the microphone, the floor reflected component is delayed about 7.6 ms, but this information is significant only if the amplitudes are reasonably comparable. Considering only path length differences (10" direct vs. 112.4" reflected) we would expect the reflected component to be about 21 dB below the direct due to spherical divergence (inverse square law). The reflection loss at the floor is only about 0.4 dB and the 90° off axis cardioid loss is another 3 dB. This places the reflected component more than 24 dB below the direct. Obviously, the resulting comb

continued overleaf . . .

filter interference effect would be negligible.

The soloist now moves back to a point 50" from the microphone, either to improve "ambience," simply shrinking in fear, or for some other reason good enough for us to get on with this example. The direct path (50") is now somewhat more comparable to the reflected (122.7") and the reflected component arrives at the mike about 5.4 ms later than the direct. This would put the first null around 100 Hz, the second one near 300 Hz, which could be serious if the amplitudes are close enough in magnitude. It turns out that the inverse square loss is about 7.8 dB to which we must add 0.4 dB for floor reflection and something like 1 dB for cardioid pattern 66° off axis making a total of something like 9.2 dB. This corresponds to a ratio amplitude of about 0.35 which, from Figure 5, indicates we can expect nulls about 4 dB deep and peaks about 2 dB high, or overall perturbations of our response of about 6 dB.

There are things we can do to improve the situation if the soloist must be 50" from the microphone. A rug placed at the position of the floor bounce can be very effective in the upper audio frequency range. A super- or hyper-cardioid microphone could be used to reduce the reflected component another decibel or two. Thus our readily available remedies are quite limited, leaving the 50" distance to the microphone with basic problems unless a "shotgun" microphone is used which may well introduce a different set of problems.

Other close and distant microphone pickups are illustrated in Figure 9. Figure 9(B) shows a common geometry which can result in serious degradation of quality. Assume that d_1 = 12" and d_2 = 25". Then sound along the d_2 path would arrive close to 1 ms after d_1 which takes significant notches from the signal spectrum. the amplitude ratio would be close to 0.5 and, referring to Figure 5, we see that interference peaks would rise to about +3 dB and nulls would dip to about –6 dB giving overall response irregularities of about 9 dB. Closer talking will help reduce this as well as a good sponge rubber pad on the desk top.

Distant microphone pickups, such as in Figure 9(C) give distinct cancellation effects. By placing the microphone on the floor, d_1 is made approximately equal to d_2 (Figure 9(D)) thus minimizing the comb filter effect. There are several stands and foam protectors designed to support the microphone very close to the floor surface. Another recent approach is the Pressure Zone Microphone of Ed Long and Ron Wickersham (Alembic, Inc.) enthusiastically promoted by Don Davis[3].

Figure 9c

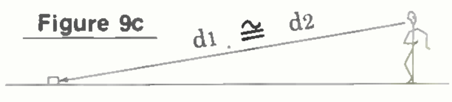
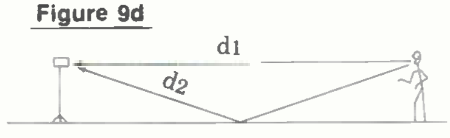


Figure 9d

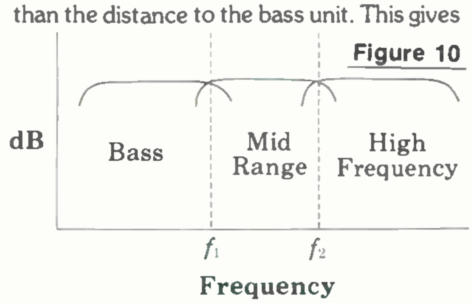


Example #4

Radiating the same signal from two separated loudspeakers or groups of loudspeakers lays down a comb filter pattern over the audience area[4]. On the line of symmetry between the two groups of radiators signals arrive at the same time and no comb filter effects are noticed, at least this could be true for one ear with the other ear plugged. Moving to either side of this line means that the auditor is closer to one group than another and delays generate the classical comb filter effect. In areas where one loudspeaker group is much stronger than the other due to path length differences and resulting inverse square fall off, the effects are modest. Directivity of the radiating sources also influences the area of the interference zone. Reverberation will also influence the detectability of the effect. With loudspeakers 25 feet apart, the 5 ms contours lie about 12° *to either side of the line of symmetry down the center aisle as seen in Figure 4 - 5 of Reference 4.*

Example #5

Multi-element loudspeakers can have their own private comb filter effects in the crossover region. In Figure 10 it is apparent that frequency f_1 is radiated by both the bass and midrange units, that they are of essentially equal amplitudes, and that the two radiators are not physically at the same point. This means that at a point in front of the loudspeaker the distance to the midrange unit may very well be different than the distance to the bass unit. This gives all the ingredients for generation of comb filter perturbations of response in the crossover region. The same process is active at f_2 between the midrange and tweeter units. Actually a narrow band of frequencies is affected, the width limited by relative amplitudes of radiations from the two adjacent units. The steepness of the crossover curve determines the width of the frequency range affected. This is a highly



complex problem that is being very actively studied at the present time and at least one monitor loudspeaker is being offered for sale claiming to minimize these defects (UREI 813)[3]. *(See editor's note.)*

How Audible Are Comb Filter Effects?

Just how serious a threat to the quality of our signal is this comb filter business, anyway? Psychoacoustical research on this subject must provide the answer to this question with any degree of finality. In the meantime subjective evaluations are all we have. Aside from its use for special effects, one inescapable conclusion is that comb filter effects certainly color the signal — but how much? For one thing, this depends on the position and depth of the nulls which, in turn, depend on the magnitude of the delay and the amplitude of the delayed signal component as compared to the direct. In general, comb filter effects give a "roughness" and unnatural "edge" to the signal. The critical bands of the human ear play a part in this as in all listening. The ear is a frequency analyzer with an analyzer bandwith (critical bands) of about 100 Hz below about 500 Hz. At 2 kHz the critical band is about 300 Hz wide and at 5 kHz about 900 Hz wide, etc. The 1/3 octave analyzer is a very, very rough approximation of the critical bands of the human ear[5]. The point here is that at higher frequencies each critical band encompasses many peaks and nulls of commonly encountered comb filter responses. In Figure 2(C) the length of the lines labelled "CB" indicate roughly the width of the critical bands of the human ear at four frequencies. For example, at 10 kHz the critical bandwidth would include 3 peaks and associated nulls for the 1 ms delay case. As far as the response of the ear is concerned, these 3 peaks and nulls would not be delineated individually, but be averaged together in some way. On the other hand, for the 0.1 ms delay case of Figure 2(A) the peaks and nulls throughout the audible band would be well within the analyzing capability of the ear. It would seem, therefore, that the effects of the 0.1 ms comb filter on our signal quality would be much more apparent to the ear than the 1 ms comb filter.

Comb filter effects that are changing catch the attention more readily than unchanging ones. With two loudspeakers in the split system sound reinforcement, those seated in certain areas might be aware of considerable change in quality by moving the head. Walking down an aisle could add an undulating swishing to the program material radiated. In outdoor split loudspeaker setups refraction due to wind changes can introduce a variable swishing or a variable rough edge to the signal.

We can all be grateful that the application of the technique of time delay spectrometry introduced in 1967 by Heyser[6] brings acoustical comb filter effects out of the closet. □□□

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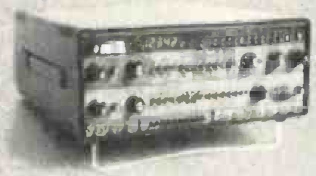
Editor's Note:
Additional information: Pearson, Don; Leo, Gary and Lubin, Tom, *Impulse Alignment of Loudspeakers and Microphones*, Recording Engineer/Producer, Part 1 - Volume 9, #6, December, 1978, page 88; Part 2 - Volume 10, #1, January, 1979, page 65.



Edcor Presents The Calrec Soundfield™ Studio System

The Calrec Soundfield Microphone Type CM4050 represents a new departure in microphone technology giving the recording engineer and producer unprecedented freedom and flexibility of microphone technique. Its outstanding feature is that it enables the effective polar pattern of the microphone, as well as its direction of pointing both in pan and tilt, to be adjusted not only remotely at a live recording session but also by post session processing of the master tape. This new facility goes much beyond the capability of the usual kinds of variable-pattern microphone, which must be set at the time of the recording.

In addition the Calrec Soundfield Microphone provides for the first time the effect of a stereo pair (or greater number) of microphones that are strictly coincident over most of the audio spectrum. This gives subjectively worthwhile improvement over the usual kind of close-spaced stereo pair in which phase errors can exceed 180° at the top of the audio band.

The Microphone Type CM4050 consists of a closely spaced array of capacitor capsules in the form of a regular tetrahedron within a single housing, and having head-amplifier circuitry incorporated in the stem in the usual way. It is based on an application of the mathematical theory of sampling on the surface of a sphere developed by Dr. Peter Craven and Michael Gerzon at the University of Oxford.

The Soundfield Control Unit Type CS5014/3 receives A-Format signals from the head amplifiers via a multicore cable, and by means of special circuitry converts these to four equalized signals at Line Level for the normal range of program level reaching the Microphone. These four signals, known as B-Format, are proportional respectively to the three orthogonal components of pressure-gradient, namely left-minus-right, front-minus-back, and up-minus-down. These signals are suitable for direct recording, or can be subject to further

for additional information circle no. 37

processing or encoding in the Soundfield Control Unit.

Controls are provided which allow any first-order microphone characteristic to be synthesized; that is to say, the complete range from omni-directional, through cardioid, hyper-cardioid, to figure-of-eight.

Moreover any number of such microphones, strictly coincident over most of the audio band, can in principle be synthesized simultaneously.

The Soundfield Control Unit has input-output facilities which allow mono, stereo or multi-track recording of any of the relevant signals with facilities for simultaneous replay and tape check. A separate gain control and selection system provides facilities for monitoring in mono, stereo or surround, the associated circuits automatically inserting the correct decoding system for the monitoring facility selected.

All circuits in both the Microphone and the Soundfield Control Unit are carefully designed for accurate gain and phase relationships, and the associated equipment should therefore be of a similar high order for best results.

The Calrec Soundfield Microphone Type CM4050 is an application of aspects of the Ambisonic Technology for surround reproduction developed at the Universities of Oxford and Reading, and with collaboration from the audio industry, under the auspices of the British National Research Development Corporation. Ambisonics is characterized by accurate encoding of direction onto two or more channels, and decoding in accordance with the hearing mechanisms of the ear.

The Ambisonic system seeks to create an illusion of the complete soundfield as recorded, irrespective of the listener's position within most of the listening area, a feature which up to the present mono, stereo and conventional quadraphonic reproduction techniques have failed to do. The latter usually suffer from very confined listening area and uncertain location of sounds from inter-loudspeaker directions resulting in listener fatigue. Single sound localization can be improved in conventional quadrophony using a signal actuated variable matrix decoder but this technique is only useful for surround-sound drama or larger-than-life directional effects and is usually unsuitable for serious music listening. The height information available with the Ambisonic system helps to improve directional and reverberant features even in a horizontal-only reproduction system.

The system employs a number of ingenious compromises based on psycho-acoustic investigations into how the listener hears a live musical or theatrical performance and what characteristics must be recorded and how later (or simultaneously) the recording may be processed so that optimized results are heard in a domestic or other environment with four or more loudspeakers.

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emphasis: REVERBERATION

Construction of a Live Echo Chamber

by
SCOTT PUTNAM
and
Tom Lubin

A live echo chamber can be a considerable asset for any recording studio, that is providing that it is a good one. That's the problem — how do you construct a good echo chamber? When someone builds a chamber, they hope it will turn out great and pray it won't turn out absolutely dreadful and good for nothing but storing echo plates. The truth is there are a number of complex variables which will make each chamber unique. These factors which effect the chamber include the type of wall construction and the selection of materials used on the inside surface. Probably the most important consideration is the cubic volume and physical proportions of the chamber. This leads to the first question to be asked before a chamber can be built. What space is available?

Space

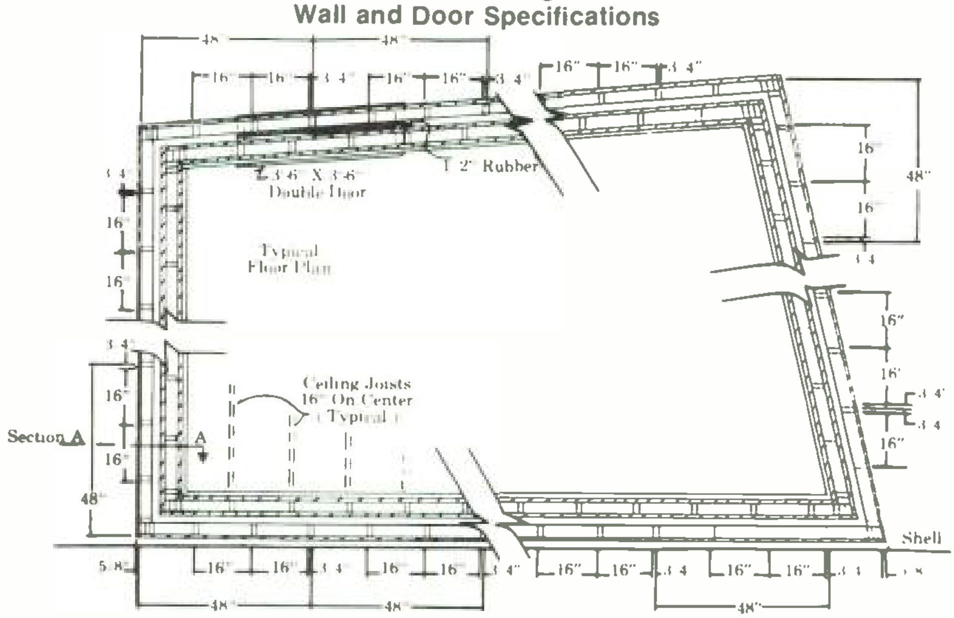
Most times echo chambers occupy surplus space. The more space that is available to start with, the easier the construction job since the builder won't have to deal with odd angles or cramped building conditions. The size of the chamber can generally vary from 1,000 cubic feet to about 2,000 cubic feet of internal volume. 1,500 cubic feet seems to get excellent results within a workable space. A small chamber won't get optimum results and the largest chambers are a luxury since they occupy a space large enough to be usable for other purposes. When determining if enough area is available, it is necessary to remember to allow foı figuring the wall thicknesses. There is also a need to consider a wide enough passageway around the structure for a hammer to be swung. If this isn't planned for putting on the exterior sheeting is going to be a difficult proposition.

It is also suggested that the space be large enough to accommodate a chamber with a minimum interior dimension of 7 feet. A chamber with a side shorter than this will usually give unsatisfactory results.

The Floor Plan

Once it has been decided that there is

Figure 1: **Floor Plan Showing Inner and Outer Wall and Door Specifications**



enough space, the next step is to design a floor plan (Figure 1). When laying out the sides of the room none of the walls should be parallel or even near to parallel. The ceiling should not be parallel to the floor. This is very important if the room is to have maximum random reflections and a smooth decay. It's at this stage that a bit of math should be introduced:[1]

$$T = \text{Const.} \times V/\bar{\alpha} S$$

where:
T = Reverberation time
V = Total volume of the room in cubic units
S = The total combined surface area of all sides in square units
$\bar{\alpha}$ = The average absorption coefficient
Constant:
.049 if measurements are in feet.
.161 if measurements are metric.

the author

Scott Putnam has been involved in studio construction since he was sixteen. He has designed and built a number of echo chambers including two for Kaye-Smith in Seattle, and a pair for the Record Plant. As a builder he has collaborated on various projects with a number of acoustic consultants and architects including Jack Edwards, George Augspurger, and Scott's Father, Bill Putnam. He is currently constructing Oceanway West in Santa Monica, California.

Figure 2: Various Ratios Used for Designing Room Proportions

Ratio	Description	Ratio	Normalized Ratio for Equal Volume
1:2:3	Harmonic	1:2:3	1:2:3
1.6:3:4	Vern O. Knudsen	1.6:3:4	1.08:2.03:2.71
3:5:8	European	3:5:8	1.1:1.84:2.95
1:1.6:2.5	J. E. Volkmann	1:1.6:2.5	1.14:1.82:2.85
2:3:5	P. E. Sabine	2:3:5	1.17:1.75:2.92
$(5^{1/2} - 1):2:(5^{1/2} + 1)$	Golden Section	1.236:2:3.236	1.12:1.82:2.94

The American National Standards Institute (in S-1.1-1960) defines reverberation time of a chamber as the time it takes for the mean square sound pressure level to decrease to 60 dB after a steady state signal has ceased.[2] Generally this level is referenced to 500 Hz although some information relates the level to 1 kHz. The following equation for figuring decay time was developed by Wallace Clement Sabin (1900). Since his time there have been a number of alternate equations developed but the original equation continues to be the most popular. This is due partly to the simplicity of the computation and the similarity of the resulting data.

Sabin determined that the reverberation time was related to the volume of a room, its surface area and the total amount of absorption. As the formula would indicate, the time is directly proportional to the volume and inversely proportional to the surface area of the room. With this being the case, the longest echo will be obtained when the required volume is achieved with a minimum of interior surface area. Trying to achieve more diffusion in the chamber by building accordian type splays will have the effect of cutting down the delay time because the total surface area will have been increased. Another point to be remembered when figuring out the length of the sides is that none of the dimensions between any two opposing walls should be the same or a multiple or fraction of any other two opposing walls.

There are a number of ratios which serve as guides when figuring acceptable proportions, including the Golden Section

$$(5^{1/2} + 1): 2 :(5^{1/2} - 1).$$

This relationship was proposed by the Greeks and divides a line in such a manner so that the smaller dimension is to the greater as the greater is to the whole. There are four or five other ratios which have been proposed and accepted to varying degrees,[3] but the one most often used is Sabin's 2:3:5 relationship. (Figure 2)[3]

Wall Angles

The angles used for the intersecting corners should not be severely acute. In practice, the simplest way to arrive at the wall angles is to build the sides so that two of the joining sides meet at right angles. Their opposing walls are constructed similarly, but without permanently nailing down the floor plate. Once these walls have been framed, the entire unit can be angled inward. For the average chamber moving one end of each of these two walls in by a foot or so should be sufficient.

Rigidity And Isolation

Two things which are important in making a good live echo chamber are maximum interior wall rigidity, and the total isolation of the chamber from its surroundings. The more rigid the wall is, the less energy dissipated when a sound wave hits it, hence the surface is more reflective. The isolation

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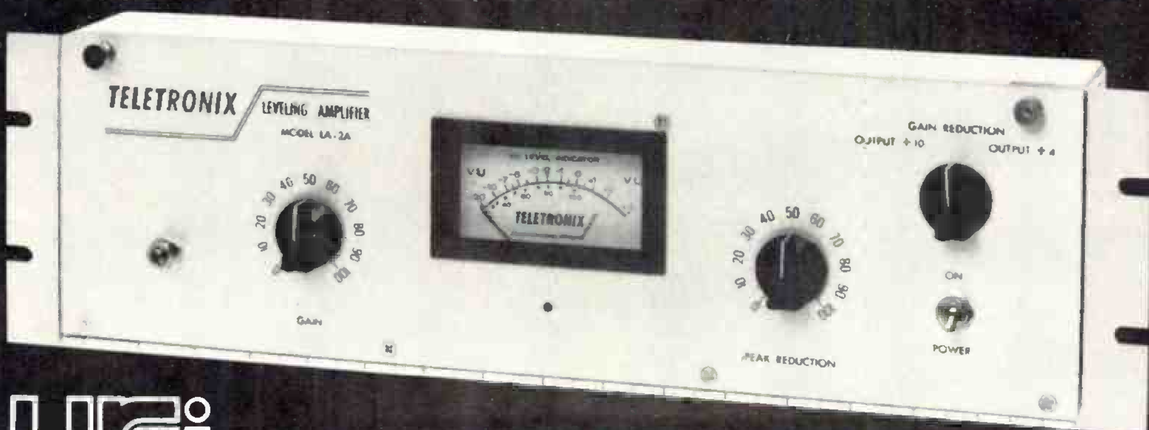

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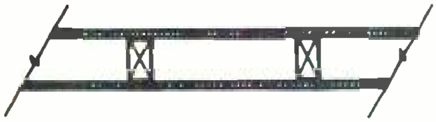
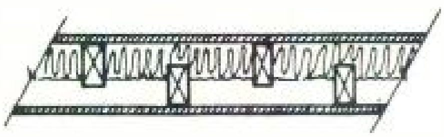


Standard stud wall, 2 x 4 studs 16" o c. 1 layer gypsum board on each side.

No insulation . STC 34 dB

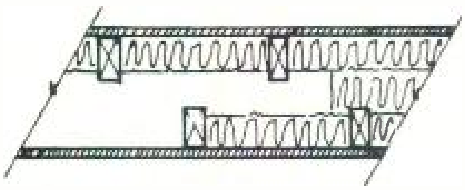
3½" Fiberglas building insulation 36 dB



Staggered stud wall. 2 x 4 studs 16" o c. on 6" plate. 1 layer gypsum board on each side

No insulation STC 42 dB

3½ Fiberglas building insulation 49 dB



Double stud wall. 2 x 4 studs 16" o c. on 4" plates, single layer gypsum board on each side

No insulation STC 43 dB

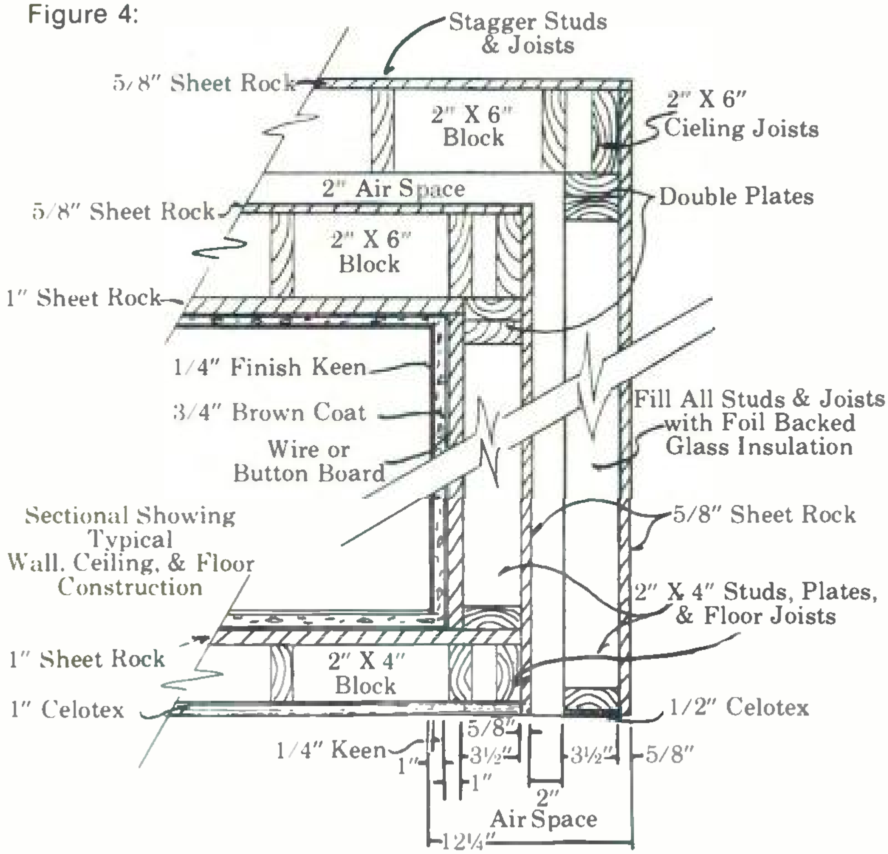
With 3½ Fiberglas building insulation 55 dB

With 9" Fiberglas building insulation 58 dB

Source Owens-Corning Fiberglas Corp

Figure 3: **Comparison of Stud Wall Construction and Transmission Loss Specifications**

Figure 4:



is important since it's essential that the rooms broadband ambience be very low.

Echo chamber wall systems which achieve these goals can be constructed from a variety of materials. There are a few chambers that have been built with walls, floors and ceilings of poured concrete. Such a design if properly executed will get very good results, but will be expensive to build, very permanent and extremely heavy. It will be there forever, providing you haven't underestimated the strength of your sub-floor.

The second most popular approach is concrete block walls. They are easier to build, and a bit easier to tear down, but once again you have the weight problem.

The most popular and cheapest type of construction is a wood frame design made of 2" x 4" and 2" x 6"s. It is the easiest to build with hand tools and more importantly is relatively light compared to concrete and block.

Walls

In recording studio construction it is very popular to use 2" x 4" staggered stud construction since the results achieve a better transmission loss than a conventional wall. (Figure 3) A staggered wall uses two rows of studs 8" apart (2" x 4" alternating 16" centers) on 6" top and bottom plates. Every other stud is flush to the opposite edge of the plates. In this way the wall sheeting of the two sides is only connected at the top and bottom plate.

A standard stud wall uses a 2 x 4 plate. The sheet rock covering both sides is connected through each of the common studs. The transmission loss between these two types of construction with insulation is STC 36 dB for the standard wall and STC 49 dB for a staggered wall.[4] The staggered wall would be the preferable wall design if transmission loss were the only consideration, however, rigidity is a more important factor in the design of the inside shell, hence standard construction is preferred. (Figure 4)

The reason for this is that staggered construction does not allow enough space for cross bracing. The interior shell should have a cross brace splitting the span of every stud, joist, and beam. There should be no unbraced span longer than eight feet.

