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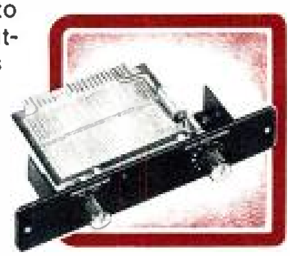

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COMING NEXT MONTH

• An Englishman, W. E. Anderton, assistant editor of Wireless World, gives his view of the developing recording industry and shows how demands from engineers and artists have led to changes in recording techniques. He explains what the influence will be on the future of commercial recording in PROFESSIONAL SOUND RECORDING—A BRITISH VIEW, Part I.

• A UNIQUE RECORDING AND REPRODUCING SYSTEM describes an elaborate home-recording and reproducing system and is written by Richard Burwen, president of Burwen Laboratories.

• RECREATING COLONIAL SOUND is the business of Richard B. Tisdale, audio engineer with the Colonial Williamsburg Foundation. His article tells precisely how to record an eighteenth century organ in a seventeenth century building without losing the colonial sound.

• And there will be our regular columnists: Norman H. Crowhurst and John Woram. Coming next month in **db, The Sound Engineering Magazine.**

ABOUT THE COVER

• Japanese preoccupation with audio production is rooted in ancient tradition, as this painting of musicians making sound waves with stringed instruments records.

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THE SOUND ENGINEERING MAGAZINE

OCTOBER 1974 VOLUME 8, NUMBER 10

32 NOISE CONSIDERATIONS IN AUDIO AMPLIFIERS
R. S. Mintz

35 BUILD A SUPER WINDOW FOR YOUR STUDIO
Stephen H. Lampen

38 RECORDING STUDIO ACOUSTICS, PART 2
Michael Rettinger

4 LETTERS

8 THE SYNC TRACK
John Woram

14 THEORY AND PRACTICE
Norman H. Crowhurst

20 NEW PRODUCTS AND SERVICES

42 CLASSIFIED

44 PEOPLE, PLACES, HAPPENINGS

db is listed in **Current Contents: Engineering and Technology**,

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advertisers index

AKG 5
Amber 39
Audio-Technica 6
Automated Processes 7
Bose 30
Broadcast Electronics 25
David Clark 8
Crown . . . . cover 2
Electro Sound 23
Electro-Voice . . . . . cover 4
Garner Electronics 16
Gately 26
Gotham 27
Hervic Electronics 15
Infonics (Visual Educom, Inc.) . 37
Inovonics 10
Lockwood Enterprises 28
London Company 39
Lumiere Productions 12
Martin Audio 29
Rupert Neve 13
N. V. Philips . . . . . cover 3
Polyline 16
Pulse Dynamics 28
Rauland-Borg 24, 30
Revox 11
Russco Electronics 14
Sescom, Inc. 12
Shure Brothers 21, 31
Spectra-Sonics 10
Standard Tape Labs 6
Stanton Magnetics 2
S. W. Technical Products . . . 25
Tascam Corporation 9
Timekeeper 22
Warehouse Sound 4
Waters Manufacturing 3
Westlake Audio 17-20
Woram Audio 29

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letters

THE EDITOR:

The article in the August, 1974 issue of **db** on *Controlled Time Delays for Speech Reinforcement Systems* by Sidney L. Silver is a most important and valuable bit of information on the subject.

The history of the Haas information is as follows: *The Influence of a Single Echo on the Audibility of Speech* (Über den Einfluss eines Einfachechos auf die Horsamkeit von Sprache) was a dissertation by Herr H. Haas for a doctor's degree, University of Göttingen. The work was carried out under the direction of Professor E. Meyer, to whom thanks are due for permission to reproduce this paper.) This is copied from the cover page of an English translation by K. P. R. Ehrenberg, Dr. Eng., designated Library Communication No. 363, Department of Scientific and Industrial Research, Building Research Station, Garston, Watford, Herts. England. This was dated December, 1949 and is obtainable by writing to the above address. The publication was held up for insertion in the first issue of *Acustica*, Vol. 1, 1951.