The preferred layout is to use a staggered exterior wall, and a standard interior one. These two walls should not be coupled in any way and hopefully will not only be de-coupled from each other but from the rest of the building.

Floating Walls and Floors

A very important part of the de-coupling of an echo chamber is isolating it from the building it sits in. What sort of isolation is needed will depend on the specific situation and the funds available.

E=MC²

"OK, I give up G.C. . . . What's the deal?"

"It's Einstein's equation!"

"I know what it is fool, what's it doing at the top of this Valley People AD?"

"Well, it's Einstein's anniversary."

"Brilliant . . . How foolish of me! Where's the champagne . . ."

"No, seriously, it's been 100 years for Einstein and one year for Trans-Amp. Trans-Amp was conceived . . . uh er . . . developed one year ago this month!"

"I think I like conceived better . . . has a nice ring to it."

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"Perhaps Trans-Amp will affect our perception in a similar manner. Maybe life as we know it will be so profoundly influenced by Trans-Amp's unerring reality, decades . . . No! . . . centuries from now, people will look back and say, 'WOW!' . . . perchance Trans-Amp's faithful reproduction of everyday occurrences will prove instrumental in the compilation of data, necessary to explain the cyclic, yet insidiously complex, secret of life!!!"

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The drawings shown use Celotex for de-coupling. As can be seen, the outer shell is de-coupled from the floor of the building, while the echo chamber has a completely floated floor. If the location requires extreme de-coupling then it might be necessary to construct the outer shell on its own completely floated floor. It is also likely that something other than than Celotex will be needed.

Machine rubber is a good alternative to Celotex. It seems to work acceptably when used in a quiet environment but it does seem to compress a great deal and might break down with time. A thickness of ½" or more is necessary.

In an extreme isolation situation involving low frequency vibration more severe measures will be necessary. The worst case may need a floated concrete floor on spring isolators, but once again you have considerable weight and expense.

The best material which has been found for most situations are Fiberglas decoupling blocks. As shown, they are two inch cubes, covered with latex (to keep the moisture out) and are specially designed for floating floors. They can be used for both concrete and timber designs. When they are used with concrete, a sheet of plywood is layed over them and a border runs around the plywood to create a pouring form. Be sure all cracks in the form are sealed so that none of the concrete will seep and re-couple the floating floor to the structural one.

Latex Coated Fiberglas Insulation Block

When the blocks are used with wood construction, they should be set under three or four 2" x 6" headers. The timbers form the base for the echo chambers floor joists. Isolators should be placed about a foot or so apart along the entire length of the headers. The only problem with the block spacers is the gap it leaves between the floating floor and the structural floor. The solution is to fill the space with Fiberglas and run the sheetrock down to ¼" away from the floor caulk.

These blocks are available from a number of suppliers including Peabody Noise Control, 6300 Irelan Place, P. O. Box 655, Dublin, Ohio 43017. They also make a full line of ceiling isolators and latex covered Fiberglas board. The blocks can also be used to isolate ceilings from walls. For this application the blocks seem to work best if they are split in two.

A chamber built on a concrete ground floor can be further de-coupled by using a concrete saw to cut a slot around the perimeter of the chamber. This helps quite a lot in isolating the rest of the building from the chamber. The newly formed slit can be stuffed with Fiberglas but a hard sealer should be avoided since it will slowly harden and compress and re-couple the slabs.

Insulation, Sheetrocking and Sealing

All the walls should be liberally stuffed with foil backed Fiberglas insulation (3½" #R-11). Note: when working with the glass, be sure to always wear gloves, goggles, a mask, and clothes that won't allow the glass to touch your skin.

Both sides of each wall are covered with sheet rock. Two layers of ½" sheet rock is specified for the interior wall but two layers of 5/8" will work just as well or better. When two layers are used one should overlap the other so that none of the seams coincide. All seams should be taped and sealed so the room will be airtight. In addition to standard seam Hydroseal which is a black gummy adhesive is highly recommended. It is normally used as a roofing sealer and will stick to anything. It should be used liberally at every structural intersection of the sheeting or the stud construction. You end up using gallons of this stuff and practically glue the building together.

The purpose of the Hydroseal is to close every crack or seam. The smallest crack should not be ignored. If you have a 1/32" crack along the floor, the actual total area of that hole is considerable.

A final note on Hydroseal. It is suggested that trowels are unnecessary for the application. Wood shims made from construction debris works a lot better. These are good for only about one or two applications. It is almost impossible to clean Hydroseal off anything, including the applicator.

The Chamber Surface and Its Application

After the walls have been built and the last inside layer of sheet rock has been taped and sealed, the chamber is ready to have its reflective surface treatment applied. The reflective walls of the echo chamber can be made from a number of different materials. The key to how good a particular material is for this application depends on how rigid it becomes after installation and what its absorption coefficient is. Referring back to the earlier formula, the length of the echo is inversely proportional to this coefficient.

The absorption coefficient of any material is defined as "the ratio of sound energy absorbed by a given material to that which arrives at the surface from the source."[5]

Figure 5: **Absorption Coefficients at Different Frequencies for Various Materials**

Material	Coefficients 125 Hz	250 Hz	500 Hz	1000 Hz	2000 Hz	4000 Hz
Brick, unglazed	0.03	0.03	0.03	0.04	0.05	0.07
Carpet, heavy on concrete	0.02	0.06	0.14	0.37	0.60	0.65
Carpet, with latex backing on 40-oz hairfelt of foam rubber	0.08	0.27	0.39	0.34	0.48	0.63
Concrete block, coarse	0.36	0.44	0.31	0.29	0.39	0.25
Light velour, 10 oz per sq-yd in contact with wall	0.03	0.04	0.11	0.17	0.24	0.35
Concrete or terrazo	0.01	0.01	0.015	0.02	0.02	0.02
Wood	0.15	0.11	0.10	0.07	0.06	0.07
Glass, large heavy plate	0.18	0.06	0.04	0.03	0.02	0.02
Glass, ordinary window	0.35	0.25	0.18	0.12	0.07	0.04
Gypsum board, nailed to 2 by 4 studs on 16-inch centers	0.29	0.10	0.05	0.04	0.07	0.09
Plaster, gypsum, or lime, smooth finish on tile or brick	0.013	0.015	0.02	0.03	0.04	0.05
Plywood, ⅜-inch	0.28	0.22	0.17	0.09	0.10	0.11
Air, Sabins per 1000-cu. ft.	—	—	—	—	2.3	7.2
Audience, seated in upholstered seats, per sq. ft. of floor area	0.44	0.54	0.60	0.62	0.58	0.50
Wooden pews occupied, per sq. ft. of floor area	0.57	0.61	0.75	0.86	0.91	0.86
Chairs, metal or wooden, seats unoccupied	0.15	0.19	0.22	0.39	0.38	0.30

Coefficients above were obtained by measurements in the laboratories of the Acoustical Materials Association. Coefficients for other materials may be obtained from Bulletin XXII of the Association.

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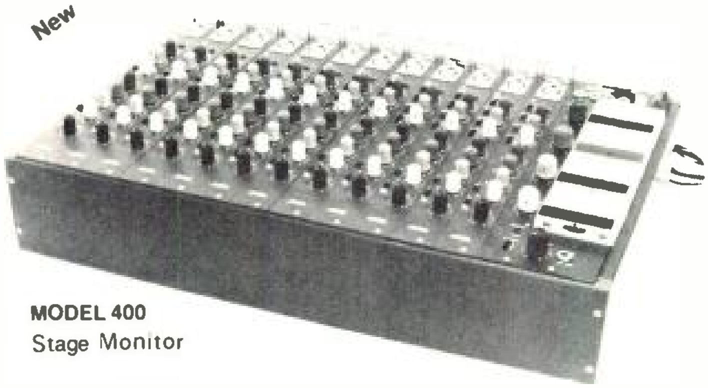

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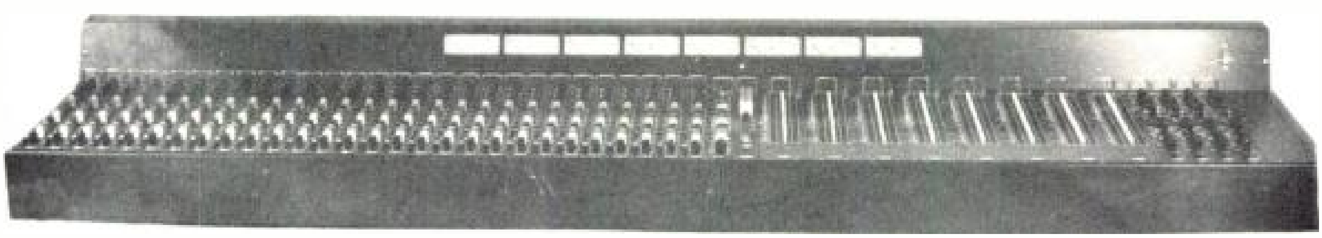

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Hence, a porous surface will have a much higher number than a reflective one. This figure will not only be different for each material but vary widely depending on the frequency. Figure 5 shows an absortion coefficient chart of materials versus frequency. As can be seen, plaster has a remarkably low number and is relatively flat. The plaster which Scott uses is Keen Cement (absorption coefficient of .015). He says, "when it is properly applied, it is as smooth as a baby's bottom."

Before plastering the surface, the sheet rock has to be prepared so that it will hold the plaster. This is done by nailing on the actual buttonboard.

The actual plastering involves the application of two different layers. The bottom one is a brown coat layer and is put on as a preparation for the top coat of Keen cement. It is strongly suggested that the actual plastering be done by a very good professional. The mixing of the two cements is nothing more than following the directions on their respective bags. One of the most important things to remember about the plastering process is the amount of time required for proper curing. As often happens the chamber is complete after the final coat of plaster is applied and there is a desire to use it right away. If the chamber doors are closed prematurely, ventilation will stop as will the drying of the walls. How long it takes for a wall to properly dry changes with the temperature, ventilation, and humidity of the environment, and can vary from a few days to a few weeks. The longer you can continue to ventilate a new chamber, the better. Scott suggests hooking up a dehumidifier in the room during the drying. He added, "you'll be surprised how much water comes out of those walls." This is a crucial part of getting the best end results.

"I have been in chambers that were built

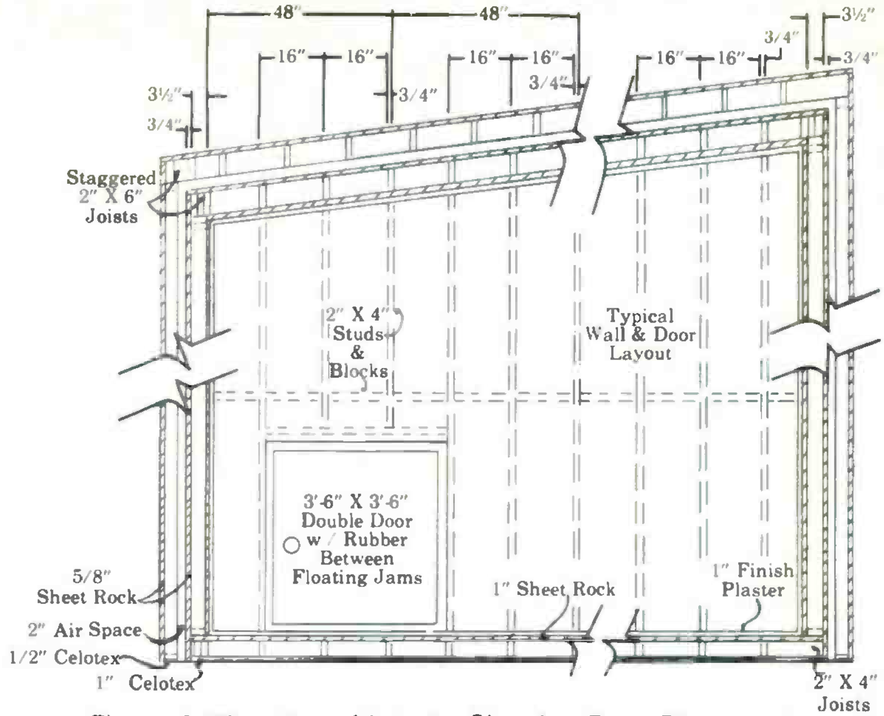

Figure 6: **Elevation of Interior Showing Door Placement**

years ago that have walls that have never completely cured." Needless to say such a room has a poor reverberation.

Some chambers sound bad only because the plaster coat was improperly applied. In such cases all that might be necessary is a re-plastering. It is also likely that an additional layer of buttonboard will be necessary so that the new plaster will have a firm base.

Connections And Doors

The chamber should have an IN and OUT opening so that the speaker and microphone lines can be kept separate. The pipe used should be flexible and have an I.D. of ¾". Increase the diameter if there is a need for more than two or three lines per pipe. The smaller it is, however, the less possibility there is for any leakage. Plastic hose works well because it is flexible, has gentle curves and there are no ridges inside for wires to hang-up on. Make the hole through which the hose is run as small as possible since you want to maintain the integrity of the wall system as much as possible. Once the hose has been installed, any gap between the hose and the wall should be completely caulked with Hydroseal.

A door is obviously needed somewhere in the chamber. Be sure the access to the door and the passageway leading to it is large enough to get a good sized speaker in and out of the chamber.

The two doors used should be of solid construction, not hollow. They are mounted on completely separate frames and jambs to coincide with the de-coupled inner and outer walls. (Figure 6) Machine rubber should be used between the two frames where they *almost* touch (½" to ¼") between the walls. Rubber should also be used completely around the jambs of the two doors. When they close the rubber should compress and make a tight seal. Since these doors are generally used infrequently, an elaborate closing is unnecessary.

Additional Construction Notes

A light should be installed somewhere inside the room. The switch for it can go anywhere convenient including right on the fixture. Be sure that de-coupling practices are maintained while running the AC conduit and mounting the fixture.

All the walls are built with studs on 16" centers. When laying out the studs, be sure to take into consideration that the drywall is 8' or 12' high and 4' wide and the centers of every third stud needs to line up with the edges of the drywall. #16 nails are used for all *end* nailing and #8 for *toe* nailing. Drywall nails are used on the first layer of drywall, but because of the added thickness #8 nails are used on the second layer. The buttonboard should be nailed on with #8 nails.

Be sure to estimate lumber lengths thoughtfully, to limit the amount of scrap. However what is left should be used for blocking. The greater the waste, the more the chamber (and for that matter any type of construction) will cost.

Speaker(s) and Microphone(s)

Changing the speakers and microphones or altering their placement will change the sound heard in the control room. Deciding what type of speaker sounds best or what microphone should be used becomes a matter of taste.

This is equally true as to where they are placed in the room.

Providing there was adaquate space to start with and care was taken with the design, construction, and isolation considerations, it is probable that the chamber will end up sounding very good. With a little bit of luck, it might turn out to be the sort of chamber that gains a reputation for itself as well as the studio that has it.

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1 · *Sound Recording Practices*, J. Borwick; "*The Acoustics*," by Alex Burd, Oxford Press, page 27.

2 · *Acoustic Design and Noise Control*, M. Rettinger, Chemical Publishing, page 25.

3 · *Acoustic Design and Noise Control*, M. Rettinger, Chemical Publishing, page 87.

4 · *Handbook of Multi-Channel Recording*, F. Alton Everest, Tab Publications, page 261.

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Studio Testing for Reverberation, Echo, Transparency, Rattle and Noise

by **Michael Rettinger**
Consultant on Acoustics

It is very difficult to apply a single-number index to the quality of acoustics in a studio. It would be convenient and significant if one could stand or walk in such an enclosure, listen to some music or dialogue generated in it, and then say that its tonal property is estimated to be 8 or 9 in respect to clarity of tone, uniform sound distribution, correct reverberation time, "warmth of low notes," etc. Even when the room is evaluated instrumentally in respect to these qualities it becomes hard to ascribe a single number to the overall listening or recording conditions of the enclosure.

Some qualities simply defy quantification. Consider what is sometimes called the "frequency response" of a room. What is frequently meant by that term is the sound transmission characteristic obtained when the output of a microphone in the room is recorded graphically while constant power is fed into a two or three way loudspeaker system. Anyone who has ever made such a test has learned that there is not one but many "frequency responses" in the enclosure, depending on the locations of the microphone and the loudspeaker.

In the following it is intended to describe only such tests in a studio which do not lead to disputes in respect to their assessment, so that at least within this scope one can say that the acoustics of the room are excellent, unimproveable, satisfactory, or else wanting in one respect or another.

1. Rattle test. The first test which this investigator generally performs in a newly completed studio consists in "cranking" the output of a beat-frequency oscillator through a studio loudspeaker at as high a level as the reproducer permits without becoming damaged. It is surprising how many rattling light fixtures, air-conditioning system register vanes, loose wall panels, etc., can suddenly be heard to make short sharp noises, sometimes threatening to fall into the room.

One may also touch one's fingers against a loudspeaker baffle or window pane in the studio to observe a sudden violent displacement at a given frequency. An interesting test of this type consists in putting a small steel ball or penny inside a paper disk, cut out in the shape of a doughnut, and placing the two on a horizontal wall of a loudspeaker cabinet or extension thereof. When the peak acceleration of the panel exceeds the acceleration of gravity (9.8 m/sec.^{2} or 386 inch/sec.^{2}), the solid object begins to chatter at a certain frequency, producing a "Bragg rattle," so named after Professor W. L. Bragg. An acceleration amplitude A of 980 cm/sec.^{2} at a frequency f of 30 Hertz corresponds to a displacement amplitude on part of the panel equal to $D = A/39.5f^2 = 980/39.5 \times 900 = 0.0275$ cm, not a small panel excursion at all particularly when the sound pressure level output from the loudspeaker diaphragm at 1 meter and at the same frequency, is only 70 dB referred to 0.0002 microbars. Such vibratile surfaces should be treated with a viscoelastic compound to eliminate these secondary radiations which are rarely in phase with the frontal emissions.

2. Sound "transparency test." This test consists in placing, in the studio, a microphone near the control room window, and opening, in the control room, both the microphone gain level control and the control room loudspeaker level control to their highest point to learn of any acoustic feedback which may occur when no signals are generated in the studio. Any howling which occurs in this test set-up indicates that there is either an acoustic short-circuit in the air conditioning system ducts in the control room window, or some other "weak" link in the sound insulation chain of the two rooms. In one such measurement this investigator was able to stop the acoustic feedback condition by merely placing a finger against the control room window, a clear indication that the window was insufficiently sound opaque or too sound transparent.

3. Noise level. There is presently available instrumentation to make a 24-hour continuous noise level recording on either magnetic tape or paper roll, using either analog or digital technique, while also obtaining the hourly average noise spectrum or the hourly A-weighted sound equivalent level. The latter term represents the level of a hypothetical *steady* sound, with the same spectrum as the noise under consideration, which steady level has the same total energy as the actual time-varying

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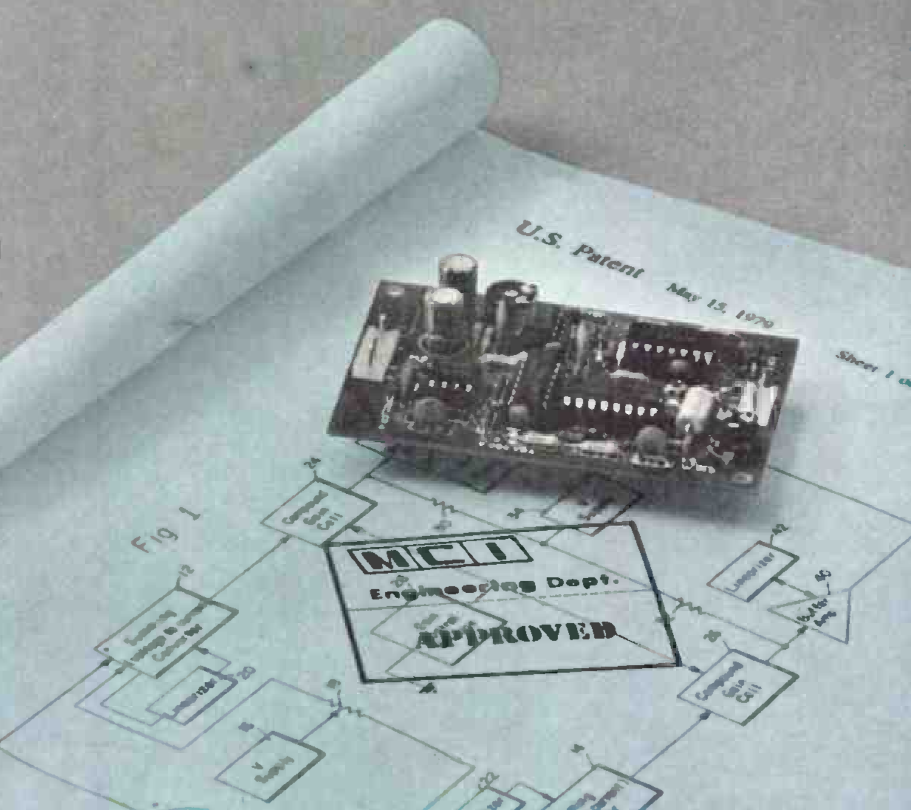

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disturbances, that is, the noise history for the hour with its temporal sound amplitude variations. It is either written as HNEL or L_n, in terms of dB-A values.

An adequately quiet studio has an L_n of 30 dB-A or less, while the HNEL of a noisy studio falls in the range of 30 to 40 dB-A. A studio with an L_n over 40 dB-A is generally unsatisfactory, particularly for digital recordings which can encompass a very high signal-to-noise ratio when the studio noise level is sufficiently low.

4. Echoes and echoettes. These pulses may be demonstrated by clapping one's hands and listening for the resulting sharp sounds when the signal is generated in front of a hard wall or ceiling. An echo is defined as a sound reflection which arrives at the listening post with such magnitude and delay after the arrival of the direct sound as to be distinguishable as a repetition of it. The figure of 70 milliseconds or 62 feet for the sound path difference is often quoted as a numerical limit for the phenomenon. An echoette may be assumed to be such a reflection which is barely audible, and which some musicians consider actually a strengthening or reinforcement of the direct sound. Not infrequently the term "attack" is also applied to this reflection effect, and is often wanted in the room by the performers in it.

In this test we are actually entering the field of psychoacoustics, the subjective evaluation of acoustic phenomena. Lord Rayleigh, the eminent English acoustician, said "Directly or indirectly, all questions connected with this subject (of sound sensation) must come for decision to the ear, as the organ of hearing; and *from it there can be no appeal.*" The italics of the last part of the previous sentence are the author's, to indicate the difficulty in judging the merit of room echoettes.

5. Reverberation time. There are a number of ways to evaluate the sound decay in a room as a function of frequency at any one test position. Usually a steady signal of some type — pink noise band, modulated frequency, or multi-tone (a number of closely spaced sinusoids generated by a number of oscillators) — is suddenly stopped and the sound decay is either recorded on magnetic tape or a graphic level recorder, or is evaluated digitally by the use of, say, a Bruhl & Kjar Type 2131 Frequency Analyzer, and a, say, Hewlett Packard HP 9825A calculator. Or a gun may be fired in the room and the decay of the signal recorded in one or another way described above. Other sound sources, like a burst toy balloon or a single handclap, have been employed in the test.

Regardless of the signal source used, it is necessary to employ a sufficiently long distance between source and microphone to achieve consistent results from point to point in the room. But this very requirement also introduces other difficulties, particularly in evaluating the results in modern studios where a short microphone distance is frequently required to achieve "clean" tracks in multiple track recording.

Figures 1 and 2 show the sound decay of a 250 Hz note 3 feet and 30 feet from the loudspeaker in a 18' x 33' x 52' studio. At 3 feet from the source, the reverberation times comes to only .3 seconds, while at 30 feet it is nearly .8 seconds.