The paper mentioned in reference two in **db** was on work done at Harvard. A statement in an Editor's note on this paper in the *Journal of the Audio Engineering Society* for December, 1973, which is a reprint from *The American Journal of Psychology*, was "A thorough reading of both papers will indicate that they substantiate each other's conclusions although the experimental methods differ." However, M.B. Gardner[1] states that up to a certain point the Göttingen and Harvard studies are in general similar on sound localization or Precedence Effect. Beyond this "they show much greater difference in the nature and detail of their respective investigations."

My first acquaintance with the Haas effect was through the *Acustica* article. Fortunately, I had the aid of a library translator while I was a member of the technical staff at the Bell Labs. Incidentally, Dr. M. R. Schroeder, director of acoustics, speech and mechanics research at BTL, was one of the observers in the Haas tests and has now taken Prof. Dr. Erwin Meyer's place at the University of Göttingen. Dr. Schroeder has told me that in the test procedure an echo was considered disturbing when listening became somewhat of an effort or a little uncomfortable, although the intelligibility was not affected. The term "echo" has been used almost

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universally in scientific language for all reflections. Actually the delayed sound does not become an echo until it can be heard separately from the direct sound. When the echo comes in the critical period it cannot be heard as such. Its effect is to increase the loudness *and intelligibility* of speech.

The increase in loudness can be as much as 10 dB greater than the direct sound before the echo is heard as a true echo. Actually a single echo (used in the Haas tests) is not representative of actual sounds in speech or any other. In E. G. Richardson's book, TECHNICAL ASPECTS OF SOUND, Vol. 3, p. 285, is shown the effect of many echoes of different times producing a sound whose echoes are 12.4 dB louder than the direct echo before they are perceived as such. In other investigations, as many as thirteen reflections were used.

While all of the above is practical information for sound reinforcement systems, the data can also be applied to the design of auditoriums. Each reflection can be controlled in time and magnitude, as per the Haas data. Since Haas used only reverberation times of zero, 0.8, 1.6 seconds these curves, actually straight lines, have been extended to three seconds,[2] resulting in a doubling of the critical period to 300 ms.

Subsequent investigations have arrived at grouping all reflections in the critical period as early reflections and all later ones as late reflections. The ratio of the two groups are important in evaluating the acoustics of rooms. Acoustic tests prove that in the room under test, even for a length of 80m and with 4600 seats, a speaker (not an electrical transducer, because this is the correct meaning of speaker) can be understood by the remotest listener without any electroacoustic amplication whatsoever.[3]

To show that research is constantly going on in this field, an article in *The Journal of The Acoustical Society of America*, August 1974 (arrived just 30 minutes ago) by Bertram Scharf states that "The loudness summation between pure tones appear to be independent of the real or apparent locus of the source."

1. M. B. Gardner. "Historical Background of the Haas and/or Precedence Effect." *The Journal of The Acoustical Society of America*, Vol. 43, No. 6, June 1968, pp. 1243-1348.
2. A. R. Rienstra. "The Gap between Good Acoustics for Speech and for Music." *The Journal of The Acoustics Society of America* Vol. 37, No. 3, March 1965.
3. W. Gabler, R. Buchlein, E. Krauth and F. Spandick, "Investigations on an Acoustically 'Ideal' Room," Acustica, Vol. 19, (1967/68).

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John M. Woram

THE SYNC TRACK

● I have a feeling this microphone thing could turn into a hook of its own. The reader reaction has been a pleasant surprise. Up until now, I had the impression that readers were not letter writing types. Oh well, wrong again!

Before going further, I should apologize to the writer who started the whole controversy with his letter (June SYNC TRACK). He certainly didn't deserve the treatment he got.

Actually, my testy remarks were more a reaction to the many similar sentiments I've heard over the years from so many people just starting out in recording.

There's that wide credibility gap between the teachers and the learners. The teachers want to speak of polar patterns, proximity effects, and 3:1 ratios. The learners want to hear what microphone to use on the guitar and drums. When the teachers persist with their talk of off-axis frequency response degradation the learners get restless and say they're being cheated out of what they want to know.

I had an interesting chat with a university professor at the Audio Engineering Society convention last week. (see next month's issue for a full convention report.) "What," he asked, "is the problem with teaching students about microphone placement?"

We discussed a typical situation: miking a guitar. It's easy enough to supply documentation of the microphone and its placement on some recent session. Working distance, angle, polar patterns, manufacturer, etc.—all this could be tabulated with not too much trouble.

But what about the other data?

● Was the guitar on the boomy side, or was it perhaps a bit twangy?

● Were there other instruments playing at the same time?

● What were those other instruments?

● Was the studio live or dead?

● Did the producer want a "close-up" sound?

● Who was the guitar player—the engineer—the artist?

● What was the arrangement like—etc., etc?

Now, the answer to any one of these questions could completely change the choice of microphone and/or it's placement. So, what can be said to the beginner who wants to do the best job he can? About the only thing is, learn the rules and then develop your own technique.

Well, although the professor agreed that mic placement could not be taught very well from a book, he thought that a little discussion now and then of specific microphone applications would not necessarily be a bad thing. Perhaps he's right.

In fact, speaking of other books, I suspect one could be written just on microphone placement alone. However, it would have to be "Volume II", with Burroughs' book considered as "Volume I", for without a solid understanding of the basics, discussions of microphone placements may do more harm than good.

I guess that may be crux of the microphone controversy. For some reason, people starting out in recording want to begin with the advanced lessons, and skip all that theory.

Perhaps medical students too would rather begin their careers in the operating room and pass up all those boring lectures. Fortunately for the rest of us, they are required by law to go through a long basic training before getting a chance to cut up a patient.

We don't have such requirements in the recording business. Many successful engineers have nothing more than the skills they picked up on-the-job. Some are doing creative work, even if they can't explain the theory behind the practice. Many of these engineers began in the dark ages of mono. As the industry developed, so did they. They're now doing 16 (or more) track sessions, but haven't forgotten "the old days" when most of the work had to be done on the session, because what you heard was what you got! The "We'll fix it in the mix" syndrome was unknown.

Today's beginner is starting out in a technology that has expanded beyond the wildest dreams of any pre-stereo forecast. And the competition is fierce. If you want to join that competition with nothing more than your good looks and enthusiasm, good luck! You're going to need it. But I still think a little time out for some theory

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wouldn't hurt.

If you want to disagree, you're entitled. But don't take it out on the text books if they don't tell you what you want to know.

Here's a couple of quickie applications notes that occurred to me after doing some reading. And of course, there are many many others.

MICROPHONE PLACEMENT TECHNIQUES

Organ with Leslie Speaker Cabinet. Where do you place the microphone? If you get too close, the mic hears the rotating speaker motor. If you back up, there's too much leakage from the other instruments in the room. Sometimes, two microphones are used; one near the top speaker, another near the bottom. But then, these problems are just multiplied by two.

Try a figure-8 microphone, placed very close to the speaker cabinet front, with the front of the microphone facing up, toward the top speaker, and the rear of the microphone facing down, toward the speaker in the base of the cabinet. Now, one dead side of the microphone is pointing into the cabinet, and the other is pointing into the room. Since the dead sides of a good figure-8 mic are a lot less sensitive than the rear of a cardioid, the leakage from the other instruments in the room should be substantially reduced. Move the mic up or down to get the proper top-to-bottom speaker balance. If you don't have a figure-8 mic, you might try two similar cardioids, back-to-back, and connected out-of-phase.

Miking a harp on a rock session. A harp on a rock session? Usually it's a sweetening session, but still there's apt to be brass, strings, and what not going on at the same time. Now where do you put the harp microphone? No matter where, it's inevitably pointing right through the harp at some other instrument.