But, even assuming that the reverberation time characteristics near and far from the source are "flat," against what criteria are they to be compared? Such standards of judging must necessarily be subjective, and since there is no disputing about tastes, it becomes difficult to arrive at a value for them which will please all. Someone has even suggested that the best reverberation time in the mid-frequency region of the audio spectrum could very well be different for different types of music, from a lengthy period for choral music to very short periods in the case of rock and roll renditions. If this is so, there is no one best reverberation period for a studio, but a range of periods made possible by a change of the absorptivity of the wall and ceiling surfaces of a studio, along possibly with an adroit addition of synthetic or electrtonic

Figure 1:

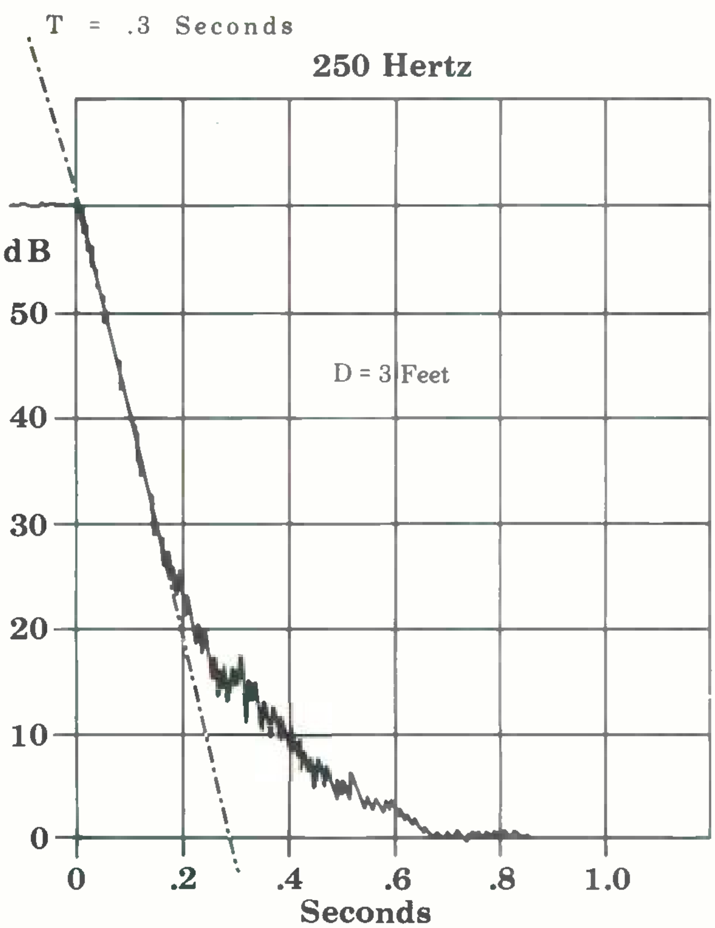


Figure 2:

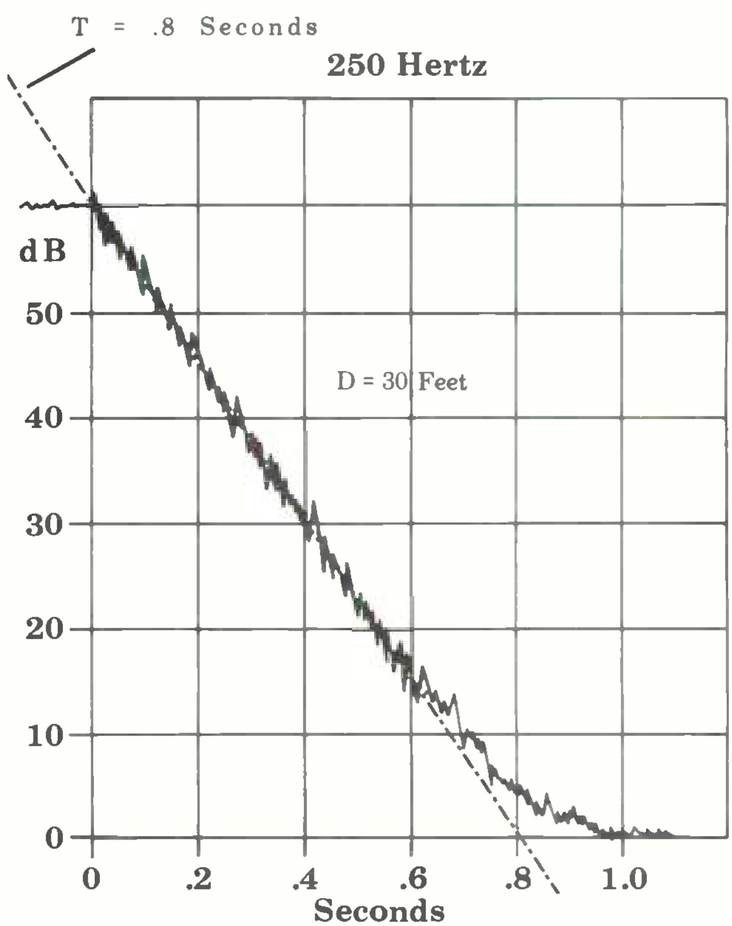


Figure 3:

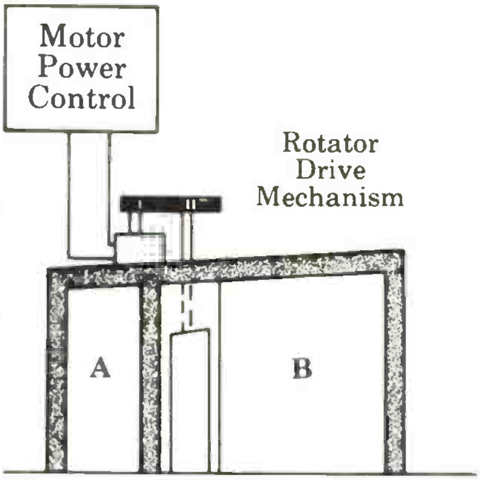
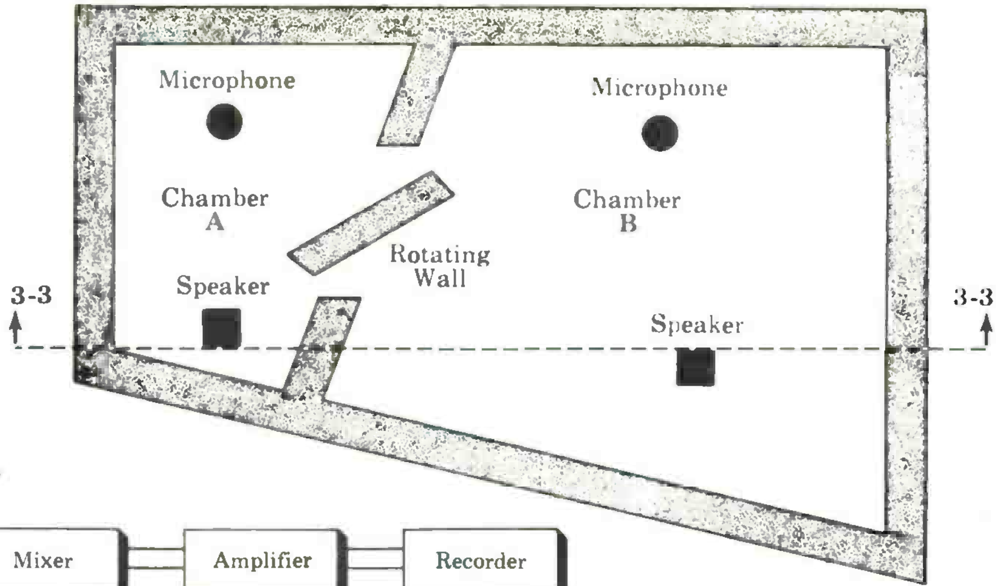


Figure 4:



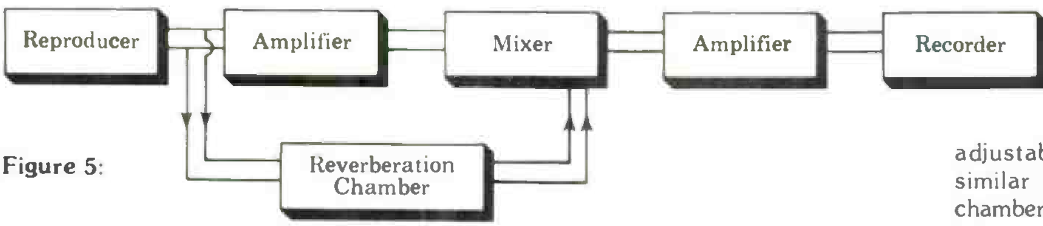


Figure 5:

reverberation.

As shown on Figure 1, the reverberation time curve for a microphone close to the source is characterized by an initial sharp decay, followed by a less steep decay. A similar effect can be had by placing the microphone close to the loudspeaker in a reverberation chamber. This effect may be adjusted to suit one's ear when the microphone position in the chamber is adjustable from the mixing console. A similar effect can be had by a double chamber, with an adjustable door opening between the two adjacent enclosures, also controllable from the mixing console. This was done at the RKO Studios in Hollywood in 1947, and this investigator was granted U.S. Patent No. 2,431,962, part of which is shown in Figures 3 - 5. □□□

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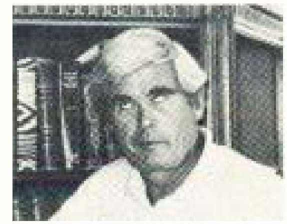

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OPTIMIZING CONTROL ROOM

REVERBERATION

by
Alan Fierstein

The importance of reverberation (T_{60}) measurements in control rooms having been established, it was felt that a how-to discussion of acoustical treatment would be useful. This article will demonstrate, by using practical examples, the procedure of measuring and adjusting reverberation. The most basic characteristic of a room's acoustical personality is its reverberation time. Although control room T_{60}s are typically less than half a second, and some will question the existence of a true reverberant field in such cases, the fact remains that this ambience must be measured and balanced with respect to frequency. This was the subject of a previous article by the author, *The Equalization Myth,* (R-e/p, June, 1977). As pointed out in that article, the graph of T_{60} vs. frequency is known as the "reverberation characteristic" or the "reverb curve." It is this reverberation characteristic that we loosely describe when we refer to a room as "muddy" or "harsh" or some similar term. It is important to keep in mind that the room is not some kind of graphic equalizer that boosts certain frequencies and attenuates others. Rather, the room absorbs sounds at different rates so that continuous music in the room will be "smeared" at different rates, as the frequency varies. Of course, a real-time analyzer will show steady frequency response irregularities with a steady pink noise test signal, but in reality there is a dynamic process of addition and subtraction of energy in the room on a moment-to-moment basis. I stress this often because many people see a T_{60} vs. frequency curve as nothing more than a fancy amplitude vs. frequency plot. It is not that! In fact, a room could conceivably have a lack of energy at the precise frequency

continued overleaf . . .

about the author . . .

Alan Fierstein is the president of Acoustilog, Inc., manufacturers of reverberation measurement equipment. Previously, he served as maintenance engineer at Media Sound and Electric Lady Studios in New York. He also designs, builds and maintains recording and film transfer operations in the New York area. He is owner, operator and chief engineer of Sorcerer Sound Studios in New York City.

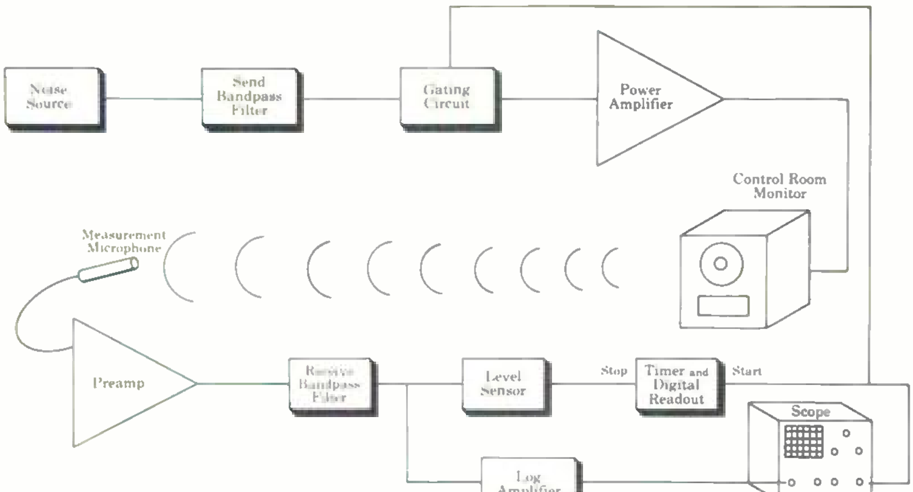


Figure 1: Measurement Set-up

where the reverberation time is highest.

I can describe two perfect examples of this. In some artificial spring echo chambers, the excessive reverberation time in the bass region is dealt with by the manufacturer by deliberately rolling off the low end either pre- or post-chamber with a simple bass cut filter. To the unsophisticated listener, or to a dumb Real Time Analyzer, this may flatten the response of a pink noise signal but to music the dynamic balance will remain as it was before adding the equalization. The lows, no matter how

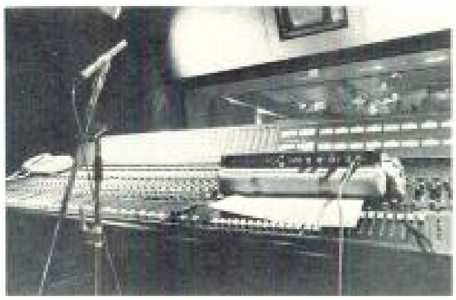


Figure 2: Equipment in use.

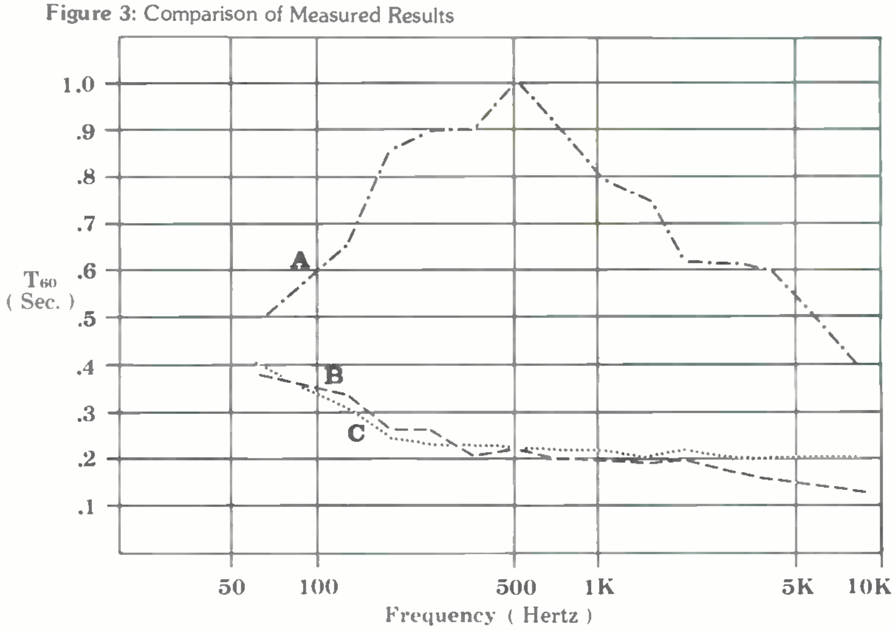
far down in level compared to the rest of the spectrum, will still linger for a longer period of time as new sounds enter the chamber. In a word — mud. Of course, in this application, which is to simulate the sound of a large hall, this mud is somewhat pleasing and natural. A similar situation occurs in control rooms, with excessive low-frequency reverberation, that is equalized simply by rolling off bass. T_{60} measurements will show the highest reverberation time at these frequencies that have been rolled off, indicating that the clarity of the bass has not been improved; in fact, nothing has been improved save the picture on the Real Time Analyzer's screen. Thus, we see that equalizing a loudspeaker in a room will not change the reverberation characteristic. However, changing the reverberation characteristic of the room *will* affect the frequency balance. If the room is first equalized, and then the reverberation time measured and adjusted acoustically, the equalization will have to be re-done. Add this to the growing distaste for any room equalization at all and we are led to the conclusion that the T_{60} must be corrected first, before the room is finalized or equalized.

The equipment setup for T_{60} measurement is shown in Figure 1. A noise source is filtered into the desired frequency band and applied to the monitors. When the noise has built up to a steady-state level, the noise source is gated off and the microphone senses and times the level drop. Because of a limit on the acoustical signal-to-noise ratio in the room, the full 60 dB of decay is usually not measured. Rather, the T_{60} is calculated by extrapolating the first part of the drop. At low frequencies, fluctuations in the pink noise test signal reduce the repeatability of measurements. Therefore, it is necessary to make multiple measurements and take their average.

In Figure 2, we see the equipment in actual use. The microphone should be positioned at ear height at the mixers position. Note that since the only factor being measured is the time taken for a level drop in any one frequency band at a time, the frequency response of the monitor system or the microphone is relatively unimportant.

In Figure 3A we see the first measurement of a completed control room, minus any acoustic treatment. Pretty terrible. Notice that the vertical scale of the graph starts at 0

Figure 3: Comparison of Measured Results



seconds. This way, you are certain not to run off the bottom of the graph if later measurements have lower T_{60}s and you wish to overlay the new curves on the old. Also, changes in the reverb curve will be shown in their proper perspective, that is, in relation to the total amount of T_{60}.

Sabine's Definition

Professor Wallace Sabine discovered an equation to predict the T_{60} if the room volume (in cubic feet) and area of absorbing materials (in square feet) is known. In his honor, one square foot of a perfectly absorbing material is said to equal one sabin of absorption, and this material has an absorption coefficient of 1. If a material only has a 50% efficiency of absorption, then the material has an absorption coefficient of 0.5. A square foot of this material has ½ sabin absorption, and 10 square feet will have 5 sabins, and so on. A material may have different absorption coefficients at low and high frequencies due to a differing ability to act on the different wavelengths involved. The manufacturers of acoustical materials rate their products by specifying the absorption coefficients at each of 6 frequencies, from 125 Hz to 4 kHz. We in the music business are interested in a wider range of frequencies than this. This is one reason why it is necessary to measure T_{60}, rather than simply calculate the expected reverberation of your control room. A more important reason for measuring T_{60} is the high degree of dependence upon mounting method and numerous detailed phenomena that may differ greatly from one room to the next.

But the published coefficients and the equations have a *very* important application after the T_{60} is measured. They enable us to calculate the amount of materials necessary to correct the room. Using the Sabine formula, you know in advance whether to order 100 square feet or 1,000 square feet of material to adjust your reverb curve to optimum. Now, although the Sabine formula has the least accuracy in dead rooms, we will use it in this article for several reasons:

1. It is very easy to use.
2. It is best suited for use with diffuse, scattered room treatments, and this is the prevailing situation in control rooms which are, in my opinion, properly designed.
3. This article presumes that the reader will actually make T_{60} measurements. This will allow for fine tuning of any inaccuracies due to the Sabine equation or published data.

The Sabine equation is:

$$T_{60} = .049V/Sa$$

where:

T_{60} = Reverberation time in seconds
V = Volume in cubic feet
Sa = Number of sabins.

Figure 4: Table of Results of Figure 3 Measurement Curve

Frequency	Present (T_{60})	Present Sabins	Desired (T_{60})	Desired Sabins	Sabins to be Added
63	.5	161	.2	404	243
125	.65	124	.2	404	280
250	.9	90	.2	404	314
500	1	81	.2	404	323
1K	.8	101	.2	404	303
2K	.65	124	.2	404	280
4K	.6	135	.2	404	269
8K	.4	202	.2	404	202

For our purposes, we will turn the equation around to:

$$S\bar{a} = .049V/T_{60}$$

and we can calculate the number of sabins we've got at each frequency. Then we make up a table that shows us where we are, where we're going, and how to get there.

Solving The Problem

For example, look at the room graph of Figure 3A. At 125 Hz the T_{60} measured .65 sec. The equation to solve is:

$$S\bar{a} = .049\ (1{,}650\ \text{cubic feet})/.65\ \text{seconds} = 124\ \text{sabins}$$

The next step is to list the results of the equation in a Table, shown in Figure 4. You can see the 124 sabins listed next to 125 Hz under the column "Present Sabins." What I deemed suitable for this room is a T_{60} of .2 seconds at all frequencies, and this is listed next to every frequency in the "Desired T_{60}" column. How many sabins would a room with .2 seconds have? Using the equation once more, we would calculate

$$S\bar{a} = .049\ (1{,}650\ \text{cubic feet})/.2\ \text{seconds} = 404\ \text{sabins}$$

and so I've listed 404 sabins next to each frequency in the "Desired Sabins" column. Now again turning our attention to 125 Hz, if we desire 404 sabins but we presently have 124 sabins, then I hope it is obvious that we should add the difference of 404 – 124 or 280 sabins and we should then reach our desired T_{60} of .2 seconds. After 10 minutes of work, the table is complete. The last vertical column "Sabins to be added" sums up our requirements. Since the numbers are all in the 200-300 range, it is easiest to add a broadband absorber like 3 lb. density 2" thick Fiberglas. This material has an absorption coefficient of .99 at all but the lowest frequencies, so each square foot will equal almost one full sabin. 240 square feet will equal 240 sabins at most frequencies.

Results

Thirty 8-square foot pieces were distributed over the ceiling and walls and the T_{60} measured once again. This was plotted as curve B in Figure 3. This is now a perfectly workable room and at most frequencies we see that we came very close to our goal of .2 seconds. In fact, this curve is flatter than most high-budget control rooms that I am called in to measure, but it could still be better. We could double the thickness of the Fiberglas, which would considerably tame the low end. However, it was decided to concentrate first on the high end. We covered 20 square feet of Fiberglas with a ¼" plywood panel and temporarily held it in place while taking reverb measurements. Curve C shows the improvement in smoothness. A little more low end is absorbed by the thin panel while the other frequencies are reflected. In this way we can modify the room to our requirements. If you're wondering about how to flatten the bass rise all the way, this can be done by using large areas of thicker Fiberglas, adding air spaces or by using frequency-selective absorbers such as large areas of panels, holes or slots. Anyone who tries to tell you that a properly designed 20 square foot hole in the wall can somehow provide 200 sabins of absorption or take care of the whole low-end problem is pulling your leg. Lotsa sabins means lotsa square footage. There are no magical acoustic vacuum cleaners that can reach out into the room and suck out all the low-end mud, although this is exactly what has been implied in recent literature. Reverberation data is usually lacking in such articles, and in the final checkout of too many studios.

Absorption Placement

Placement is very important for your absorbers to work correctly. If the absorbing surfaces are not scattered uniformly about, the Sabine equation won't work. Neither will the room. If large areas are left untreated, flutter echo will develop. This will show up as a high reverb time that cannot be reduced even by treating all the other surfaces. The untreated surface is dominating the other surfaces. Don't forget the ceiling. Left-Right, Up-Down, and Front-Back surfaces must each make some contribution to the total absorption. The new Live End-Dead End rooms are not exempt from these basic considerations. The front half of the room is treated completely on all surfaces. The rear half of the room is adjusted to yield the desired T_{60} by uniformly scattering the absorbtivity over all the surfaces to give a very diffuse field back to the mixer.

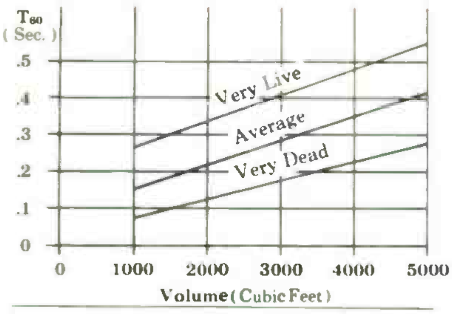
Don't be too concerned over a gentle rise at the low end. It usually means that your room is well-built and the bass frequencies will be solidly present in the room. But certainly don't let it rise uncontrollably.

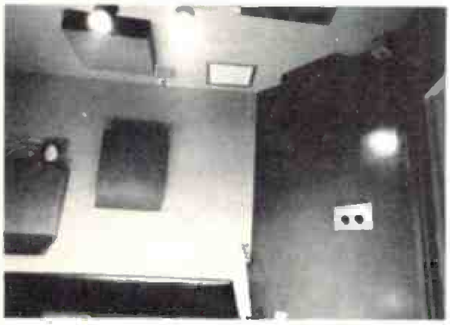
Many people wonder what is the proper reverb time to have in their control rooms. This is a matter of taste and my opinion on the subject is summed up in Figure 5.

I stated early that reverberation is the most basic parameter of an acoustical environment. However, I certainly do not wish to leave the reader with the impression that this is the only factor. In fact, I deliberately ignored all other considerations in order to concentrate on a fundamental like reverberation that many studio people have not seen articles on. Naturally, speaker placement, room shape, and construction materials, are critical to acoustics and I hope to comment on those important subjects in the future.

I would like to thank my friend F. Alton Everest whose book *Acoustic Techniques For Home And Studio* provides many useful examples of absorptive treatment and numerous photos of his work. □□□

Figure 5





Right: The rear corner of Hudson Sound, N.Y., showing scattered fibreglas panels and thin wooden panels which reflect mid and high frequencies

Below: The control room back wall at Zeami Studios, N.Y. Observe the widely spaced panels of this LEDE room. This approach contributes to excellent diffusion.

Bottom Right: The front wall at Zeami showing the border between the Live and Dead ends of the control room. Note the extreme (6") thickness of the absorbing material. (High density fibreglass)

References:

1 - Everest, F. Alton, *Acoustic Techniques for Home and Studio*, Tab Books, Summit, PA, 1973.

2 - Davis, Chips and Davis, Don, *Live End - Dead End Control Room Acoustics*, R-e/p, February, 1979.

3 - Rettinger, M., *Acoustic Design and Noise Control*, Chemical Publishing Company, New York, NY, 1973.

PROGRAMABLE DIGITAL REVERBERATION

by
David Griesinger

Reverberation — natural or artificial — is essential in recording. The versatility, cost and portability problems of large acoustic spaces being what they are, artificial reverberation from echo chambers, plates, or springs have become essential to the modern record producer. Recent advances in digital technology have made possible the construction of digital audio reverberation systems. However, building such a system is no simple matter. The major problem facing the designer of an artificial reverberation system is that so many different types of reverb are useful. Reverb, like EQ, miking and mixing, is very much a matter of taste. One producer's ideal of clarity, space and impact might just be sonic mud to another.