So, try a lavalier mic, placed in one of the sounding board holes. Some sponge rubber wrapped around it will hold it in place and absorb vibrations. A regular microphone might sound excessively boomy here—especially a cardioid with proximity effect. However, the rising mid range response of the lavalier seems rather well suited to the location. This would probably be a disaster on a classical session, but it may be your only chance when the harp is trying to compete with a wailing brass section.

For a future column, I'll try to come up with a few more applications notes that might be of passing interest. In the meantime, consider the words of one writer who wrote,

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Or the medical centers that use specially adapted A77's for electrocardiographic recording.

We could go on and on (see accompanying list), but by now you probably get the point.

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" . . . try using mics to meet your specific situation and needs, not some formula that worked for one person, one time, in one situation!! . . . After twenty years with audio I'm still learning and Burroughs' book was a big *lesson."*

Moving right along, here's *another* letter.

Dear Mr. Woram:

HELP! I've got a hum.

Two hums, to be precise. The first is so bad that I have given up using the equipment which causes it. This, inconveniently enough, happens to be my entire network of mic jacks in the studio. Entirely bypassed I get perfect results, even though I am still running through the same wall. But if I hook a mic into the jacks and patch the other end into one of my decks, I get unbelievable hum. (Would you believe a full 0 dB at one quarter gain?) I get hum even if I don't have a mic on the other end. Every engineer on my staff has become bald trying to figure this one out, and I've even asked a few people from other studios hoping an outside viewpoint would give a fresh idea. No go. We are not running balanced lines, but we tried that too. Also no go. The length of each extension is less than ten feet, all cable is woven shield, all grounds are hooked together properly, and the entire ground network, (8 gauge copper baling wire) is hooked to a cold water pipe and sunk 7 feet into the ground on the end of a spike. It would seem I have plenty of grounding.

The other hum is an insidious one. It is not very noticeable except at very high gain levels, but it is there. It is not constant either. It sort of "wows" very regularly in just about every recording we make. Needless to say this is not good for business, and besides that, I won't take money from a man and hand him something with that on it. So I'm really ready to open bidding on a nice small 4 track studio. Unless you can help with a few suggestions. One suggestion which has been made but not investigated because of the magnitude of it is that the building we are in (built c. 1919, wooden frame with its original wiring) is creating the hums, (both of them) and we would have to completely re-wire the entire building to get rid of it. Does this sound possible? Or more like someone who won't admit he doesn't know either?

Needless to say, I feel like a jerk, since I am supposed to be chief engineer of this lunacy factory, so I will remain, faithfully yours,

John Doe

I've been staring at this one for a couple of weeks now. At the risk of sticking my neck out, I'd suspect it's an easy enough problem to solve, yet Mr. Doe says that neither he nor anyone else has had any luck.

As I read it, if a cable is run from the microphone in the studio to the tape recorder in the control room, everything is o.k., but when the mic is inserted into a jack(?) and the other end is patched(?) into the tape deck, there's trouble.

Do you really mean studio mic *jacks?* Jacks in a microphone line are always an invitation to trouble. Mic levels are quite low, and any patch points before the microphone preamp are potential trouble spots. Or, do you mean regular mic plugs, or the Switchcraft/Cannon type? If so, the trouble probably lies elsewhere.

Is there a console involved? I'd suspect so, although there is no mention of one in the letter. When Mr. Doe plugs his microphone directly into the tape recorder there is no problem. Obviously, (I think) the tape recorder has a mic level input. But if the patch bay arrangement is feeding some type of console, maybe its output is at line level. Does the tape recorder also have a line level input? Or is the console output being improperly fed into a mic level input on the recorder?

If there is no console, why the jacks? And if there is a console, with an unbalanced output, perhaps there is a phase reversal somewhere that is electrically grounding both sides of the lines.

I doubt that the building is contributing much to the problem, since bypassing the jacks clears up the problem.

As for the "other hum," is it present in *every* recording, or just in some, or most? Do you hear it *only* on playback, or always? Depending on the answers, different parts of the system may be under suspicion?