Because of the versatility of its hardware and software, the Lexicon 224 gets around such problems by providing two to eight very different types of reverberation programs. Its sound can be varied or completely changed while the music is playing. The ability to change types of reverb is unusual in any digital system, let alone a non-digital one. This particular digital design will never become obsolete since bringing it up-to-date is simply a matter of swapping a few integrated circuits containing the latest programs.

the author

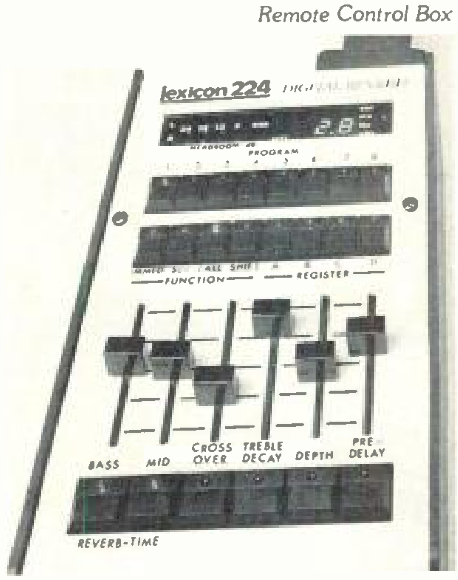
David Griesinger is a senior consulting engineer at Lexicon, and is responsible for audio software development. He is also an independent recording engineer, with records on Nonesuch, Sheffield (Town Hall), Desmar and other labels. He designed the software and preliminary hardware for the Lexicon 224. He holds MA and PhD. degrees from Harvard University in physics, and occasionally teaches in that department. He has published articles in the Physics Review Letters, The Journal Of The Audio Engineering Society, and Audio magazine. He is a member of the Audio Engineering Society.

Just What Is A Reverberation Program?

What is electronic reverberation and what is an algorithm? Natural reverberation is the combination of a very large number of time-delayed versions of the original sound (reflections). The strength of each reflection, its time delay, and its frequency response determine the sound of the reverberation. If we had a digital delay line with a very large number of taps, and we could control the amplitude and response of each tap, we could duplicate natural reverberation exactly.

Such a system can be built. We need a DDL with a very large memory. We then couple it to a high speed computer that can withdraw numbers from this memory, multiply them by constants representing the

Remote Control Box



Digital Reverberation Mainframe

proper amplitudes, add them together, and send the sum to the output. The program for this computer would be a reverberation algorithm. It describes the time delay and amplitude of each reflection.

Unfortunately, the number of taps needed is immense, between 1,000 and 10,000 per second of decay. The computer which performs the algorithm must have a new output sample calculated each time one is needed. If our sampling rate is 20 kHz the computer has only 50 microseconds to perform about 5,000 multiplications and additions. This is about 1,000 times faster than a general-purpose computer.

The amount of processing can be greatly reduced by using many shorter delay lines instead of one big one, and feeding back some of the sound in each line. This can greatly increase the effective number of taps. All practical electronic reverbs use such a technique. These principal factors are the time delay of each segment, how much feedback is used, the digital filtering applied to the feedback around each loop, and how all the lines are summed to make the reverberation output. These parameters constitute the program or mathematical algorithm which simulate the sounds of reverberation. Because the time available for computation is finite, it is not possible to exactly duplicate the properties of natural reverberation.

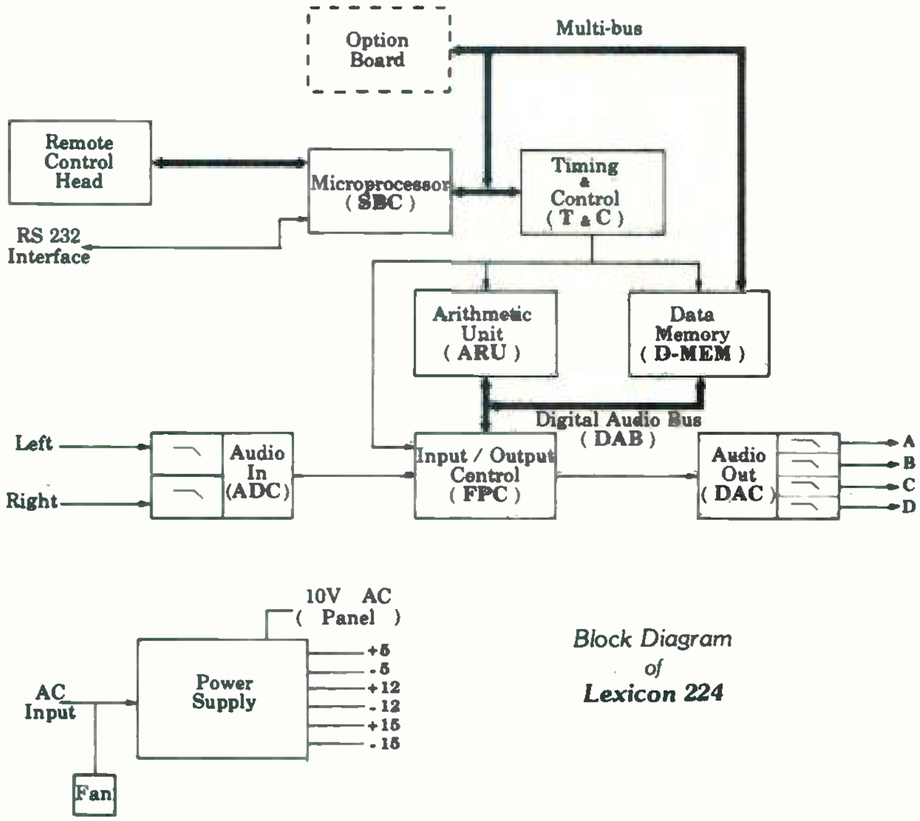
The 224's programs differ greatly both in the type of natural acoustics they try to simulate, and in the ways they differ from the idea. Writing such a program is trickier than it appears. It is easy to write repeating loops which give enough diffusion. Unfortunately, they usually sound impossibly metalic, much worse than a $2.00 spring reverb. The all-pass reverberator described by Schroeder is a big help, and Schroeder's reverberation algorithms are a good start *(Natural Sounding Artificial Reverberation - Journal Of The Audio Engineering Society, Vol 10, pp. 219,223 July 1962)*. Making an algorithm which sounds better involves a lot of listening.

What Is A Digital Processor?

The basic techniques of digital audio are getting to be old hat these days, but for those who are still in the dark, digital audio systems sample the input signal at some high rate. These samples are then converted into a stream of digital numbers, each representing the value of the signal voltage at the time the sample was made. Output from the device is obtained by reconverting this stream of numbers back into voltages, and smoothing the little steps which result.

A digital processor operates on the stream of numbers before they are reconverted. A reverberation processor is capable of putting the numbers in a memory, withdrawing them at later times, multiplying them by constants (the equivalent of gain control), and adding (mixing) them together. The resulting sums can then be placed back in memory, or they can be presented to the output.

Whatever the processor does, it must complete its operations before the next input sample is digitized. Any operation done by the processor must be done exactly the same for each input (and output) sample. The processor cannot put off some operations for later! This requirement means the processor must be fast (by today's standards). For example, a typical microprocessor can perform a multiplication of the type required for reverberation in about 50 microseconds. If a system based on this microprocessor had a sampling rate of 20 kHz (allowing a bandwith of 8 kHz) it could do only one multiplication in the time allowed. This is clearly inadequate. Fast technology is mandatory for reverberation.



Block Diagram of Lexicon 224

Griesinger:

PROGRAMABLE DIGITAL REVERBERATION

(continued)

How Does The 224 Work?

Basically it is two computers (see block diagram). The first is a proprietary high-speed processor which operates on the digitized music inputs to generate reverberation. The program for this processor is controlled by a 8080 microcomputer. The microcomputer continuously scans the front panel, and calculates the changes desired in the main reverberation program. Every reverb program requires different alterations, and the microcomputer must be programmed to make all these changes.

The reverberation programs and the program which controls the 8080 are stored in read only memories (ROMs). They can thus be easily updated. The microprocessor also performs a thorough set of diagnostics on itself and the music processor each time the power is turned on. If a fault is discovered, the nature of the problem is displayed, allowing quick isolation of any problems to a specific board.

The reverberation programs are written with the aid of compiler and assembler programs running on a large computer. It takes at least two months of work to develop a new reverb program. Much of that time is spent in revising the program of the microcomputer to allow it to control the new reverb program.

Initial Sound And Decay

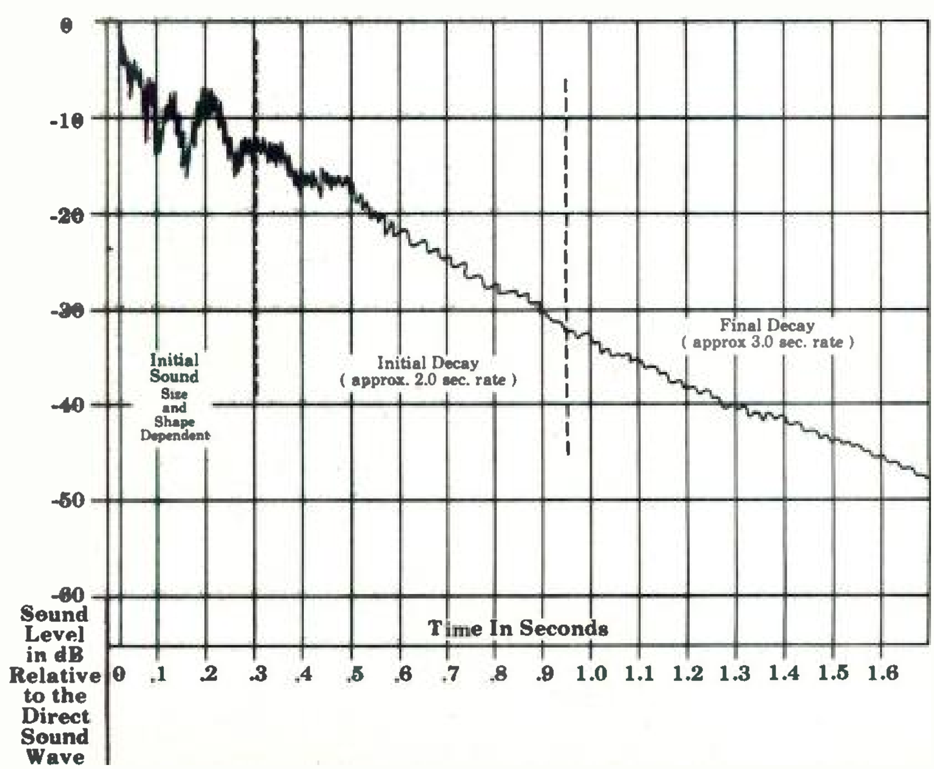
Figure 1 shows what we mean by initial sound and decay. It shows the amplitude of the reverberation in a real hall as a function of time after a single explosive input. It takes about 200 ms before the sound starts to die away smoothly. During the initial 200 ms period, relatively discrete reflections reach the listener from the stage floor, stage walls, hall walls, ceiling, and the hall's back wall. The apparent rate of reverb decay during this initial period depends both on the size and shape of the hall and on the position of the listener. The ear uses the first 200 ms or so of reverberation (the "initial sound") to perceive the size and shape of the hall.

After the inital period the sound decays smoothly. This portion of the reverberation curve we call the decay. The decay rate is usually assumed to be constant, resulting in a die-away of a certain number of decibels per second. However, in many real-world churches and halls used for recording the initial rate of decay is faster than the final rate. The slower final decay rate allows the space and depth of the reverb to be heard better when the music is played in a noisy environment or at less than the original sound pressure levels. The numbers displayed with the reverb time controls on the 224 refer to the time it would take for the sound to decay 60 dB if the decay were constant at the initial rate.

Diffusion

Diffusion is the ability of an acoustic chamber (and the 224) to spread a single pulse input into a very closely spaced series of output pulses. The best real-world example is the reflection of a click from an irregular wall. Very high diffusion spreads the click into a swish of sound. Lesser amounts of diffusion cause a more grainy reverb, and low diffusion can produce a series of obviously discrete clicks. Contrary to popular opinion, high diffusion is not always desirable, at least for the first 200 ms or so after an impulsive input. In concert halls, the floor and stage walls are usually not

Figure 1: *Reverberation Amplitude versus Duration of Decay*



very irregular. In halls diffusion is a function of time. As sound bounces around the room diffusion builds up. Low diffusion in the initial arrivals contributes to a very clear uncolored sound. Small rooms and echo chambers on the other hand, are designed to produce high initial diffusion, and work well on impulsive material. Such reverberation tends to spread and color the sound it is added to, giving it a louder, fatter quality. All Lexicon 224 programs have high diffusion after the reverberation has built up, but they have different degrees of initial diffusion. We find that symphonic music sounds best with low initial diffusion, vocals with moderate diffusion, and drums with high diffusion. The 224 reverb has the ability to manipulate all these elements.

Program Selection

Given all this flexibility, what types of reverb should be included in a digital system? A good popular record may contain many different types of artificial reverberation. A short, very diffuse reverberation, such as that from a fully-damped plate, may be used on the drum tracks, with perhaps some live chamber on the vocals, and a medium-damped plate with some time delay on the brass instruments.

In classical music the overhang from previous notes is frequently harmonically important — and a dry recording simply misses much of the music. Several types of reverb can be used. Engineers may deliberately use several mikes on a solo to enhance the spreading effect of nearby surfaces and create a less "close-miked sound." They may then add sound from a distant pick-up to add the depth and sound decay characteristics of the hall.

Empty halls or churches are frequently used in classical recording. But the natural reverberation at the recording site may not be useable. Either the hall has unsuitable acoustics for the music being performed, or balance problems in the performance make the reverb useless. Unfortunately, the type of artificial reverb useful in popular music is not always helpful in classical.

Lexicon has been developing reverberation algorithms for about five years. Both the programs and the final form of the 224 hardware were shaped by what we learned as we used these programs on popular and classical music. Our development is ongoing. We now have a large concert hall (it also makes a good stone cathedral), a small (more diffuse) concert hall, an acoustic chamber, and two types of plate. We are working on several programs, including a different type of percussion plate and another concert hall.

Percussion Plate Program

In popular music, a common use of artificial reverb is to remove the sense of close miking by spreading out the sound. Doing this requires reverberation with high diffusion and short overall decay time, to simulate the effect of a floor and walls around a performer. On drums it simulates a live and irregular drum cage. The initial sound should be short, and the pre-delay (the time period between input and the first processor output) should be close to zero. The percussion plate program is ideal here: it has very high initial diffusion, over 2,000 reflections per second in the first 50 milliseconds. The diffusion becomes more dense with time. Because of this amount of diffusion the sound has a slight but often desirable color. Transients from drums get smoothed into a swish.

The overall reverberation time can be made quite short (about 0.5 s) and is useful to about four seconds, when the coloration begins to become more apparent. If desired you can set the bass control to emphasize the treble as some plates do. Pre-delay is normally zero with the percussion plate, although the pre-delay can be increased to 100 ms with the pre-delay control. This program will take a stereo feed, and give a wide returned image. However, for best diffusion a mono feed (bridged inputs) should be used.

Vocals Plate Program

Compared to the percussion plate program, the vocals plate program has a brighter, clearer sound. It is very similar to the percussion plate but has less initial diffusion. While not as effective as the percussion plate program on a solo drum track, it has less color. It also has a slightly more non-uniform decay, and can give a real treble bloom, just like some metal plates do.

Acoustic Chamber Program

This program is at the mid-point between the reverb-oriented plate programs and the ambience-simulating concert hall algorithms. The chamber program is not highly diffused, has low color and a chamber-like initial sound. It is very pleasant on vocals and jazz.

Concert Hall Programs

Both concert hall programs have been designed to make their reverberation sound as if it is behind and around the music, not on top of it. This is commonly known as ambience. Low coloration and non-uniform decay permit the concert hall programs to give a sense of being at a performance of the music in a good real-world hall. They can greatly add to the naturalness of a mix, but since their initial diffusion is low they are not usually used with a solo track.

In popular music, the large concert hall program is ideal for effect, especially a sustain effect on a solo instrument. The register storage feature on the front panel can be used to set up the program with several very different decay times. The operator can then play the call buttons like a keyboard, to sustain some notes or chords much longer than others (see photo).

In classical music the main function of reverb is to sustain the harmonies in the music and to blend different instruments and phrases into a whole. It also adds a sense of depth to the sonic image, making some instruments recede deeper into the image between the speakers than others. The reverb should also give a sense of being in the hall — the size and the volume of the hall should be identifiable. Artificial reverb used as ambience should not muddy the direct sound at all. The reverb should be behind and around the original sound.

The concert hall programs create this type of ambient reverb. They are stereo programs. They take a stereo feed (usually from the two stereo channels) and produce four outputs. (They do a rather nice job of synthesizing quad.) The stereo input information receives primarily stereo processing. Two independent reverberation algorithms are used, although there is some mixing of the two channels at some time after the input.

Diffusion with the concert hall programs is a function of time. The first outputs after an impulsive input are not very diffuse to simulate reflections from the stage floor and walls. The second set of reflections are more diffuse, simulating bounces from walls and the ceiling. The last group simulates rear wall reflections, and is even more diffuse. It takes about 200 ms for the last group of outputs to occur, which gives the impression of a large hall. The relative levels of the three time groups are controlled by the depth control. Adjusting the depth control moves the apparent position of the listener forward and backward in the hall.

The large concert hall program has a much slower final rate of decay than initial rate of decay, and very low initial diffusion. It is good on orchestral music, or in synthesizing a cathedral for chorus or organ. The small concert hall algorithm is more diffuse and more uniform, and is better for music which contains some percussion.

It is not currently possible to *exactly* simulate a given real-world hall with the 224. However, most of the characteristic sound of a real hall is determined by the shape of the reverb time with frequency. This property *can* be closely matched to a real hall with the 224's two reverb time controls, crossover control, and treble decay control.

Although Lexicon has built up both the necessary software tools and the experience to write new programs, they are continually reminded of how difficult it is. They feel their reverberation programs are at the current state of the art, which is why they keep them proprietary. They do, however, welcome any input from users on new programs and on possible improvements to the old ones. □□□

Mobile Fidelity + JVC = Original Masters

doing something about

The Record Pressing Problem

by

Tom Lubin

For some time now, there has been an ever increasing outcry for better and higher quality record pressings. The appeals come from almost everyone concerned; the artists, manufacturers, and the consumers. There has been much written on ways to improve individual aspects of the pressing chain, but, unfortunately, many of these articles have been on a rather esoteric plane. Reality seems to dictate that no matter how much care is taken with each step along the way, some seemingly insignificant stumbling block occurs to foul things up. Sometimes the problem is technical and at other times human error, but more often than not it's a quantity versus quality problem.

Many people in the business have given up the battle, feeling there is not much they can do. Those brave souls who continue to push for quality find that in order to get better pressings they often must battle with the label, the pressing plant, or both. In the final analysis, it always seems to get back to cost, and just how much quality is worth. At every point along the line, volume is being increased by shaving off a minute here or a second there.

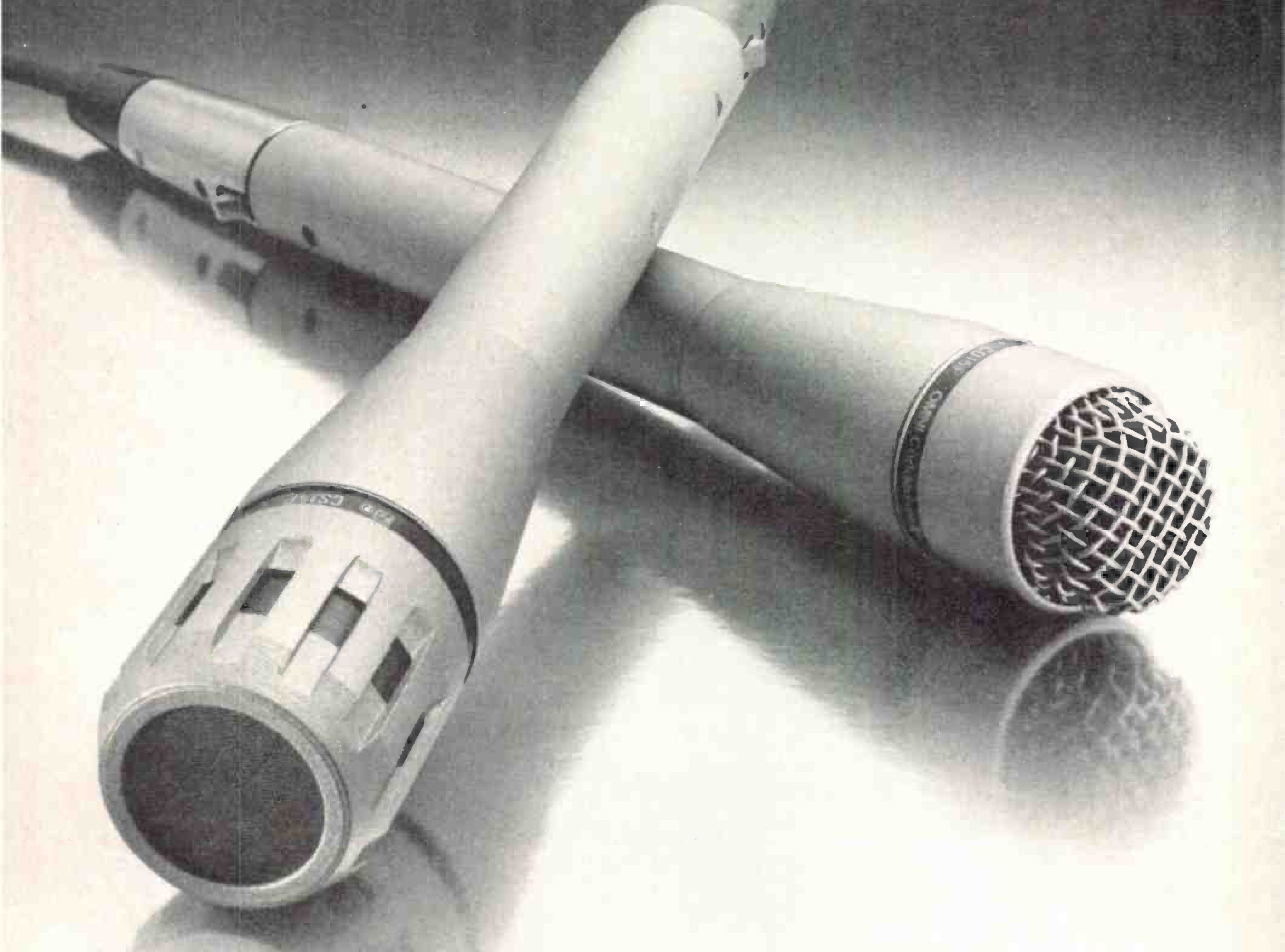
The few people delegated to handle quality control try hard, but seemingly are not able to keep up with the millions of units that must be on the loading docks every day. It all adds up to why records sound and look the way that they do.

Artists and producers have expanded their control to encompass every aspect from creation to mastering, but, alas, the product is usually pressed in some far-off place. Those who were involved in the production are left to wonder what type of pressing will return from areas such as Saugus, Terre Haute, Santa Maria, or some other plant seldom visited by those who create the music. Even when the test pressings are ordered they fall under a veil of suspicion. It's very possible that it won't sound like either the master tape or the record that reaches the store. Test pressings are usually made on whatever press is available at the time the foreman schedules the day's run of LPs. However, when the stamper is moved to another press the results might be quite different. Another consideration is that the test pressing is only a check on just one A/B set of the stampers. Unless a test can be obtained from each stamper there is no real check to determine how the majority of the end product will sound. Re-orders fall into the same category since test pressings are never sent for approval once the product has been released. Often, the artist, engineer, and producer will accept test pressings that are, at time, less than perfect since there is a deadline to meet. Fussing with quality control always means a delay.

The creators of records have expressed their concern; the loudest voice is yet to be heard. More and more consumers are complaining about the poor pressing. They are paying more for a product that, in many cases, seems to be getting worse. This may or may not be so. However, the consumer's perception of the problem is definitely keener now than in years past. This is due to the record buying public's progression from small portable record players to very expensive, high quality stereo reproduction systems. They have been buying them at an incredible rate as indicated by the mass media merchandising of expensive components. The technology available to the consumer has been in a race with that available to the record creators. Between the people who make records and the people who play them there has emerged a competition for excellence. Unfortunately, the manufacturing that bridges the gap between the two has not developed at the same rate. Pressing technology in this country is about the same as it was ten years ago, and the gap seems to be widening.

continued overleaf . . .

Studio quality microphones that don't need a studio to survive.



The CS15P condenser cardioid microphone is equally at home in a recording environment or broadcast studio. When hand-held it puts sex appeal in a voice with its bass-boosting proximity effect. With shaped high-frequency response and its ability to handle high sound pressure levels (140 dB with 1% THD at 1kHz), the CS15P is ideal for close-up vocal or solo instrument miking applications.

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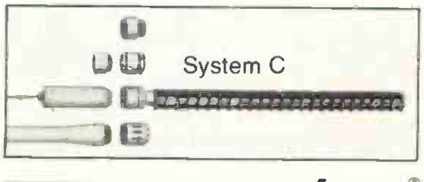
Electro-Voice backs up these two microphones with the only unconditional warranty in the business: for two years we will replace or repair your CS15P or CO15P microphone, when returned to Electro-Voice for service, at no charge – no matter what caused the damage!

We can do this because we build these microphones to meet our standards for performance, ruggedness and durability. We accept nothing less, and if you're a professional, buying a professional quality microphone, you shouldn't either.



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Comparison of JVC and DOMESTIC Pressings From Photographs Taken Using a Scanning Electron Microscope

William Little, professor of physics at Stanford University analyzed with a Scanning Electron Microscope one sample of a conventional pressing and one of the Original Masters. Since the sample was limited to these two disks, the observations and conclusions that were arrived at can hardly be considered conclusive. Nonetheless, some very imporant differences were noted between the two samples.

Dr. Little pointed out the defects in the two and the type of distortions that would result. No conclusions were drawn as to what caused the abnormalities.

The Preparation Of The Sample

In order to use the SEM, the surface of the samples must be made conductive. This is done by a "sputtering" process which plates the non-conductive vinyl surface with one millionth of an inch of gold-palladium alloy. This preparation plating requires that the samples first be mounted on little posts. They are then placed in the sputtering chamber. The sputtering occurs as a result of a glow discharge between an anode and a cathode with the sample between the two. The cathode above the sample is covered with the gold alloy. Once the sample enters the chamber a vacuum is created around it and the two electrodes. Argon gas is then bled into the apparatus and when the pressure is right a high voltage is applied across the two electrodes and a discharge results. This causes the Argon ions to bombard the cathode with such force that they knock off minute bits of the gold alloy. This microscopic fall-out lands uniformly on the samples creating the required conductive film. This procedure does generate a certain amount of heat. From experience they know that this temperature stays below 200 degrees F, which is acceptable for almost all types of substances being analyzed, i.e.: bugs, butterflies, previously living tissue, etc. A major difference between the two records was noted during this preparation procedure.

First Observations

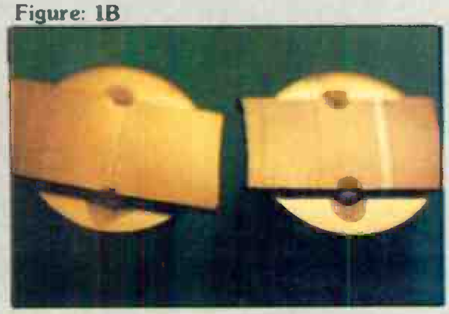
The two samples shown in Figure 1A were given equal amounts of continuous voltage (three minutes). The domestic pressing on the right could not stand the heat and began to decompose. The JVC disk was not significantly affected. Specific melting temperatures for the vinyl are difficult to arrive at, since there is not a fine line as to where melting begins. However, a conclusion that is easy to reach is that the domestic record is significantly more susceptable to heat-related distortions.

Another set of samples were prepared and placed in the sputtering apparatus, but this time the high voltage was applied in bursts of fifteen seconds with a minute cool down

continued overleaf . . .

Figure: 1A

Figure: 1B



Questions

There are many questions that must be asked: What can be done to improve the situation? Why are pressings from Japan and Europe superior to ours? How long will it be before the consumer starts demanding better quality to match the ever increasing price? If higher price means better quality, will the public be willing to pay for it? What part of the system needs the most improvement, or is there a need for an entire revamping of the manufacturing process from mastering to selling? Does the entire industry need the same type of far reaching changes that have occurred in recording studio operations and home stereo systems? Are the complaints of the consumer, producers, engineers and artists falling on the deaf ears (no pun intended) of the record company executives?

Since there are no easy answers, the problems mount and compound — becoming very complex very fast. It's unlikely that the near future will see an average non-audiophile American pressing that is comparable in quality to those available overseas. However, in the last couple of years a few American companies have emerged with a viable alternative to the run-of-the-mill product. The first being the re-discovery of direct-to-disk and most recently, digital recording. Direct-to-disk and digital records are most often sold exclusively through audiophile distributors who handle nothing else but these high quality pressings. With few exceptions, program content of these records is not what you would consider in the mainstream of commercial music. Most of it is classical. Direct-to-disk has the limitations of having to record everything at once and in real time, and current technological restrictions and monetary considerations have minimized the amount of production that can be undertaken with digital recording. As digital becomes more flexible, and the cost of using it comes down its current limitation might be nullified, but once again this will have little effect on the actual pressing. Generally, audiophile quality records have not worked their way into conventional record sales, nor has the more popular broad-based product been available on high quality pressings, until now.

An Alternative

Mobile Fidelity Productions is an organization which for some years has been producing and manufacturing what would be considered a strictly audiophile record. Their product is of the highest technical quality, but the material was generally that of sound effects or products such as the Mystic Moods Orchestra. Last year the company, under its Mobile Fidelity Sound

continued overleaf . . .

SCANNING ELECTRON MICROSCOPE

period between each energizing. The sum total time of voltage application was still three minutes. Using this method both samples easily withstood the preparation process (Figure 1B).

The SEM

Once the samples are prepared they are mounted in the Scanning Electron Microscope. An electron beam scans across the surface of each sample. When the electrons strike the conductive surface a shower of secondary electrons are thrown out. This is a result of localized surface heating caused by the beam striking and entering the alloy. The heat agitates a minuscule shower of outgoing electrons which are directly proportional to the surface composition. When measured these electrons form an extremely accurate representational image of the surface.

Though the beam continues on through to the vinyl, the microscope only "sees" the much harder gold-palladium plating. However, the sputtering process gives a very accurate faithful reproduction of the surface that forms its base. Its accuracy is within a few hundred angstroms (which is very, very good).

Second Observation

Taking a look at the microscope pictures, the small pebbles on the surface of Figures 2 and 3 are dust particles. This despite a cleaning procedure more thorough than any normally used at home. The light areas are the groove walls. The darker stripes show the land inbetween.

Figures 4 and 5 are the same


Figure 2: 200X

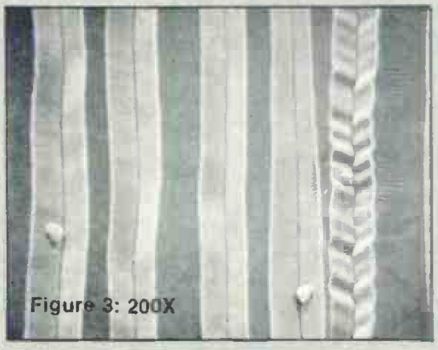


Figure 3: 200X


Figure 4: 500X


Figure 5: 500X

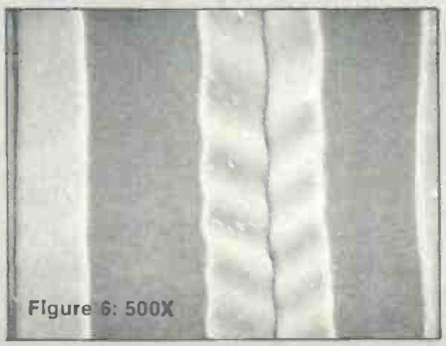


Figure 6: 500X

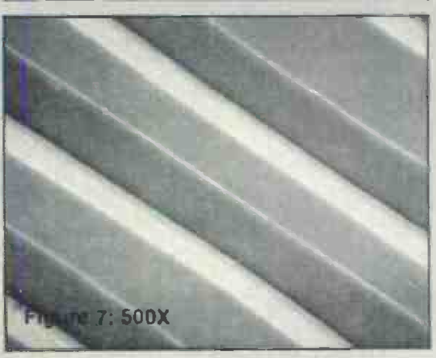


Figure 7: 500X

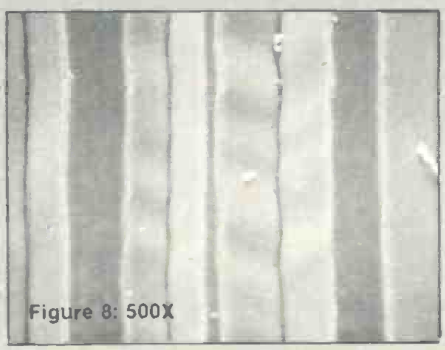


Figure 8: 500X

Lab division, started to negotiate with major labels for the rights to release audiophile versions of previously released successful product. The company's president, Brad Miller, was interested in a product that not only had a sales track record but was technically superior in recording technique. All Mobile Fidelity Records are mastered at half-speed in the U.S. and pressed in Japan, and sell for $15, with retail discounting strongly discouraged. Who buys them? Brad Miller says, "The same guy who's just gone out and paid a thousand bucks for a new stereo." Mobile Fidelity sells direct to audiophile retailers and is in the enviable position of having more orders for the "limited edition" product than they can deliver.

Just what makes them so good? It's not one single thing, but the step-by-step care in manufacturing that literally starts with the master tape. When **"Original Masters"** makes a deal with a label for a particular title, their first job is to get the first generation master. Miller commented, "Most of the problem in the beginning is educating the label on just what we're trying to do. That we want no more degradation of the program. We don't want an EQd copy, or a leveled master, we want the original. Sometimes just finding the original may be a major task. After a record has been released for a while, the whereabouts of the master is occasionally lost in the shuffle of current product. If we have to, and if there's a multitrack available, we'll re-mix it."

Half-Speed Mastering

Once the original master has been obtained the disk mastering can begin. All mastering is done half-speed at the JVC Center in Hollywood, by Stan Ricker. Half-speed mastering has received quite a lot of interest in the last couple of years, but like direct-to-disk it's not new, only rediscovered. Decca of England used half-speed cutting in the early days of stereo to achieve the overall high performance of their recordings. In 1968, when the Neumann SX-68 became available, the practice was discontinued. RCA occasionally used the process when a particular recording had an exceptional amount of sibilants. Most companies never used half-speed since real cutting time was acceptable for the quality of reproduction at that time. The added time as well as increased cost generally made half-speed cutting completely out of the question.

The time required is substantially more than twice the normal running time, since the only way the addition of mastering EQ can be checked is by cutting a test run and playing it back at normal speed. Half-speed cutting requires much more trial and error to determine the proper mastering EQ than does real time cutting.

The original CD-4 half-speed disks were

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(continued)


Figure 9: 2000X


Figure 10: 2000X

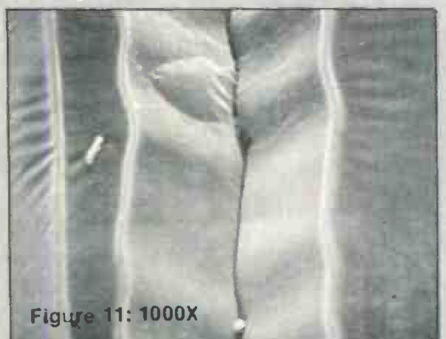
Figure 11: 1000X


Figure 12: 2000X


Figure 13: 2000X

magnification but 4 is of a silent groove and 5 illustrates modulation. Figures 6 and 7 do the same thing. Figures 13 and 9 show a non-modulated groove wall on each of the two samples.

The first blemish to be discovered was a puckering at one isolated spot on the groove surface of the JVC disk, see Figure 3. Figures 11 and 12 amplify this flaw. Aside from this rounded bump, the surface of the JVC disk is very good. The domestic record, however, had blemishes that were not apparent or as individually large. They were, however, much more dramatic in their collective implication.

With each amplification of the domestic pressing a small, continuously periodic, non-regular flaw became visible. It occurred with every modulation on both groove walls. It does not exist on the non-modulated groove walls. Figure 10 shows that the deformity is a ridge rather than a trench. This ridge is not only very small, but has very sharply defined edges, or sides. When the stylus hits this ridge it is bounced up and over the defect. The length of time that it takes this to occur is very short, and would correspond to a frequency wavelength seemingly high above the audible range. However, this is not so. These spikes actually contribute a substantial amount of broadband noise.

Let's say it takes the sylus 1/20,000 of a second to pass one of these ridges and a certain amount of energy is generated by the movement. Since the spikes are frequent but irregular in height, length, and their distance from one to another the energy generated by the stylus movement will be of a broad frequency band. To visualize this it would be very much like running a stick along a fence that has pickets of varying size and spacing. As the stick runs along the top it would have a high frequency component from one stick to another, but would also have a low frequency one as it followed the height changes of the pickets. Since the height of the ridge is not that much smaller than the actual modulation, the signal-to-noise is pretty significant. The JVC disk shows no signs of this surface noise.

SEM PHOTOS

The SEM photos were taken to Stan Ricker at the JVC cutting center to ask him his opinion as to what might have caused the blemishes shown on the pictures of the two samples. These are his observations and possible causes for the defects.

criticized for not having enough bottom. Apparently this was not a problem of the cutting system, but a function of the CD-4 process which didn't care much for low end. Hence, the records were often rolled-off.

The JVC modified Neumann SAL/SX-74 cutting system is flat down to 7 Hz, and down 3 dB at 4.5 Hz. When the record is played real time that would be 14 Hz and 9 Hz respectively.

Half-speed playback is an important benefit of half-speed mastering. In the October, 1978 issue of *R-e/p (Magnetic Reproducer Equalization Accuracy, pages 84-101)* Pete Butt described playback head ringing that occurs above the audible range when audio high frequency transients are reproduced. Half-speed playback effectively slows down all the transients. The frequency spectrum is also moved downward and away from those frequencies that aggravate the resonance peak. The presence of a resonance ring will not be apparent at the output of the tape machine because of all the electronic bypass and filtering circuits in the playback amplifier. The playback head generates it and the first stage of amplification reproduces it. The output level of the ringing can be just as loud or louder than the lower audio bandwidth. That being the case, the first stage amp may be clipping without regard to the audible signal.

Slowing the tape down does not effect the low frequency head bump (contour effect) associated with the playback head since this phenomenon is a function of the reproducer head and shell geometry and its relationship to the physical distance on the tape of one wavelength, and has nothing to do with the time it takes a given point to go past the gap. If the bump normally occurs at 30 Hz, then at half-speed it will be 15 Hz. Recording speed, not playback speed, becomes the determining factor. The heads themselves are very close tolerance glass ferrite, hence the parallel edges of the gap hold their shape with a minimum of fraying and tearing along the parallel pole pieces. Pictures showing a comparison between conventional heads and ferrite ones are part of an article by Walter Scott in the April, 1976 issue of *R-e/p (Magnetic Tape Heads: Ferrite vs. Metal Tape Heads, pages 26-33).*

The JVC system is transformerless with the exception of those transformers associated with the cutter feedback circuit. The output transformers have been replaced with opamps with similar gain and impedance. The benefits of such an audio chain is minimal phase shift and improved high frequency transient response or rise time. Low frequency roll-off associated with transformers is also eliminated.

Phase shift is also lessened by the by-pass of the acceleration limiters from the cutting amplifier. This circuit normally introduces a small amount of delay into the high

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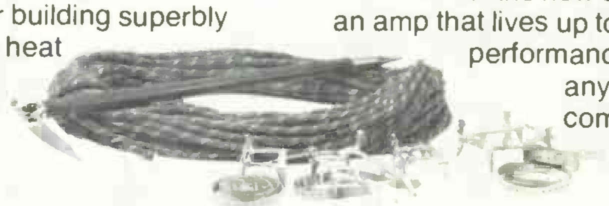

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(continued)

Both records under the greatest magnification exhibited fine striations of the surface of the groove wall. These are the result of the cutting tool surface not being absolutely and completely smooth. These sort of scratches are so small that they can be generated by a brand new acceptable stylus. It's not at all uncommon to find variation among stylus manufacturers as to the smoothness of their surface polishing. If the scratches are bad enough, obviously the cutting tool is shot.

The Domestic Pressing

The domestic disk has "Noise Modulation." This is caused when the leading edge of the stylus cutting surface tears away at the groove wall. This would indicate the cutting tool used was worn out, further that its tip radius to begin with was quite possibly larger in diameter than the JVC stylus. The JVC stylus has a one micron tip radius with a burnishing facet (or cutting edge so to speak) of one micron. (25.5 microns = one mil) A standard speed stylus straight from the manufacturer may have a tip radius and burnishing facet dimension of between 2 to 4.5 microns, or about four times as fat as the JVC cutting tool. Therefore, the JVC stylus seems to cut a more accurate groove, since the larger the stylus the greater the corresponding loss of high frequency resolution.

The drawback to the JVC stylus is that it wears out quickly. For this reason the stylus is changed at least every other side. The abrasiveness of the lacquer substance can quickly wear away any stylus.

To place some sort of perspective on the wear and tear that a stylus sees, a brand new tool that executes a perfectly acceptable outside diameter test groove will have laterally traveled more than half a mile before it reaches the inside lead out groove. Add to this the amount of horizontal motion caused by vigorous audio modulation, and you'll end up with a total of considerable distance. Plus, all of this rubbing force is being exerted at what is essentially a single point.

Most people aren't going to hear the noise graphically shown in Figures 2, 5 and 10 since they're not going to play the record on an expensive reproduction system of high resolution. Under the pressure of production this type of defect might slip by. It obviously did in this case.

The JVC Defects

There are more possibilities as to what has caused these blemishes. Photos 11 and 12 appear to show a gas bubble which is just under the surface of what's been cut. The bubble has not succeeded in breaking through. Had that happened, the result would have been a huge pop when the record was played. As it is, it's only momentarily audible. What we have here is smooth metal. Had it exploded, it would have looked like a moon crater with rough edges.

In the same vicinity of this blemish, there are also surface ripples or "pickling." Photo 3 shows this occurred on the land area adjacent to the more obvious groove imperfection, as well as near the "bubble" and extending into the next groove. To speculate on the origin of this occurrence one might suggest a problem in the electro-plating process, but this is just a suggestion. Stan commented that he hadn't seen a blemish like this before except when a lacquer has been exposed to MEK or some other form of alcohol that would attack the surface. It is possible that this might have occurred due to the separation of the silver from the lacquer in the pre-plating stage.

In order to make the lacquer surface conductive, a silver-nitrate/sugar water spray is evenly distributed one molecule thick over the surface of the disk. It is possible that this silver layer came up off the surface of the disk. This might have been caused by some outgassing of the lacquer during plating. This could cause the puckering or pickling. There is nothing in the pressing operation that could have caused this phenomenon to occur.

Another possibility might be that not all the lacquer's surface oil had been removed before the silver was sprayed. An indication of this is that the wrinkling is more prevalent in the land area.

If the normally vigorous pre-plating degreasing is not complete, the silvering will not adhere quite as well as it should. The water based silver spray contains a wetting agent as it is essential that the lacquer surface tension not be broken. An example would be an oily plate of glass that has been sprayed with water. The liquid will have a tendency to bead up. Anything that makes this occur is death for the plating process.

These are only two possibilities as to the nature of these flaws. To make an unequivocal statement would require further and extensive investigation.

frequency program while the logic circuit determines which of 25 toroidal windings the signal will pass through. These coils are a source of additional phase shift in the audio. Their function is to protect the cutter head from any high speed, high frequency transient. Half-speed cutting makes the circuit unnecessary as the cutter easily handles the slower rise times of half-speed transients.

All the amplifiers are also able to operate well below their rated levels and function with improved performance. This is due to a number of factors. The first has to do with the RIAA pre-emphasis curve. Consider that at 10 kHz the high frequency boost is 13.7 dB above the 1 kHz reference level. Since the power required is commensurate with the frequencies being cut, lowering the frequency by an octave effectively results in the cutter needing 3 dB less power. Secondly, when a tape is played back its energy is distributed over its full length. Altering the speed of the tape does not change the total amount of energy. It does, however, change the amount of energy per unit of time. Again, the output of the amplifiers can run lower than normally required. By taking twice as long only one-fourth of the power is used. When the 600-watt per channel Neumann SAL-74 cutter amp is used half-speed their power is equivalent to 2,400 watts per channel in real time.

Half-speed cutting requires that closer scrutiny be given to cutting lathe wow. If small speed variations occur at 16-2/3 ips the frequency of occurrance will double when the disk is played at normal speed. In order to eliminate motor wow and vibration a Neumann has a considerable amount of de-coupling as a part of the turntable drive train. There is a slight amount of unavoidable play in the system. The motors commonly used in this lathe maintain a constant speed and torque. Hence, a continuous and steady pressure is exerted against the dampening linkage. Conventional motor wow is eliminated but non-periodic wow can still occur. If for an instant the stylus cuts a particularly deep groove the table will slow down for a split second. The motor maintains its speed since the slow down is absorbed in the drive linkage. The JVC cutting lathe uses a quartz lock DC drive motor that is controlled by a logic circuit that assesses the rotational speed of the turntable almost 5,000 times a second. Motor torque is instantly varied to compensate for the speed change that results from deeply cut grooves.

Better cutter feedback control is also achieved with half-speed cutting. Feedback at the cutters resonance frequency (about 1 kHz) is in excess of 35 dB, but as the audio frequency increases the amount of feedback control decreases. At 8 kHz there is 5.5 dB and only .6 dB at 15 kHz.

Additionally, if the RIAA curve is taken into consideration the problem of insufficient feedback at the higher frequencies is compounded.

New Cutting Tool

The JVC cutting stylus was developed to accommodate the CD-4. Existing cutting tools were unable to meet the challenge. A new stylus was designed to handle the very short wavelengths of the 30 kHz high frequency carrier. A micron is one millionth of a millimeter. A conventional stylus with a diameter of 4 to 5 microns has a tendency to get in its own way while cutting the physically short wavelengths. The JVC tool has a tip radius of one micron with two micron burnishing facets. The relief angle of the stylus is 35 degrees rather than the conventional 45 degrees. The JVC stylus is also shorter, 1.17 mm as compared to 2.1 mm for a normal shaft length. This reduces crosstalk because less mass is accelerated and less whipping action occurs at the stylus tip. The improved stylus design has moved the high frequency limits upward by 3 dB, especially at the inside diameters as compared to a standard stereo cutting tool at the same level and diameter.

Stan Ricker, JVC's mastering engineer, stated another point, "I've seen mastering engineers 'try' to get just one more side out of a stylus before they replace it. In the entire scheme of things the cost of replacing the tool is pretty insignificant. If there's any question of its integrity it should be changed." He added, "It's damned hard to predict how long one will last. I've had brand new stylus go bad by the time they reach the end of the first side they cut. Typically, I change it every two LP sides."

Another advantage is the elimination of the folding to center of loud passages which real time cutting has occasion to do. Ricker continued, "There's a lot of labels starting to use us, mostly at the insistence of the artist or producer. Mobile Fidelity Sound Labs' records are cut this way with one variation. They don't try to fill all the available surface. Mobile Fidelity records use only as much surface as necessary. If a side ends with a larger diameter than normal — all the better. Most labels will reject a lacquer that's not filled up, supposedly on behalf of the consumers who might complain the record isn't long enough or defective. Poppycock!" Stan Ricker gave the analogy that at the largest diameters the quality of the cut is like recording at 30 ips, and in the inside diameters are comparable to 7½ ips on tape.

Lacquers

The quality of the lacquers used in mastering has always been a problem. The mastering engineer does have a selection between the three major suppliers: Transco, Audiodisc, and Pyral. The brand that's preferred comes down to a personal taste and experience. The practical reality of getting quick delivery of an adequate and regular supply becomes a consideration in deciding what blank is used. Many mastering facilities don't carry a large inventory. They know how many blanks they use on the average and they keep up with that. If they suddenly have an avalanche of orders, or a new shipment of blanks contains an unusually high percentage that are unacceptable, they're going to need an immediate shipment of more blanks. Quality normally is defined in terms of good, better, or best; and occasionally and unavoidably falls into a fourth category of "What's Available." Lacquer surface problems range from surface ripple to the subtle changes caused by the slight alterations of the emulsion formula. The surface is critical and by and large both the lacquer manufacturers and the mastering engineers are pretty diligent in their surface analysis. A disk with the slightest surface blemish will be rejected by most mastering engineers. Every now and then a blank will have a problem not readily apparent on the surface, such as microscopic bubbles inside the emulsion that are exposed only after the record is cut. The pressing that would be made from such a master would have small pin holes in the bottom and sides of its grooves.

Curing

Once the Mobile Fidelity disks have been cut, they're allowed to aerate for half a day while the freshly cut lacquer surface degasses. Then they're refrigerated for three days so the surface can firm up before the long flight to Japan. Stan comments, "They're chilled as cold as you'd want a six pack of beer."

The lacquer surface is a bit like jello and exhibits an elastic nature. When a record is cut not only is a thread of the surface removed, but the groove walls are pushed aside. If it's left for any length of time these compressed groove walls will relax, hence the reason for getting a lacquer plated as soon as possible. Chilling the master retards this relaxing, whereas heat accelerates the deformation. Many good lacquers are ruined by being allowed to sit in a hot truck or on a sun-warmed loading dock for a few hours. Stan Ricker continues, "I don't like to send anything out on a Friday, but they often [large companies] insist."

After chilling, the disks are put in cases and taken straight to the airport. When they reach Japan, JVC picks them up and takes them straight to the plant. The JVC pressing plant utilizes technology that rivals the best in the world.

All employess must pass through air filter locks. All the plating and pressing operations are done in "clean room" atmospheres. The plating as well as the pressing is automatically controlled. One of the problems with plating and pressing operations in this country is the lack of strict environment controls. If a speck of dirt contaminates the surface of any of the plating processes or gets pressed into the vinyl, it ends up being a click or pop. Plating tank temperatures are strictly controlled. Because heat is generated by the plating process, if the heat is not dissipated the temperature of the plating bath will increase quickly. Stan Ricker noted that "For anyone who might never have been in a plating plant, they are usually hot as hell. And that doesn't do the lacquers much good." Speed of the plating process will also have an effect on the final results. Naturally, it's cheaper to do if it's done fast. The problem with fast is that it's hard to determine when it's too fast.

The Plating Process

Briefly decribing the process: the lacquers are sprayed with deionized water and then with a fine layer of silver, one molecule thick. The silverplated master has its center hole bolted to an anode and then is placed inside the plating tank. When the voltage is turned on, the free nickel ions which are suspended in the bath are attracted to the conductive silver surface. Each ion subsequently bonds itself to the ones that preceded it. Once the plating is a few molecules thick the process can be speeded up, but a slower speed is a necessity in the initial plating. If it's done too fast the required higher voltage will burn the thin silver layer. The result will be the introduction of clicks and pops to the newly plated matrix.