That's not much help I guess, but more information is needed. The point of printing this is to suggest that such problems be very clearly stated, with *complete* details about *every* component in the system given. Often, in so doing, the solution will be pointed out. For if you find yourself writing "high impedance mic into low impedance input,' or "line level out into mic level in," you'll have uncovered the trouble spot by yourself. ■

Norman H. Crowhurst

THEORY AND PRACTICE

• One thing about the workshop at Brigham Young University was that too many of the participants did not have sufficient math, and in particular, understanding of the dB versus voltage and/or power ratio bit, to handle problems. So the week before I arrived on the scene had been spent by the students polishing up some of those deficiencies. One outcome of this was the suggestion that for future workshops some individualized instruction be made available to generate familiarity with such calculations.

That, of course, is something very much in line with some of the materials I have been producing for *Educational Research Associates,* but for this purpose, they really should be tailor made, and we hope they can be.

But, to get down to some real theory and practice, one of the things that took quite a little time related to "constant voltage distribution," also commonly known as "70-volt line." At the time, copies of the May issue of **db** were floating round the workshop and it seemed appropriate that Don Davis had the second part of his two-part article in there, which dealt with 70-volt transformers.

However, although that article looks as if it breaks it down to real easy-bite size, it did not prove easy for all the participants. We had to work through what is assumed constant, and what changes, and how those changes interact, quite a bit before most of the participants felt they really knew what it was all about.

I suppose what starts the confusion, is the fact that in 70-volt transformers) the secondary is rated, or labeled, in ohms, which is not so uncommon in transformers, while the primary is marked off in watts, which is unusual. How can a transformer transform watts into ohms? Of course the answer is, it doesn't. What all transformers "see" is impedance, and it is *impedance* they transform.

Thus, if you connect a 4-ohm speaker to the 4-ohm secondary tap, or an 8-ohm unit to the 8-ohm tap, on the secondary side, the primary will transform that 4 or 8 ohms, as the case may be, to some other value, that appears on the primary side. If the primary has a tapping rated at 2 watts, this means the impedance is such that 70.7 volts rms across that tapping will deliver 2 watts into the 4 or 8 ohm unit.

As we pointed out last month, that is a nominal 4 or 8 ohms, whose actual value varies with frequency. Taking the design figure, for 70.7 volts to deliver 2 watts means the formula for power, given voltage and impedance, which is V^2/Z, will give 2 watts as the answer. As V^2 is 5,000, this means Z must be 2,500 ohms.

Now, if your speaker, rated as having an impedance of 8 ohms, has an actual impedance that varies between, say, 6 and 20 ohms, then the impedance reflected on the primary will vary accordingly. The fact that the nominal 8 ohms reflects as 2,500 ohms means that the impedance ratio is 312.5:1, which follows that 6 ohms reflects as 1,875 ohms and 20 ohms reflects as 6250 ohms. Those are the realities.

All the transformer knows is that whatever you put on the 8-ohm tap, the primary tap labeled *2 watts* will present an impedance that is the same thing multiplied, at each and every frequency, by 312.5. If there is a 5-watt tap, the nominal impedance will be $5{,}000/5 = 1{,}000$ ohms, with the same kind of variation in actuality with frequency. And if there is a 1-watt tap, the nominal impedance will be 5,000 ohms with just the same kind of deviations with frequency.

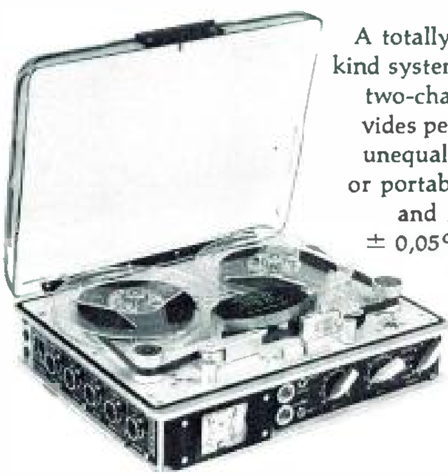
Now, in Don's article, he suggested that the common or garden variety 70-volt transformer isn't very good, especially at low frequencies. As Don has been around almost as long as I have, I was a little surprised that he did not at least conjecture how this situation arose. So let me fill that in for you.