This tends to be most apparent at the inside diameters where the anode current is most concentrated. Besides putting little holes in the surface, the heat generated by the burning silver also causes the grooves of the temperature-sensitive lacquer to physically distort. At this stage the plating would conform to the newly created distortion. Once the lacquer/matrix process is completed temperature becomes less of a factor, but the speed of nickel build-up is still very much a problem. A good analogy can be drawn from photography. Film and prints that are processed quickly tend to be much grainier than those developed slowly. With plating a similar situation occurs. The faster the plating happens the larger the nickel grain. The larger grain size also means the space between each grain is greater. Put another way, the density of the nickel plating and the fineness of the grain is inversely proportional to the plating speed. The advantage of a fine grain is pretty obvious since detail is improved as the grain gets smaller. However, density as a quality consideration is not readily apparent. The crunch comes (no pun intended) when the stamper is put into operation. The extreme pressure exerted on the stamper will have a tendency

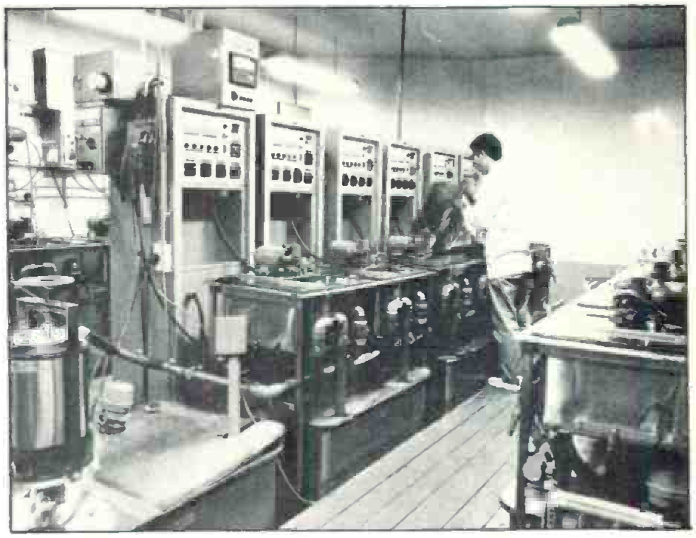
JVC

Left: **Clean-Room Air Lock Entrance to Plating Room**

Below: **Plating Room**

Bottom Left: **Plating Quality Control**

Bottom Right: **Pressing Room**



to collapse those areas that are less dense than others. A high-density stamper and close observation of the pressings is the only prevention of this problem. It's not at all uncommon to find records that have imperfections that are bumps in the vinyl surface. Often these are due to the stamper surface collapsing.

Dips on the other hand may come from small dirt particles lodged between the stamper plate and the mold that it's mounted in. If a bit of debris is left behind when the stamper is mounted the pressure on the metal surface will be great enough to distort the nickel and cause it to mold itself around the speck of dirt, hence a bump is put on the surface of the stamper.

Some years ago, Steve Allen observed that almost everything in the world contains "little black things" of unknown origin. If any part of the plating, stamping, mastering, or the vinyl itself is contaminated by these microscopic specs, the quality of the end result is effected.

JVC plates their parts slowly and does not "de-horn." Most plating operations take the mothers and lightly, ever so slightly, burnish the nickel surface. The term used to describe this is de-horning. De-horning makes the subsequent plating separations easier to accomplish since any little spikes or hooks that might catch in the process are removed. Opinions vary as to whether or not this effects the final results. JVC and Mobile Fidelity Sound apparently feel that it does and, therefore, they don't follow this practice.

After the matrix has been separated from the lacquer its surface is chemically treated to encourage a highly controlled oxidization that only effects the first molecular layer of the matrix surface. It's this thin film of oxidization that allows the subsequent plated parts to be separated from one another and not permanently bonded. JVC handles each step as a separate process to eliminate any cross-contamination. However, each step is carried out in close succession. The process is continuous. The various parts are never sitting collecting microscopic debris as might be the case if plating were done in large "runs." For an Original Master two mothers are made from each matrix. Each mother produces 10 to 20 stampers.

Vinyl

JVC calls their vinyl "Super Vinyl." The name isn't particularly imaginative, but the compound does have some interesting properties. It has a much higher melting point than conventional vinyl. (Actually it doesn't really melt, it just slowly softens to a putty-like consistency at about 300°.) This harder vinyl is less susceptible to heat related warping caused in shipping, packaging and storage. Another plus for the high definition vinyl is that it exhibits less memory than American pressings tend to do. This type of distortion is caused when a record is played more than once in close succession. The groove walls of the softer, conventional vinyl deform under the extreme instantaneous pressure of the playback stylus. If the record is not played immediately, its walls will slowly relax to their original form. JVC records can be played repeatedly without this problem occurring.

The JVC pressing rooms are completely automated yet closely supervised. There is one technician for every five presses. JVC normally presses approximately 5,000 copies per stamper. However, the Original Masters are limited to 2,000 pressings per stamper. One out of every 50 to 100 pressings is checked for quality.

Labels

When a record is pressed the blob of vinyl isn't the only thing put into the press. A label also goes in. Obviously, the label is not part of the playing surface but it nonetheless can be a factor in the quality of the final product. The JVC labels are printed on a different type of paper than their American counterparts. The ink and the actual paper are able to withstand the required higher pressing temperature. The ink is much more colorfast than conventional labels, and will not bleed into the vinyl surface. During the hot pressing cycle the ink used in this country is said to occasionally contaminate the vinyl.

Another consideration involving the label is a type of stress distortion called "Saddle Warp." Few companies take this problem into consideration. If the paper grain of both labels is not running in the same direction when they're embedded in the vinyl, the natural curl of the paper will put unequal stress on the center surfaces. When the record is first removed from the press and is still fairly flexible this label stress can warp the record. JVC aligns the label paper grain to minimum "saddle warp." Once again, it's the little things that count.

Shipping and Packaging

After the records are pressed their edges are trimmed and then they're put into static-free sleeves. These specially made sleeves contribute quite a lot to maintaining the integrity of the product *after* the consumer has bought it. There is a significant absence of clicks and pops due to the static charge that often happens when a record is removed or slipped into its sleeve. The sleeves are made of fairly heavy translucent plastic. One side of the sleeve is plastic, while the other side is made of a sealed pocket inside of which is a sheet of paper. Apparently the sealed sheet of paper eliminates the static charge normally associated with conventional plastic sleeves. In the past neither paper or plastic has been totaly acceptable as sleeve material. The paper ones are always mildly abrasive. After a record has been slipped in and out of its sleeve a few times hair fine scratches start to appear on the record surface. Plastic sleeves won't cause abrasions but static charges do occur. The combination of the paper and plastic has eliminated the problems associated with the individual materials while combining their benefits.

The sleeved records are shipped from Japan in specially constructed cartons to insure that no edge pressure is placed on the records. Edge pressure in shipping is a major contributor to the warp problem. If records are exposed to any kind of heat during shipping they will re-form or permanently bend in an attempt to reduce the edge pressure stress.

Customs

It takes about four weeks for the records to make their journey by ship from Japan. Once they reach L.A. they come in contact with the most perilous part of their journey, the U.S. Customs Service. The duty rate is established at approximately six per cent. The question is six per cent of what. Six per cent of what is paid to JVC for the actual pressing? Six per cent of the estimated artist royalties? Six per cent of the studio sessions involved? Six per cent of what?

Brad Miller seems to feel that of all the details and procedures which are required, the Customs issue is the most confounding. He says, "You're never quite sure how much they're going to want. Everytime a new shipment arrives they interpret the law differently. It just depends on what agent handles it because the codes, regulations, and rulings are so confusing and antiquated. You find contradictory Customs law everywhere you turn, and they're always right until you go into court and prove them wrong. We ended up posting a really high bond so we could continue to bring the records into this country. It's a real pain."

After Customs the records are mated with their jackets which are printed in this country. Again, the quality of the printing is closely supervised. The package also includes a cardboard inner sleeve and a quality control notice which explains conditions which might be misinterpreted as pressing defects when in fact they're turntable and tracking problems. After all other possibilities have been explored, Mobile Fidelity will request that the record be returned to them with the completed defect slip noting the nature of the flaw. Mobile Fidelity does not want the bad record returned to the retailer. When there's a bad record they want to know what's wrong with it. A new record is always sent back to the consumer. When a record is sent in as defective and it turns out to have no problem, along with the new record is enclosed another note describing other possible playback misalignments. Quality control for Mobile Fidelity involves the education of the buyer as much as constant supervision of their suppliers.

In Closing

The records are sealed in loose plastic bags rather than the traditional shrink wrap. Shrink wrap, though quick, easily automated and cosmetically neat, has a couple of real nasty detractions. In order to get shrink wrap to shrink it's necessary to heat it with the record inside. Hopefully, the vinyl record won't be effected by the heat, but the possibility can't be ignored. If the shrink wrap machine isn't running exactly as it should be the record is likely to have problems. Secondly, overtight shrink wrap will tend to pinch the record's sides at the four points where the record touches the sides of the jacket and also pull up the corners. Loose plastic bagging isn't terribly neat, but it puts no strain on the record since the record is exposed to no heat or side stress. Heat is only used to seal shut the open-ended bag. Once again, if the record is put under any type of stress, as soon as it's exposed to the slightest heat the record will try to adjust to the stress, and warp.

Original Masters are currently only available in a few select audiophile shops and selected record retailers. No returns are allowed from the stores. Their artist roster now includes Fleetwood Mac, George Benson, Steely Dan and many other prominent acts. After listening to one of their records it becomes clear that with patience, tenacity, and a bit more money spent on manufacturing, the quality of the record **can** come close to matching that of the tape. □□□

New Products

TEAC 85-16 16-TRACK UNIT

The TEAC Tascam Series new 85-16 recorder/reproducer is a 16-track unit offering 16 full channels of dbx.

The unit has a suggested list of around $10,500 and will be available for September delivery nationally, according to Bill Mohrhoff, TEAC/Tascam national sales manager.

The model's rugged transport system is controlled by three servo motors and is designed for easy tape threading with flutterless roller. The take-up and supply reels are controlled by sensors in the left and right tension arms. The transport is equipped with motion control and end-of-tape is detected by a photosensor located between the capstan and the head housing. The 85-16 also has a zero search function and a four-digit display indicates tape position and velocity in percentage. The cue lever has a built-in tape lift defeat.

Mohrhoff said depressing the stop button during the stop mode converts the transport to the editor mode, allowing greater ease in manual editing. Another editing convenience, Mohrhoff pointed out, is depressing the fast forward or rewind buttons twice which appreciably slows the tape speed in each mode.

The unit's record speed is 15 ips and it takes 10½-inch reels.

The 85-16 contains a single board plug-in amplifier with record/reproduce/bias circuits. Both record and reproduce amps are DC with FET input elements. Trim pots on the PCB are mounted for easy front or side adjusting. Sync performance is the same as for reproduce.

The transport mounting angle in the console cabinet can be changed in three steps. Maintenance on the transport underside is made easy because the transport can be tilted open to 100 degrees. Mohrhoff said the 85-16 also allows tandem operation of two transports. Sync between transport and VTR, or film recorder, or other machine, is possible.

Specifications of the 85-16 are impressive. Overall frequency response (in sync or repro modes) is 40 to 18,000 Hz, ±3 dB. Bias and erase frequency is 150 kHz. Signal-to-noise ratio is 67 dB WTD, 62 dB, UNWTD. Wow and flutter, measured on flutter test tape, is ±0.04% peak weighted (ANSI) and 0.03% RMS weighted (NAB). Record electronics headroom is 28 dB or greater above operating level at 1 kHz. Total harmonic distortion is less than 0.1% at 1 kHz at 25 dB above operating level. Cross talk is better than 45 dB at 1 kHz.

The 85-16 has a number of optional features, including a remote control unit. Mohrhoff said most functions of the 85-16 have remote control capability. The optional AQ-85 automatic cue unit is a search cue with six memory points.

TEAC CORPORATION OF AMERICA
7733 TELEGRAPH ROAD
MONTEBELLO, CA 90640
(213) 726-0303

for additional information circle no. 58

ELECTRO-VOICE INTRODUCES SHOCK-MOUNTED CARDIOID MICROPHONE

A new shock-mounted super-cardioid microphone, the RE18, was introduced by Greg Silsby, Professional Products Sales Manager at Electro-Voice.

for additional information circle no. 59

ACOUSTICOMPUTER
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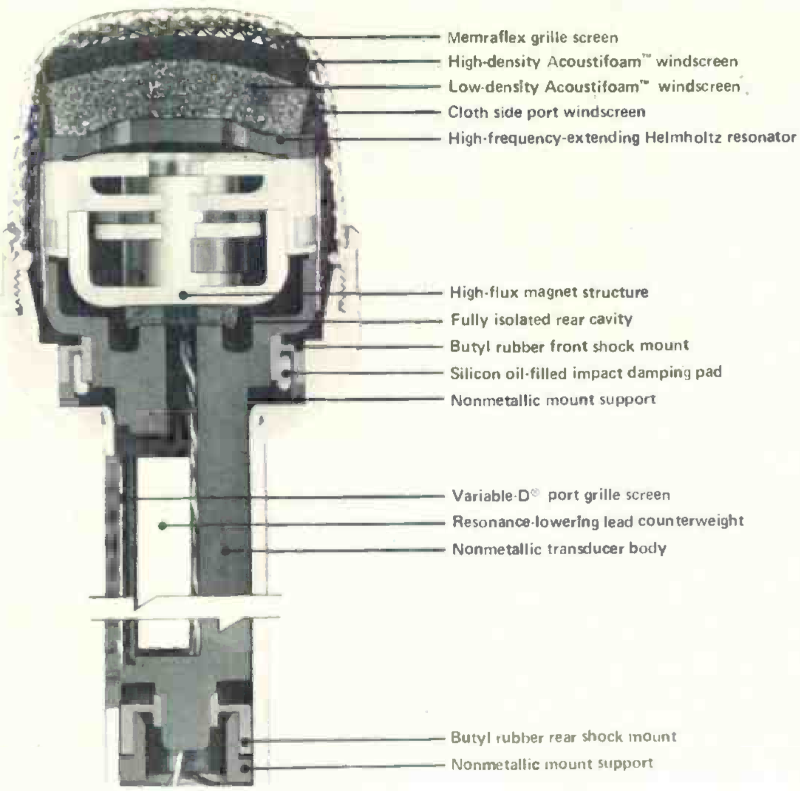
EFFECT YOUR IMAGINATION!



Although primarily intended for hand-held broadcast applications, the RE18, is equally at home in any situation where ambient noise rejection and isolation from handling noise is a consideration. "A good example," says Silsby, "is the mechanical stand and lectern noise commonly encountered in sound reinforcement systems. The RE18 effectively silences these annoying sounds."

Silsby notes that the RE18 has a great heritage, "The RE18 maintains the superb frequency response and super-cardioid polar pattern of the famous RE15 microphone while having an integral blast filter for "P-pop" protection as does the equally famous RE16." The Variable-D® design of the RE18 has the added advantage of maintaining its frequency response regardless of mike-to-talent working distance. "Frequency response is also maintained if the talent happens to get a little off-axis," adds Silsby, "With these additional advantages, it would not surprise me to see the RE18 showing up in recording studios or other music environments."

The RE18 carries a suggested retail price of $226.00.

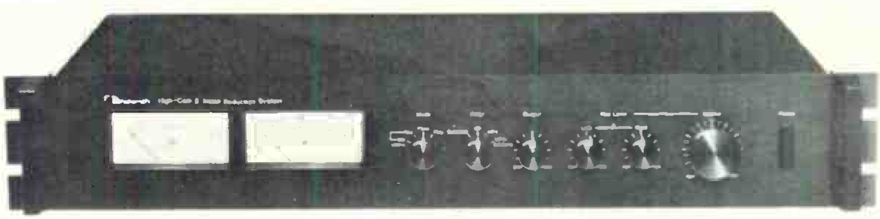
ELECTRO-VOICE, INC.
600 CECIL STREET
BUCHANAN, MI 49107
(616) 695-6831

for additional information circle no. 61

NAKAMICHI AND TELEFUNKEN JOIN FORCES TO PRODUCE A HIGH-QUALITY CASSETTE NOISE REDUCTION SYSTEM

"Hi-ComII" is the official designation for an all-new noise reduction system offered by Nakamichi U.S.A. Corp. The system is being made available as an "outboard processor" which can be used with any tape deck, although the circuitry has been optimized for high-quality cassette decks. Hi-Com II is a truly international effort as it represents the talents of Nakamichi and Telefunken, the latter of West Germany. The circuitry makes use of Telefunken's recently introduced HighCom compander IC.

Hi-Com II is so-named because it processes the signal in two frequency



for additional information circle no. 62

"Once you get your hands on this machine . . . you'll see what we mean."

PERFORMANCE:
Overall Signal-to-Noise: 66 dB unweighted at 520 nWb/m (30 Hz to 18 kHz audio filter).

Playback Signal-to-Noise (electronics): 72 dB unweighted (with audio filter).

Headroom: +24 dB. Maximum Output: +28 dBm.

Overall Frequency Response (15 ips): 30 Hz to 22 kHz ±2 dB.

Playback Frequency Response (MRL test tape): 31.5 Hz to 20 kHz ±2 dB.

RELIABILITY: An unmatched four-year track record of on the job performance for the original compact professional recorder. Day in, night out. Just ask someone you trust.

ALIGNABILITY: Any tape recorder must be aligned to achieve maximum performance. With the MX-5050-B, all primary alignments are on the front panel. So is a 1-kHz test oscillator. Secondary alignments are inside the bottom panel. You or your maintenance people can align it fast and easy. This saves you time, money, and enhances your reputation.

INTERFACEABILITY: With a flick of the output switch you can plug-in to any system: +4 dBm 600 ohm or −10 dB high impedance. No line amps or pads to mess with. A perfect match everytime.

ADDITIONAL BENEFITS: Three speeds, dc servo ±7%, ¼ track reproduce, full edit capability, over-dubbing, noise free inserts, XLR connectors, NAB/CCIR switching, unique three-position alignment level switch.

PRICE: Suggested retail price $1,945 (USA).

MX-5050-B: The best value in a professional tape recorder.

OTARI®

Call Ruth Pruett Ables on 415/592-8311 for the name of your nearest Otari professional dealer.
Otari Corporation, 1559 Industrial Road,
San Carlos, CA 94070 TWX 910-376-4890
In Canada: BSR (Canada, Ltd.),
P.O. Box 7003 Station B, Rexdale, Ontario M9V 4B3
416/675-2425

for additional information circle no. 63

bands. Both bands use a 2:1 compression/expansion ratio to achieve a 20 dB improvement in the cassette deck's dynamic range. The use of a relatively high ratio and the use of a separate companders for the low and high frequencies results in virtual elimination of noise modulation (pumping), which plagues most simple compander circuits particularly when applied to a medium with inherently limited dynamic range, such as the cassette.

The HighCom IC is characterized by high transient signal accuracy. By its nature, it is a much "faster" network than any other compander circuit of the past. It further gives the system designer a wide range of attack and release times from which to choose. In the Hi-Com II, each frequency band can thus be optimized in terms of dynamic characteristics. This makes the Hi-Com II system one of the most accurate noise reduction systems ever produced. It is especially free from coloration with transient signals, a vulnerable point with many noise reduction systems.

Realizing that a noise reduction system at best is a very carefully chosen set of compromises, especially if cost is to be a factor, Nakamichi and Telefunken have subjected the system to a wide variety of tests. Special emphasis was laid on critical listening tests with a large number of varying program sources. As a result, the Hi-Com II system has been finely tuned to meet the particular demands of cassette noise reduction. Nakamichi recommends that the Hi-Com II outboard processor be used with state-of-the-art equipment and the finest quality cassette tape.

The Nakamichi Hi-Com II Noise Reduction System will carry a suggested retail price of $420.

NAKAMICHI U.S.A. CORP.
1101 COLORADO AVENUE
SANTA MONICA, CA 90401
(213) 451-5901

for additional information circle no. 64

PARASOUND ANNOUNCES SYNTON ELECTRONICS DISTRIBUTION

Parasound, Inc., announces its appointment as the exclusive worldwide distributor (except Europe) for Synton Electronics of Holland. Highlights in the existing Synton line are the popular and reasonably priced Syntovox 221 and 222 Vocoders, which have garnered considerable acclaim in Europe since their introduction a year ago. This will be the first introduction of the Synton products in the United States and other foreign markets.

The Syntovox 221 Vocoder is a 20-channel analyzer, synthesizer, and control system which allows the user to create a wide range of vocal effects for records, commercials, film or theater. The analyzer takes an input signal (most often a voice) and breaks it up via 20 bandpass filters. The levels at each of the filter frequencies are then converted into control voltages which impose the speech characteristics onto an input signal, either an internal pulse generator or an external input (any instrument with plenty of harmonics, such as synthesizer, organ, guitar).

Also involved is a voiced/unvoiced detector, which determines whether syllables in the voice input are voiced (where they can be reproduced with the input signal alone) or, as in the case of sibilants, they are unvoiced, where a burst of white noise is required to make them sound more intelligible. To work most efectively, the "excitation signal" (the sound that is modulated by the voice input) must have plenty of harmonics, so that the range of the excitation signal is greater than that of the input signal.

Connections between the analyzer and synthesizer sections are made via a 20 x 20 matrix panel, which allows many alterations of timbre to be made. An internal pulse generator is included for speech synthesis applications.

One of the most distinguishing features of the Syntovox 221 Vocoder is its extreme intelligibility. The 221 uses 54 dB/octave filters which allow the output to retain high intelligibility relative to the original input. The 221 comes in a standard 19" rack mount for easy studio installation.

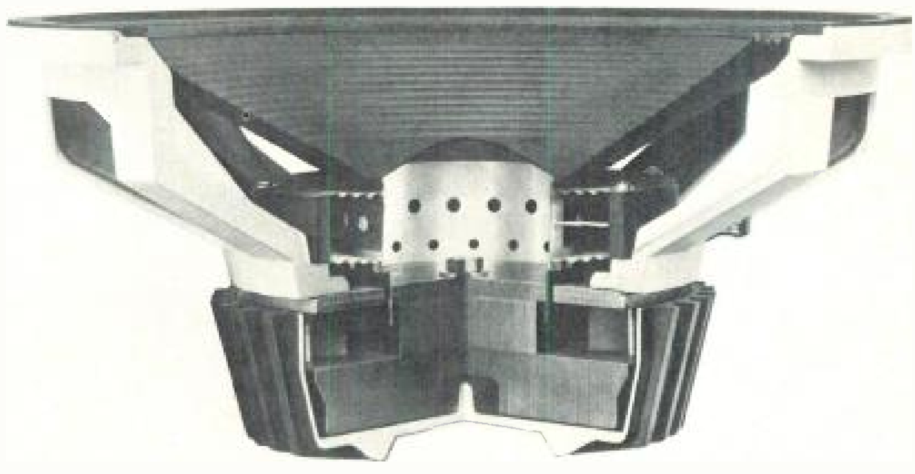
PARASOUND, INC.
680 BEACH STREET, SUITE 414
SAN FRANCISCO, CA 94109
(415) 673-4544

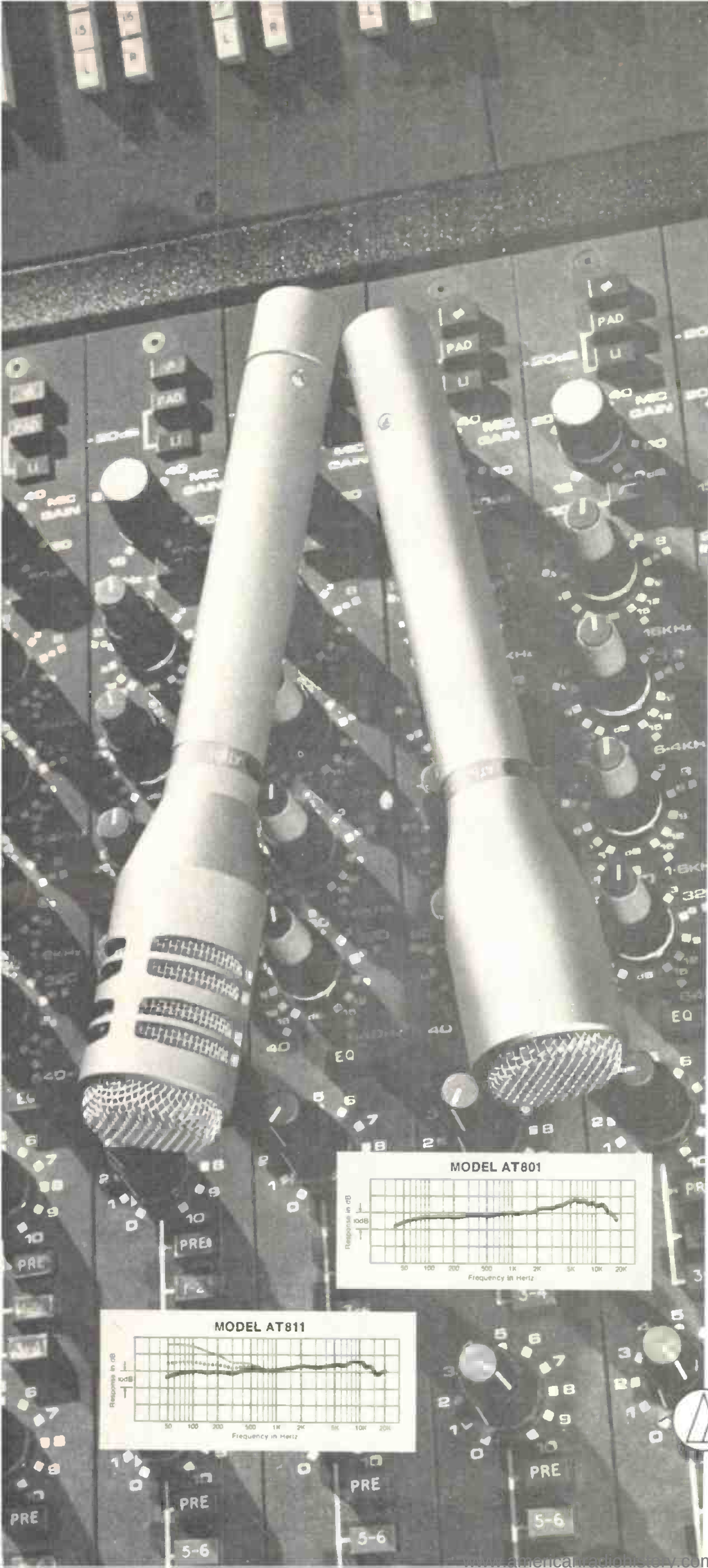
for additional information circle no. 65

GAUSS DOUBLES LOUDSPEAKER POWER RATING BY IMPROVED DESIGN

In keeping with Cetec Gauss' state-of-the-art research and development, the Gauss line of loudspeakers has doubled loudspeaker output at no additional cost to the consumer.

All of the 12", 15" and 18"loudspeakers have been upgraded to handle twice the power (RMS watts) of the existing units. The new power ratings will be 300 watts RMS for lead guitar types and 400 watts RMS for bass and low frequency units. All of the other parameters, i.e., sensitivity, free air resonance, impedance, size of the voice coil and magnet gap, have been maintained. Even though twice the SPL can be obtained from the new line of 12s, 15s and 18s, Gauss' reliability and interchangeability of new re-





The Audio-Technica philosophy:

EQ should be used to improve the sound... not to fix the mike!

Introducing affordable smooth sound. The remarkable AT801 and AT811 Electret Condensers. With curves so smooth you would have to pay a bundle to match them anywhere else.

Response like this has a number of benefits. First, your EQ is used only to touch up the sound, not to correct built-in errors of the microphone. Which leaves more leeway to control the overall sound.

And without unwanted peaks you have more usable headroom. That's vital when you're working near the dynamic limit of a preamp or line amp. Sound stays clean and sharp. Compressors or limiters sound less forced, because they are controlling peaks in the *sound,* not peaks in the *mike!*

But perhaps the biggest advantage is the versatility of these A-T condensers. Because they have just the right amount of presence for today's recordings, they're not limited to just one kind of instrument... just one type of voice. Put them *anywhere* in the mix: brass, reeds, percussion, chorus, or strings. Then listen. What you hear in the studio you'll hear at the console. Which is a great place to start in miking any session.

At well under $100 these are two of the best bargains you'll find these days. Reliable, clean-sounding, and the most predictable microphones you can use. Make them a mainstay in your studio today. Write for spec sheets and dealer list.

Model AT801 Omnidirectional Electret Condenser

Model AT811 Unidirectional Electret Condenser

Great sound, right from the start!

audio-technica.
INNOVATION □ PRECISION □ INTEGRITY

AUDIO-TECHNICA U.S., INC., Dept. 79RE
33 Shiawassee Avenue, Fairlawn, Ohio 44313

cone kits with existing units have been maintained.

Gauss states that several design innovations have been employed in order to achieve a doubling of the power handling capacity. They are as follows:

— Winding the voice coil directly on the voice coil support, thereby improving voice coil roundness and uniformity.

— Using anodized aluminum for the voice coil support to assist in dissipating the additional heat generated by the increased power.

— Providing additional breathing holes in the anodized aluminum voice coil support to aid in convection cooling the voice coil.

— Utilizing a 20 lbs. (9.1 kg) magnet assembly with a high energy ceramic magnet weighing 4.75 lbs. (2.2 kg).

— Specially designed magnet assembly, cast bottom plate and die cast finned aluminum structure providing additional area for heat dissipation.

The overall height and weight of the new units is unchanged from the existing models. And, because of the finned cover, the appearance is the same as well.

These design changes were all possible by the success of the 10" loudspeaker introduced two years ago, the 10" being the forerunner of the newly designed 12s, 15s and 18s.

The 10" unit was used to perfect the direct wind process and the ceramic magnet assembly. As a result, Cetec Gauss claims the 10" unit has the greatest sensitivity and power handling capacity of any 10" loudspeaker on the market.

CETEC-GAUSS
13035 SATICOY STREET
NORTH HOLLYWOOD, CA 91605
(213) 875-1900

for additional information circle no. 67

NEW CONSOLES FROM RUSLANG

Ruslang Corporation is offering two new tape transport consoles, the RL 600 and the RL 700. These new models complete the Ruslang tape transport console line when

added to the RL 500, a previously announced console which can handle 19" x 15¾" tape decks.

The RL console line incorporates the industry's most modern design features, including front panel access in both horizontal and vertical positions, plus a rear shelf for storing power supplies. The entire line is built with the same care, style and quality of materials that the company builds into Scully consoles. Ruslang has been a major supplier of Scully consoles for 15 years.

The RL 600 is designed for any of the new style 19" tape decks and can handle any width up to 21". The RL 700 also accepts any standard 19" tape deck but can accommodate widths up to 24½". One of Ruslang's three consoles will accept all popular tape decks, including models made by Ampex, MCI, Tascam and Otari.

RUSLANG CORPORATION
247 ASH STREET
BRIDGEPORT, CT 06605
(203) 384-1266

for additional information circle no. 68

TYPE 85 FET DIRECT BOX

The Type 85 is an active direct box employing a low noise FET amplifier to reduce loading effects by 200 times and reduce distortion by 10 times when compared with a direct box of conventional

for additional information circle no. 69

design. It is housed in a single piece extruded housing that will withstand over a ton of crushing force. Power is provided from a standard 9 volt battery or from 48 volt phantom power through a built in DC to DC converter that preserves complete ground isolation.

The Type 85 is ideally suited for interconnecting a guitar or other high impedance instrument pickup with a low impedance balanced microphone input. Bass guitarists will especially enjoy the Type 85's extended low frequency response and complete freedom from loading effects. Its extremely wide frequency response, low distortion and low noise will enable it to be used where other direct boxes prove unsatisfactory.

Power Requirements: 48 volt phantom power or 9 volt battery. Input Impedance: 10 Meg Ohms. Frequency Response: 20 Hz to 20 kHz ±.5 *dB*. Noise Referred to Input: 2.0 uV max 50 Hz to 15 kHz. Total Harmonic Distortion: .05% at 1 kHz and 1 VPP input. Dimensions: 1¾" x 3" x 5".

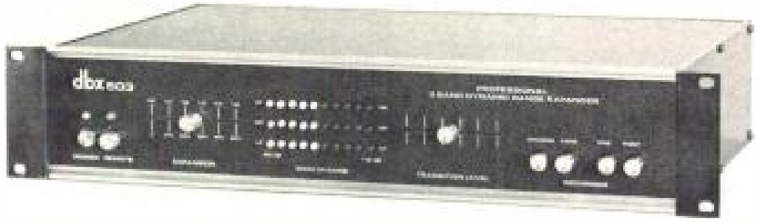
COUNTRYMAN ASSOCIATES
424 STANFORD AVENUE
REDWOOD CITY, CA 94063
(415) 364-9988

for additional information circle no. 70

dbx DISCO SOUND ENHANCEMENT

The dbx Model 503 is a three band dynamic range expander particularly suited for disco sound application. It is often necessary to expand dynamic range in order to get the most out of a record, because of the many forms of dynamic range limitations imposed during recording. Vocals, kick drums, bass guitars, guitars, violins, and practically everything else are individually subjected to compression and/or limiting during either recording or mixing. When the tape is cut to disk, it is often further subjected to compression, in order to transfer the music to record at as hot a level as possible, and not cause skipping of the final product even when played on inexpensive equipment.

The result of all this dynamic range limiting is a record which has, perhaps, 20 dB of useful dynamic range. That is, the ratio of loudest to softest music in the record might be 20 dB or less, even though the original performance of the vocal alone might have had 30 or 40 dB of dynamic range. To recover some of the excitement of the original, live performance, the 503 dynamic range expander will make high level sounds even higher in level, while making low level sounds lower.



The 503 operates independently within three separate frequency bands, so expanding the cymbals won't get in the way of the bass, and vice versa. The effect of a 503 has to be heard to be believed; kick drums and bass guitar become punchier, vocals stand out against the music, highs take on new sparkle.

Furthermore, the 503 allows the DJ control over how much expansion he wishes to use. If a record with more dynamic range than 20 dB is to be played, the expander can be adjusted to less than full expansion, as appropriate to the situation.

dbx, Incorporated
71 CHAPEL STREET
NEWTON, MA 02195
(617) 964-3210

for additional information circle no. 72

NEW AKG PORTABLE REVERBERATION UNIT

AKG Acoustics has announced that it has begun shipping the new "E2" version of the BX-10 to dealers. The "E2" has been significantly improved over the original BX-10E.

Generally, the BX-10E2 is audibly smoother and cleaner than the original BX-10E, which is already widely acknowledged as a superior reverb for a broad range of applications. This improvement in sound is due to a re-design of the Torsional Transmission Line system plus addition of equalization to the TTL system electronics.

Throughout the world, the BX-10 has found enthusiastic acceptance in music recording studios, road sound reinforcement systems, film production facilities, jingle studios and broadcast facilities. For speech applications (e.g., AM and FM radio) there is a special short-decay time version available: the BX-10E2-Short. On-air advantages of the BX-10E2-Short are gaining increased recognition at major U.S. radio stations. On special order only — cost of the BX-10E2-Short is 5% over the standard BX-10E2.

The RM-10 accessory rack mount fits all versions of the BX-10 reverberation unit, and provides a sturdy and attractive method for mounting the reverbs in a standard 19" equipment rack with all controls fully accessible.

AKG ACOUSTICS
PHILIPS AUDIO VIDEO SYSTEMS
91 MCKEE DRIVE
MAHWAH, NJ 07430
(201) 529-3800

for additional information circle no. 73

SANSUI E-1 PROFESSIONAL DISC PREAMPLIFIER DEBUTED

"For precision monitoring during disc

mastering or in broadcast production, for PA applications or an advanced audiophile system, or for laboratory A-B comparison of different pick-ups, the new professional series Sansui E-1 Disc Preamplifier offers unparalleled flexibility and quality," according to Mr. Tom Yoda, vice president of marketing and sales for Sansui.

Up to 3 moving magnet or 3 moving coil cartridges, or 3 line-input level sources can be switch-selected with the Sansui E-1. Input impedances for MC cartridges are provided at 10, 33, 100 and 330 ohms, and input sensitivity at these inputs is rated at 0.1 mV. Equivalent input noise of the MC inputs is rated at -150 dBV (IHF, A-weighted). For moving magnet cartridges the resistance at the input can be switched to 22 k-ohms, 33 k-ohms, 47 k-ohms, or 100 k-ohms, and input capacitance can be separately switched to 220, 330, or 470 pF. Signal-to-noise ratio (IHF, A-weighted) for MM cartridges is 90 dB, with a 110 dB signal-to-noise for the 200 mV, 50 k-ohm Line inputs. Phono deviation from the RIAA curve is within ±0.2 dB from 20 to 20,000 Hz, and total harmonic distortion is less than 0.003% (Line in; 0.008% MC).

A calibrated, master attenuator, plus separate channel level controls, is provided, along with additional line/record switching and level setting controls. Separate MC and MM high-frequency adjustments are included for each channel, covering +6, -3 dB at 10 kHz; +3, -7 dB at 20 kHz. In addition, separate, switched MC and MM subsonic filters with a 12 Hz turnover and 6 dB/octave attenuation are included, to help eliminate the effects of warped LPs. Rack mountable, the E-1 requires an EIA-standard 3½" panel height.

"The Sansui E-1 Disc Preamplifier utilizes entirely Direct-Coupled, DC circuits and has a 300 V/uSec. slew rate," Yoda noted. "Its low output impedance — 75 ohms or less — is capable of delivering up to +28 dBm into a 600-ohm load. The provision of three separate outputs, one at a +4 dBm level and the other two at -10 dBm, makes the E-1 ideal for professional applications, while still 'at home' in an advanced audiophile system."

SANSUI ELECTRONICS CORP.
1250 VALLEY BROOK AVENUE
LYNDHURST, NJ 07071
(201) 460-9710

for additional information circle no. 75

NEW BGW MODEL 600 PROFESSIONAL POWER AMPLIFIER

BGW Systems, the Hawthorne, California, manufacturer has announced introduction of their new Model 600 Professional Power Amplifier.

The BGW 600 is the first in a series of economical, basic power amplifiers from BGW. Incorporating the quality and ruggedness found in the well-known line, a few appearance and convenience features have been deleted.

The company claims the continuous 175 watt per channel stereo amp offers THD of no more than 0.1% (8 ohms, 20 Hz - 20 kHz). At 40 ohms there's 250 watts with no more than .15% THD, and from a convenient rear panel switch, the mono mode yields 500 watts, .15% THD, (both load impedances measured 20 Hz - 20 kHz, 250 milliwatts to rated output).

The 600 retains these BGW features: Modular design with massive heat sink assemblies; redundant output stages utilizing ten 150 watt complementary transistors each; sophisticated loss-of-feedback clipping indicators; high speed 15 MHz Op-amp front end; independent front panel gain controls; separate signal and chassis grounds and rugged, heavy-gauge steel construction.

Hum and noise is better than 106 dB below rated output into 8 ohms (unweighted, 20 Hz - 20 kHz) and frequency response is +0, -.25 dB, 20 Hz - 20 kHz.

BGW SYSTEMS, INC.
13130 S. YUKON AVENUE
HAWTHORNE, CA 90250
(213) 973-8090

for additional information circle no. 77

KEN SCHAFFER GROUP INTRODUCES NEW B&T WIRELESS

The Ken Schaffer Group has announced the availability of its long awaited, low cost wireless instrument and microphone systems. The Schaffer B&T, as the unit will be called, is set to retail at $2,150, versus the $3,450 price tag on the company's well known Schaffer-Vega Diversity System, the first wireless ever developed.

Essentially, the Schaffer B&T has identical performance capabilities as its more expensive counterpart, with one exception: musicians must allow a few minutes for a sound check to insure the absence of dead spots by selecting optimum antenna placement. Features like a signal-to-noise ratio of better than 90 dB, crystal controlled stability, interference proof performance, and no tuning are common to both system.

Schaffer-Vega Systems are currently used by over 200 of the world's top bands, including the Rolling Stones, Kiss, ELO, Earth, Wind & Fire, and Foreigner. The Schaffer B&T allows the wonders of wireless to now be made available to a whole new market of musicians and groups.

Dealers will find the B&T an excellent promotional product for their own stores, by virtue of its wide range of audience grabbing capabilities.

THE KEN SCHAFFER GROUP
10 EAST 49TH STREET
NEW YORK, NY 10017
(212) 371-2335

for additional information circle no. 78

MEYER SOUND LABS SUB-WOOFER SYSTEMS

The Meyer Sound Laboratories Sub-Woofer Systems were created to reproduce the 25 Hz to 100 Hz region of the audio spectrum accurately at high sound levels. Applications include recording studios,

motion picture, video soundtrack and sound effects monitoring, small movie theaters, and ultra high quality sound reinforcement.

With this complete system, a new area of possibilities opens up. Information below 100 Hz may now be controlled; as it is now possible to monitor these frequencies they may be added or subtracted, used to enhance music or sound effects. It is no longer necessary to disregard low frequency information and simply roll off the low end as a matter of course.

This low frequency information is there on many tapes and records. The mediums are capable of reproducing low bass. With the coming of the new digital recording techniques, all audio system capabilities will dramatically increase. The recording medium will not be limited — the limiting factor will be the speaker systems. This system is designed to eliminate the remaining weak link.

400 ES Stereo Sub-woofer Supplemental System consists of two speaker cabinets and a specially designed two channel power amplifier. The initial (original) signal source is fed directly into the power amp which contains the conditioning circuitry that processes the signal information below 100 Hz. Information about 100 Hz is supplied, via the built-in crossover, to the existing systems power amps or electronic crossover at line level.

The 400 ES will produce sound pressure levels of up to 8 acoustic watts continuously from 30 Hz to 100 Hz. It is a seventh order system.

MEYER SOUND LABORATORIES
1440 SAN PABLO AVENUE
BERKELEY, CA 94702
(415) 527-7700

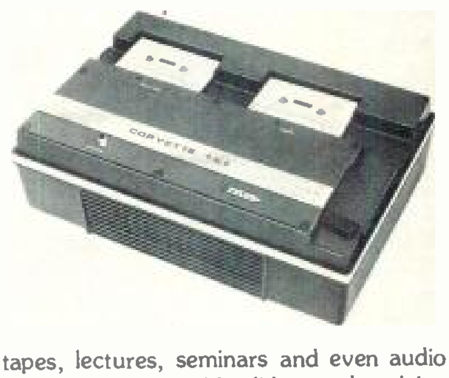
for additional information circle no. 79

NEW CASSETTE COPIER BREAKS PRICE BARRIER

A totally new design for a high speed cassette copier has been announced by Telex Communications Inc. The new Copyette 1&1 carries a suggested retail price of $449.00.

According to Telex this represents a significant price breakthrough and is less than half the cost of most copiers available today. A company spokesman said the low cost reflects the market potential the company anticipates for this product. It is a simple, almost completely automatic copier that reportedly does the job very well, providing the quality customers have come to expect from Telex.

Telex compares the Copyette 1&1 to an office paper copier in that it copies one cassette at a time but at high speed. A standard one-hour C-60 cassette is copied in under two minutes and is then automatically rewound, ready to be used. It is primarily intended for the copying of the user's teaching, training or orientation



tapes, lectures, seminars and even audio visual programs with slide synchronizing cue tones. The unit is as simple to operate as an office paper copier so it can be used by anyone without any technical training.

Housed in a portable case, the Copyette 1&1 weighs under 12 lbs. and is said to be the most compact copier on the market.

Telex also manufactures audio visual cassette recorders, language laboratory systems and headsets which are distributed throughout the U.S. and in over sixty foreign countries.

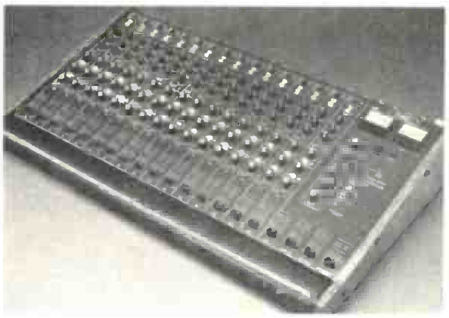
TELEX COMMUNICATIONS
9600 ALDRICH AVENUE SOUTH
MINNEAPOLIS, MN 55420
(612) 884-4051

for additional information circle no. 80

AUDIO MIXING CONSOLE FEATURES INDEPENDENT MONO AND STEREO OUTPUTS

An audio mixing console with independent mono and stereo outputs for live performance and recording applications is being introduced by Audy Instruments of Salem, Massachusetts.

The Audy Series 2000 Mixing Console produces simultaneous mono and stereo outputs that allow mono and stereo formats to operate independently. It also provides separate monitor and effects sends. Available in 12 or 16 channels (stackable up to 32), the console utilizes high speed, low noise IC technology to reduce TIM distortion to 0.03% and enhance sound quality.



Providing input preamps with a dual LED system, the Audy Series 2000 Mixing Console maintains headroom at 24 dB throughout. Other standard features include: individual channel and output patch points; transformerless balanced inputs and outputs; 3 band EQ with switchable midrange; switchable pre- and post-monitor and effects sends; soloing for any input or output; phantom power; and work lamp socket.

The Audy Series 2000 Mixing Console is priced at approximately $2,450 for 12 channels, and $2,995 for 16 channels, including an Anvil flight case for each model.

AUDY INSTRUMENTS
SHETLAND INDUSTRIAL PARK
P. O. BOX 2054
SALEM, MA 01970
(617) 744-5320

for additional information circle no. 81

SHURE ANNOUNCES NEW PHONO CARTRIDGE SERIES FOR PROFESSIONAL APPLICATIONS

Shure Brothers, Inc., Evanston, Illinois, has combined the best of two worlds in a new line of phono cartridges especially designed for broadcasting, recording, disco, and other professional applications.

Called the Shure SC39 Series, all models in the new line are designed to meet professional standards for ruggedness and reliability, yet also provide sound reproduction quality and high trackability that until now has has been available only in top-of-the-line phono cartridges designed for the hi-fi consumer market.

The new line includes three models tailored to specified professional applications. To meet professional requirements, the phono cartridges incorporated several unique engineering designs, which provide true high fidelity performance, resistance to stylus damage, and prolonged record (and lacquer master) playback.

A major new feature of the SC39 Series is resistance to stylus damage, even when subjected to the rigors of slip-cuing, backcuing and the abuse of the stylus brought about by fast-paced studio situations. Each SC39 cartridge has an internal support wire and special elastomer

continued on page 123 . . .

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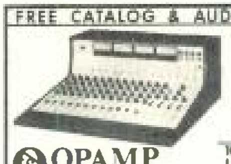

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. . . continued from page 119

bearing that insures stable and accurate backcuing without groove jumping.

To guard against another common form of stylus damage that occurs when a stylus is pushed sideways and bent (i.e., stylus is accidentally bumped against the edge of a record), SC39 models have a unique lateral deflection assemply. Called the Side-Guard, it responds to side thrusts on the stylus by withdrawing the entire stylus shank and tip safely into the stylus housing before it can bend.

Another feature protecting the stylus is an exclusive lever-operated, locking guard that protects the stylus tip against drops, bumps, or accidental mishandling when not in use. In the playing position (when the guard is up), a V-shaped cutout on the control lever provides a highly visible cuing aid for precise groove setdown.

In addition to the record preservation advantages of its high trackability, the new series offers prolonged record and lacquer master playing life by minimizing noise build-up, particularly when using new styli. To protect against this problem, styli in SC39 cartridges feature a unique finishing process that produces a Shure MASAR tip. This new stylus tip results in minimum asperity and abrasion, even on 45 rpm records made from reprocessed or substandard vinyl, and virtually eliminates noise build-up.

The SC39ED cartridge offers an essentially flat frequency response that compares favorably with top quality hi-fi cartridges. The response of the SC39EJ and SC39B is extremely flat (±1 dB) up to 15,000 Hz, with a smooth rolloff up to 20,000 Hz to minimize high-frequency "splatter" that may result from high-frequency pre-emphasis in FM broadcasts.

Models included in the Shure SC39 Series are: (1) SC39ED with a recommended tracking force of .75 to 1.50 grams, bi-radial (elliptical stylus, and user net price of $100.00; (2) SC39EJ with a recommended tracking force of 1.50 to 3.00 grams, bi-radial (elliptical) stylus, and user net price of $80.00; and (3) SC39B with a recommended tracking force of 1.50 to 3.00 grams, spherical stylus, and user net price of $60.00.

SHURE BROTHERS, INC.
222 HARTREY AVENUE
EVANSTON, IL 60204

for additional information circle no. 85

news

dbx® TO PRODUCE ENCODED DISK LIBRARY

dbx, Incorporated, a subsidiary of BSR, Ltd., has launched an ambitious program in cooperation with a number of record companies to produce a sizeable library of **dbx** Encoded Discs.™ The records, which are remastered using **dbx** Type 11 noise reduction, are compatible with standard record playing equipment, although they require a **dbx** disk decoder to yield the ultimate benefits claimed for the system. The "noiseless disks" are said to provide a 50% greater dynamic range than is available on most conventional records in addition to being free of audible surface noise.

To reach the largest possible audience, repertoire is being selected to encompass a wide range of music tastes. Initial releases will feature classical orchestral and instrumental works, but subsequent releases will include light orchestral and film music repertoire as well as the music of popular folk and jazz recording artists. Emphasis is being placed on tape masters with exceptional quality with respect to dynamic range and tape hiss.

The company is executing a multi-phase program geared to release approximately 25 albums, along with the introduction of a new low-cost **dbx** disk encoder, in time for the 1979 fall season. The second phase will follow with 25 additional titles, and approximately 100 select albums are expected to be in the dbx Encoded Disc library within a year after introduction at the 1979 summer CES.

"Substantial benefits will accrue to each record company participating in this venture," stated Jerry Ruzicka, vice president of marketing and sales at dbx. "Being sensitive to the concerns of the record industry which experienced marketing disasters with quadraphonic records a few years ago, we devised a no-risk approach to obtain the support and cooperation of record companies. Their financial investment is minimal and there are no multiple inventory problems. The complexity of license agreements are avoided entirely as the arrangement with the participating record companies involves a straightforward product purchase agreement."

dbx is purchasing the encoded disks from the record companies as a distributor. For the first phase of the program, dbx is absorbing all the remastering costs. They provide each record company with finished metal parts for pressing. The record company need only produce the quantity of records ordered by dbx or, at their option, they can press additional quantities for sales through their normal channels of distribution.

To distinguish the dbx Encoded Discs from conventional record releases, a conspicuous metallic "dbx Encoded Disc" label is being attached to the front of the otherwise conventional jacket under its shrink wrap. The record label will also identify the recording as being dbx encoded and state that it must be played through a dbx disk decoder. A two-sided 11" x 11" sheet, containing information about how dbx Encoded Discs work and how they are produced, is inserted along with the vinyl pressing in each record jacket. For the three series of disks planned for release, dbx has established suggested retail prices of $8.00, $12.00 and $16.00 depending upon the label of issue.

Regardless of how the master tape is made, dbx claims that their encoded disk delivers the best possible sound on record, although the company points out that the music program itself is not changed by the dbx encode/decode process. Various forms of distortion and noise, such as signal overload, tape hiss, room noise, modulation distortion, echo and print-through that may exist on the master tape are not modified.

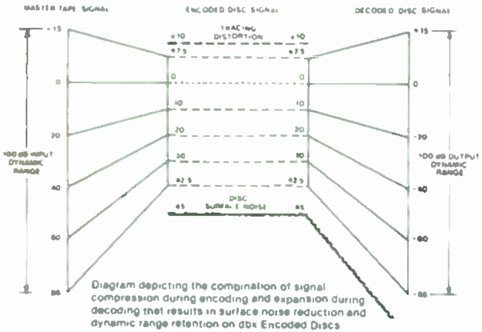


Diagram depicting the combination of signal compression during encoding and expansion during decoding that results in surface noise reduction and dynamic range retention on dbx Encoded Discs

The dbx noise reduction process applied to disks is effective solely in preventing the introduction of additional noise during disk mastering and record playback. The surface noise on dbx Encoded Discs is typically 30 dB lower than on conventional records and, in many cases, they will provide up to a 50% increase in dynamic range (the difference between the loudest and quietest musical passages).

The "whooshing" sound caused by the granular composition of even the best vinyl, and most high-frequency clicks due to inevitable imperfections in the manufacturing process, generally become inaudible in the dbx Encoded Disc. Turntable rumble and groove echo also disappear. Finally, since narrower grooves can be used for the dbx encoded signal, cutting is done further away from the center of the disk, thereby sharply reducing inner groove distortion.

Summarizing his position at the time of the announcement, Ruzicka stated: "The program we have initiated is intended to bring this new disk format to the attention of the public and to encourage record manufacturers to make use of dbx noise reduction technology that is now available to them. As record companies release dbx Encoded Discs in response to increasing consumer demand, the costly inducements we are absorbing will become unnecessary. We will then withdraw from this activity, while continuing R and D on noise reduction and other types of audio equipment intended to increase the enjoyment of recorded music.

"We have seen extraordinary technical advances in the last two decades relative to improving the quality of master tape recording using, for example, Dolby®* or dbx noise reduction and digital recording techniques. Even with the improvements made in phono cartridge design, phono preamp circuitry, vinyl compounds, disk mastering equipment, and direct-to-disk recordings, I believe the dbx Encoded Disc represents the most significant advance in recorded disk technology since the introduction of stereo some 25 years ago. I know that's a strong statement, but I believe it to be true. The benefits of this technological breakthrough can be enjoyed by everyone immediately, since the addition of a dbx Disc Decoder to a stereo system represents an extremely modest investment. [*At this time about $100.00. — Ed.*] While a totally digital sound system theoretically could provide even better performance, there undoubtedly will be a substantial cost penalty for such a system and its widespread availability is probably many years off. In the meantime, we hope that our efforts in creating the dbx Encoded Disc library will eventually lead to the dbx disk encode/decode format being adopted as the record industry standard." □ □ □

*'Dolby' is a registered trademark of Dolby Laboratories, Inc.

NEVE EXPANDS WEST COAST OPERATION

Rupert Neve, Incorporated, is continuing continued

to enjoy a rapidly expanding business on the West Coast and therefore, to give more prompt and efficient service in both Sales and Engineering. B. Morgan Martin has been appointed as Regional Technical Manager based in Los Angeles. He will handle all console commissioning, service, and technical inquiries in the West Coast operation.

Morgan has held senior positions in the audio field, principally in broadcasting, since he graduated from the University of Texas in 1971. Morgan's most recent position was Manager of Technical Operations for Metromedia Television's Metrotape West in Los Angeles.

Morgan can be reached at the Neve Hollywood Office, 6255 Sunset Boulevard, Suite 609, Hollywood, California 90028, or telephone 203-465-0135 which is the new Technical Service Inquiries number.

SOLID STATE LOGIC REPORTS RECORD SALES — $2.6 MILLION IN FIVE MONTHS

British recording console manufacturer Solid State Logic has received more than 2.6 million dollars in orders for the new SI-4000 E Series Master Recording Console and Studio Computer System, which was announced last March at the Brussels AES Convention.

SSL President, Colin Sanders, attributes

State Logic B Series console and computer, earlier this year.

Other studios awaiting the new system are: The Manor, Virgin Records country estate complex in England; Andre Perry's "Le Studio" in Montreal; CBS Studios in Rome; Nidaros Studio, Tronheim, Norway; and the new Splash Studios in Capri, Italy. Solid State Logic "B" Series systems are currently in use at studios in Detroit, England, Denmark, Sweden, and West Germany. According to the company, a number of other major purchases will be announced shortly.

Solid State Logic is represented in North America by Washington Musicworks Inc., headquartered in Washington, D.C.

'CADAC' BACK IN PRODUCTION

The assets of Cadac (London) Limited (in liquidation) have been acquired by C. A. Audio Systems, Ltd., and production of consoles for music recording, broadcast and theater applications is once again in full swing.

An improved version of the recently introduced 'In-Line' series provides full function flexibility for recording and mixdown. As options DC sub-grouping, sophisticated automation and centralized routing are available.

The address is C. A. Audio Systems, Ltd., 141 Lower Luton Road, Harpenden, Hertfordshire, AL5 5EQ, U.K. Telephone:

relatively ample and cheap energy; and last but not least, a climate that reminds Britishers of home.

One of ADR's top research and development engineers from Great Britain, Richard Strang, will be joining Nigel Branwell, vice president/marketing, in Seattle to assist in the expansion and to supervise ADR's 'state-of-the-art' service program.

Audio & Design Recording manufactures a range of sophisticated signal processing equipment, including the world reknowned Vocal Stresser, Compex Limiter, and SCAMP System.

Correspondence should be addressed to: Audio & Design Recording, P. O. Box 786, Bremerton, Washington 98310. The new telephone number is (206) 275-5009.

RECORDING CENTER ENGINEERING CLASSES

The Recording Center, Inc., of 25 Van Zant, East Norwalk, Connecticut 06855, will be resuming classes in recording engineering this fall. Chief engineer Ethan Winer will also provide private and semi-private instruction for the more advanced student or anyone desiring personalized attention.

The Center will also offer audio system planning, set-up and service to studios, industry and education.

Oliver Berliner 3M-Mincom's Marshall Hatfield

3M PRESENTED MAKER OF THE MICROPHONE AWARD FOR DEVELOPMENT OF DIGITAL RECORDING SYSTEM

The Maker of the Microphone Award was recently presented to 3M's Mincom Division for "an outstanding contribution to the world of sound;" specifically, the development of a practical digital audio recording system with electronic editing and disc-mastering interface.

The "Mike" Award is a trophy presented annually in memory of Emile Berliner, who invented the microphone, the disc record, the disc record player ("gramophone") and

continued overleaf . . .

the method of mass-producing discs from a single master record. Some previous recipients of the award include Peter Goldmark Jr., The Bell System, Goddard Lieberson, Dr. Ray Dolby and the National Library of Canada.

SOUND WORKSHOP AND AUDIO MACHINERY DISSOLVE MARKETING PACT

Sound Workshop Professional Audio Products, Inc., and Audio Machinery Corporation, both of Hauppauge, New York, have announced the cancellation of their exclusive agreement regarding the marketing and distribution of the Shared Access Memory System. "It just wan't working out. The arrangement put Sound Workshop in an awkward position," noted Michael Tapes, president of Sound Workshop and a consultant in the development of the Shared Access Memory System. "Audio Machinery did not keep their original commitments as to price and delivery which has caused us to be very vague and elusive with our dealers as well as the potential buyers in the studios.

"The response has been great, and we hate to lose the Shared Access System. This was the first time we marketed a product not of our own manufacture and the inability for us to react to the public demand was tremendously frustrating."

Bob Raifman, vice president of Audio Machinery, informs us that the Shared Access Memory System is now in its final stages of development and new marketing plans are being negotiated and will be announced before the end of the year. "When Shared Access is completed, it will represent a unique approach to digital signal processing, and we feel that whenever it is introduced, it will be accepted on its merits. We did not feel comfortable accelerating its development to yield to market pressure. Because of our different viewpoints on the product's development, we thought it best for both Sound Workshop and Audio Machinery to dissolve our relationship regarding Shared Access Memory."

The dropping of the Shared Access project has allowed Sound Workshop to focus more strongly on its own product line which will include a new major console series which will be announced shortly and introduced at the upcoming AES convention in New York.

KEVIN DAUPHINEE NAMED SALES MANAGER OF NEW SANSUII PROFESSIONAL PRODUCTS DIVISION

Mr. N. Kouchi, President of Sansui Electronics Corporation, has announced the appointment of Kevin Dauphinee as Sales Manager of Sansui's Professional Products Division.

Dauphinee officially began his responsibilities for the Sansui Professional Products Division on January 1, 1979. Headquarters for the division are at Sansui's new facility at 1250 Valley Brook Avenue, Lyndhurst, New Jersey.

Dauphinee had previously been associated with Dolby Labs, first as Recording Studio Applications Engineer and, most recently, as FM Development Manager. He holds a degree in Electrical Engineering from Colorado State University, where, as Communications Technician, he was responsible for coordinating all concert activities on campus.

When interviewed about his new position, Dauphinee said, "I am, of course, very excited about heading up the sales team for Sansui's Professional Products Division. In addition to being greatly flattered by the opportunity of working for a company with so fine a reputation as Sansui, I am also tremendously impressed with the initial product line-up of the division. The B-1 Power Amplifier, E-1 Professional Disc Preamplifier and P-1 Parametric Equalizer are, in my opinion, the finest quality products of their kind available in the professional market today. I am also convinced that, as the product line expands, it will be of the same high quality. After all, Sansui is recognized throughout the world for its technological achievements in the field of music reproduction."

'SPARS' INVITES PROSPECTIVE MEMBERS TO QUESTION/ANSWER SESSION AT NEW YORK AES: NOVEMBER 3

from: **Joe Tarsia**
Sigma Sound Studios
and
Interim President and
Chairman of the Board
of Directors,
The Society of Professional
Audio Recording Studios

Among the professional participants of the audio recording industry, there has been a recognition of the need for communication, training, discourse of business and engineering practices, and a free exchange and understanding of the economics of the industry.

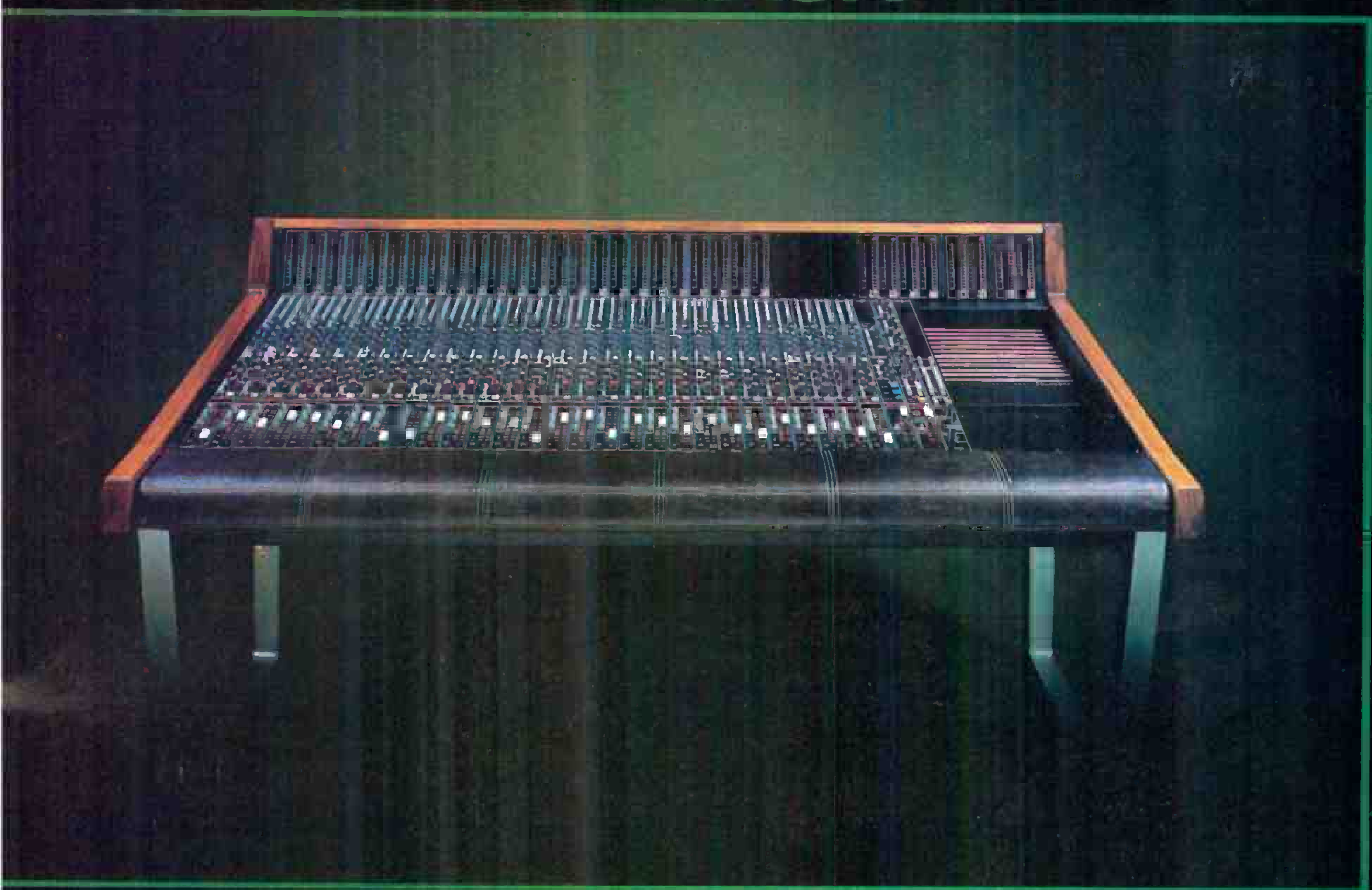
To that end, the organizations listed have formed the Society of Professional Audio Recording Studios. The purposes of the Society are:

(a) To establish a forum for professional audio studios as a positive and creative force;

(b) To establish a standard of excellence throughout the professional audio recording industry;

(c) To establish a Code of Professional

ROLLBACK



When we offered a temporary discount on our 24 series last year we expected a favorable response.

We didn't expect a 40% surge in sales on 2824 and 3624 Master Recording Consoles.

Obviously such an increase in demand has meant a step-up in production. And thanks to Harrison engineering, that increase in production now allows us to manufacture with improved efficiency and reliability.

As a result, we'd like to give you something <u>you</u> didn't expect. A price <u>ROLLBACK</u>. Even though the 10% discount is no longer in effect we're not just holding the line on the 24 series, we're rolling back the price a full 5% off last year's pricing.

Now, more than ever, Harrison <u>means</u> NO COMPROMISE.

HARRISON 24 SERIES RECORDING CONSOLES
3624—$72,724 2824—$60,390
U.S. Prices—Subject to Change

Harrison

HARRISON SYSTEMS, INC. • P.O. Box 22964, Nashville, Tennessee 37202 • (615) 834-1184, Telex 555133

— SPARS —
(continued)

Engineering Practices to which the organization members will adhere;

(d) To maintain high technical and cultural standards in the craft and to require high standards of ethical practice by members of the corporation;

(e) To act as a voice to address manufacturers and others in the important issues confronting the industry's present and future equipment needs;

(f) To analyze, evaluate, and comment upon professional audio equipment;

(g) To foster the dissemination of information concerning techniques of studio management and technical innovation;

(h) To educate the organization members in matters affecting professional audio;

(i) To assist in the development of projects, undertakings, studies and other activities which will benefit audio recording studio owners and operators.

Applications of those interested are invited. Write or call:

Mr. Joe Tarsia, Sigma Sound Studios, (215) 561-3660, Chairman of the Board;

Mr. Chris Stone, Record Plant, Los Angeles, (213) 653-0240, West Coast Regional Vice President;

Mr. Bob Liftin, Regent Sound, New York, (212) 245-2630, Northeast Regional Vice President;

Mr. Mac Emerman, Criteria Recording, Miami, (305) 947-5611, Southeast Regional Vice President;

Mr. Mack Evans, Masterfonics, Nashville, (615) 327-4533, Mid-America Regional Vice President;

Mr. Kent Duncan, Kendun Recorders, Burbank, (213) 843-8096, Treasurer;

Mr. Dave Tieg, Atlantic Studios, New York, (212) 484-6000, Secretary.

To qualify for membership applicants must meet the following qualifications:

(a) Applicant must be a Professional Audio Recording Studio which has been in business for at least two (2) years prior to application for membership.

(b) Applicant studio must contain at least one state-of-the-art twenty-four track room, or quality mastering room, or quality audio recording remote truck.

(c) Applicant studio must have demonstrated the highest professional technical and ethical business practices in its community as well as in the industry.

(d) The primary business of the applicant studio must be professional audio recording or mastering time and material sales.

The anticipated activities of the Society may include: establishing technical criteria, management technique exchange, education in common bond sessions, group insurance, professional referral service, newsletter and survey services, investigation of sharing legal services, group buying of computer services, and equipment theft alerts.

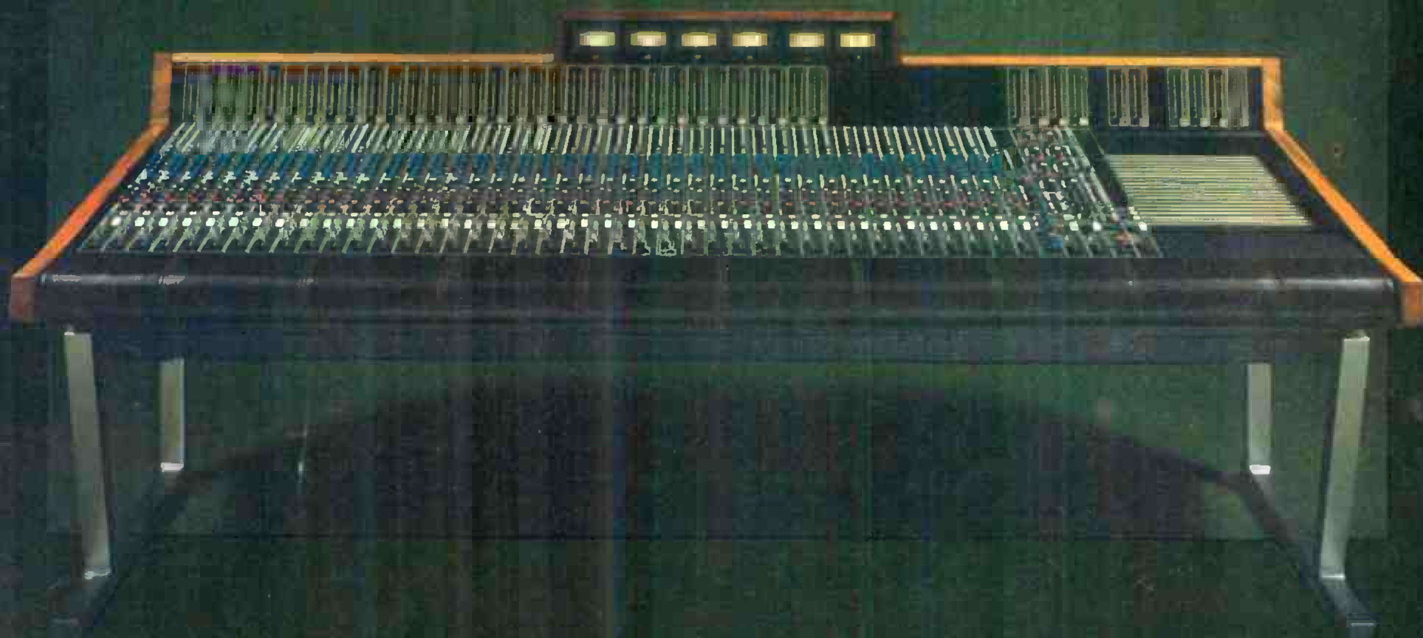
Please consider submitting applications prior to October 15th to enable you to participate at our first general meeting, November 1, in New York City, where permanent officers will be elected.

The newly elected Board of Directors will welcome questions at the SPARS suite at the Waldorf-Astoria Hotel, in New York, Saturday, November 3, 3:00 to 8:00 p.m. (Please check hotel information for suite number.)

The following studios have submitted membership applications at this time: A&R Recording, New York, NY; Ardent Recording, Memphis, TN; Atlantic Studios, New York, NY; Automatt, San Francisco, CA; Chateau Recorders, North Hollywood, CA; Criteria Recording, Miami, FL; Fanta Professional Services, Nashville, TN; Frankford/Wayne Recording, Philadelphia, PA; Group IV Recording, Hollywood, CA; House of Music, West Orange, NJ; Howard M. Schwartz Recording, New York, NY; Kendun Recorders, Inc., Burbank, CA; Location Recording, Burbank, CA; Masterfonics, Nashville, TN; Pecan Street Studios, Austin, TX; Record Plant, Los Angeles, CA; Regent Sound, New York, NY; Sigma Sound Studios, Philadelphia, PA; Soundmixers, New York, NY; Studio 55, Los Angeles, CA; The Hit Factory, New York, NY; Universal Recording, Chicago, IL; Wally Heider Recording, Hollywood, CA.

The Society of Professional Audio Recording Studios sincerely invites all recording companies, equipment manufacturers, suppliers, prospective clients and interested parties of the recording industry to submit suggestions, questions and general comment to any SPARS Board member. We are as interested in understanding you and your opinions, as we are in having you understand us and appreciate our position.

Our common denominator is our mutual interest and love of our industry. We invite all to join us in furthering its progress.



For the last four years you have heard about the innovative design and technical superiority of Harrison consoles. In no place has the Harrison "no compromise" philosophy been more evident than in the Harrison 32C series recording console. This philosophy pays off for you in the knowledge and confidence that your Harrison 32C series console is the most reliable console built anywhere in the world. What does this innovative design, technical superiority and reliability mean to you

success

The Harrison 32C series console has become the industry standard with installations in every major recording center in the world. True, a Harrison console initially costs more than most other consoles, but in the long run this investment in quality pays off in less down time, increased operating efficiency, and a superior recorded product. What does this investment security and operating efficiency mean to you

success

Success is full-time utilization of your recording facility, and that translates into profit for you. Harrison consoles are used in major recording studios around the world. Many of these studios have purchased a second, and even a third Harrison console. Harrison 32 series consoles have been instrumental in the production of Grammy Award winning recordings for "Best Engineered Recording (non-classical)" for the last two years in a row, and you know what that means

success...

the bottom line of our "no compromise" philosophy!

Harrison

HARRISON SYSTEMS, INC. • P.O. Box 22964, Nashville, Tennessee 37202 • (615) 834-1184, **TELEX 55513**

fact: this condenser microphone sets a new standard of technical excellence. & it sounds superb!

The Shure SM81 cardioid condenser is a new breed of microphone. It is a truly high-performance studio instrument exceptionally well-suited to the critical requirements of professional recording, broadcast, motion picture recording, and highest quality sound reinforcement — and, in addition, is highly reliable for field use.

Shure engineers sought — and found — ingenious new solutions to common problems which, up to now, have restricted the use of condenser microphones. Years of operational tests were conducted in an exceptionally broad range of studio applications and under a wide variety of field conditions.

As the following specifications indicate, the new SM8' offers unprecedented performance capability — making it a new standard in high quality professional condenser microphones.

SM81 puts it all together!

- WIDE RANGE, 20 Hz to 20 kHz FLAT FREQUENCY RESPONSE.
- PRECISE CARDIOID polar pattern, uniform with frequency and symmetrical about axis, to provide maximum rejection and minimum coloration of off-axis sounds.
- EXCEPTIONALLY LOW (16 dBA) NOISE LEVEL.
- 120 dB DYNAMIC RANGE.
- ULTRA-LOW DISTORTION (right up to the clipping point!) over the entire audio spectrum for a wide range of load impedances. MAXIMUM SPL BEFORE CLIPPING: 135 dB; 145 dB with attenuator.
- WIDE RANGE SIMPLEX POWERING includes DIN 45 596 voltages of 12 and 48 Vdc.
- EXTREMELY LOW RF SUSCEPTIBILITY.
- SELECTABLE LOW FREQUENCY RESPONSE: Flat, 6 or 18 dB/octave rolloff.
- 10 dB CAPACITIVE ATTENUATOR accessible without disassembly and lockable.

Outstanding Ruggedness

Conventional condenser microphones have gained the reputation of being high quality, but often at the expense of mechanical and environmental ruggedness. This no longer need be the case. The SM81 transducer and electronics housing is of heavy-wall steel construction, and all internal components are rigidly supported. (Production line SM81's must be capable of withstanding at least six random drops from six feet onto a hardwood floor without significant performance degradation or structural damage.) It is reliable over a temperature range of −20° F to 165° F at relative humidities of 0 to 95%!

Send for a complete brochure on this remarkable new condenser microphone! (AL577)

SM81 Cardioid Condenser Microphone

® SHURE ®

Shure Brothers Inc., 222 Hartrey Ave., Evanston, IL 60204, In Canada: A. C. Simmonds & Sons Limited
Manufacturers of high fidelity components, microphones, sound systems and related circuitry.

for additional information circle no. 94