Back some forty years ago, when 70 volt line got started—in those days I worked in England, where we used 100 volt line instead—it was used primarily to make things easier and more

Stellavox. Your best investment in professional sound recording equipment.

All Stellavox sound systems are designed and manufactured in Switzerland (land of engineering perfection) and incorporate the latest, most advanced "state-of-the-art" features* which provide *the* highest quality sound reproduction *every time* (even under the most demanding on-location recording conditions).

* **Plug-In Head Modules** (Only Stellavox recorders feature the capability of being able to accept all the mono and stereo track configurations in modules that are each individually tailored for ultimate performance on a specific recorder)
* **Increased Reel Capacity** (With the Stellavox ABR extension accessory, reel capacity is increased up to 10½" for longer recording time)
* **Battery Operation** (In addition to AC operation, all Stellavox recorders operate on batteries thus permitting the utmost in portability and flexibility)
* **Professional Accessory Line** (A full selection of professional accessories are available for each Stellavox recorder)



Stellamaster SM-7
A totally unique, one-of-a-kind system, the Stellamaster two-channel recorder provides performance abilities unequalled in either studio or portable recorders. Wow and flutter (weighted): ± 0,05% (0,04% typical). Overall frequency range: 20 Hz to 28 kHz ± 2 dB. Signal to noise ratio: 74 dB (Maxell UD 50)

Quadrasonic SQ-7
The only professional 4-channel battery operated unit of its kind available today. Wow and flutter (DIN weighted): ≦ ± 0,06%. Overall frequency response: 25 to 25 kHz ± 2 dB. Signal to noise ratio (rel. 800 nW/m): ≦ 66 dB ASA A.

Stellavox SP-7 The ruggedly built SP-7 is so compactly designed that you'll be amazed at how little it weighs (only 8 lbs. with tape and batteries). Wow and flutter: < ± 0,07%. Overall frequency response: 30 to 16,000 Hz ± 2 dB. Signal to noise ratio, weighted (ASA A filter) > 65 dB stereo and > 70 dB full track (at 50 mMx/mm).

Each Stellavox recorder is backed by warranty service which is available through our service center in Los Angeles. Factory trained technicians, utilizing a full stock of parts and the latest in laboratory measuring devices, will handle all aspects of quality control, maintenance and repair of Stellavox products.

If you would like to receive full technical information on the Stellavox line of professional sound recorders and mixers, please write to:

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Circle 18 on Reader Service Card

efficient for the installer. Most of them were used on horn loudspeakers, installed out of doors at the top of tall poles.

Now, being of the horn variety, they had a low-frequency cut-off, usually around 500 Hertz (or cycles, as we knew them then). And those—— horns were heavy enough to manipulate, with the installer standing on a ladder leaning against the pole without having a big fat transformer that would needlessly handle audio down to 20 Hertz!

So, to save money, as well as the installation man's muscles, we decided to use transformers that would handle only from 500 Hertz up. Then, to protect the amplifiers, we put a capacitor in series that made the whole thing behave as a high pass filter. Incidentally, this also protected the horns so they were not driven at frequencies below their cut-off. So everything was happy, as well as everybody.

Where the same system also fed indoor, full-range speakers, we had another variety of transformer, built huskily enough to handle the low frequencies as well. So the same amplifier could feed full range speakers indoors and horns with low-frequency cut-offs outdoors. To us at the time it all seemed quite simple.

For the outdoor installations, we would use a capacitor appropriate to the wattage, and thus the impedance, involved. Thus a 2-watt transformer for 500 Hertz roll-off would require a capacitor with a reactance of 2500 ohms at 500 Hertz. You want to figure that, whatever way you like? It should come to rather more than 0.1 mfd. In practice, a 0.1 mfd. does the job okay.

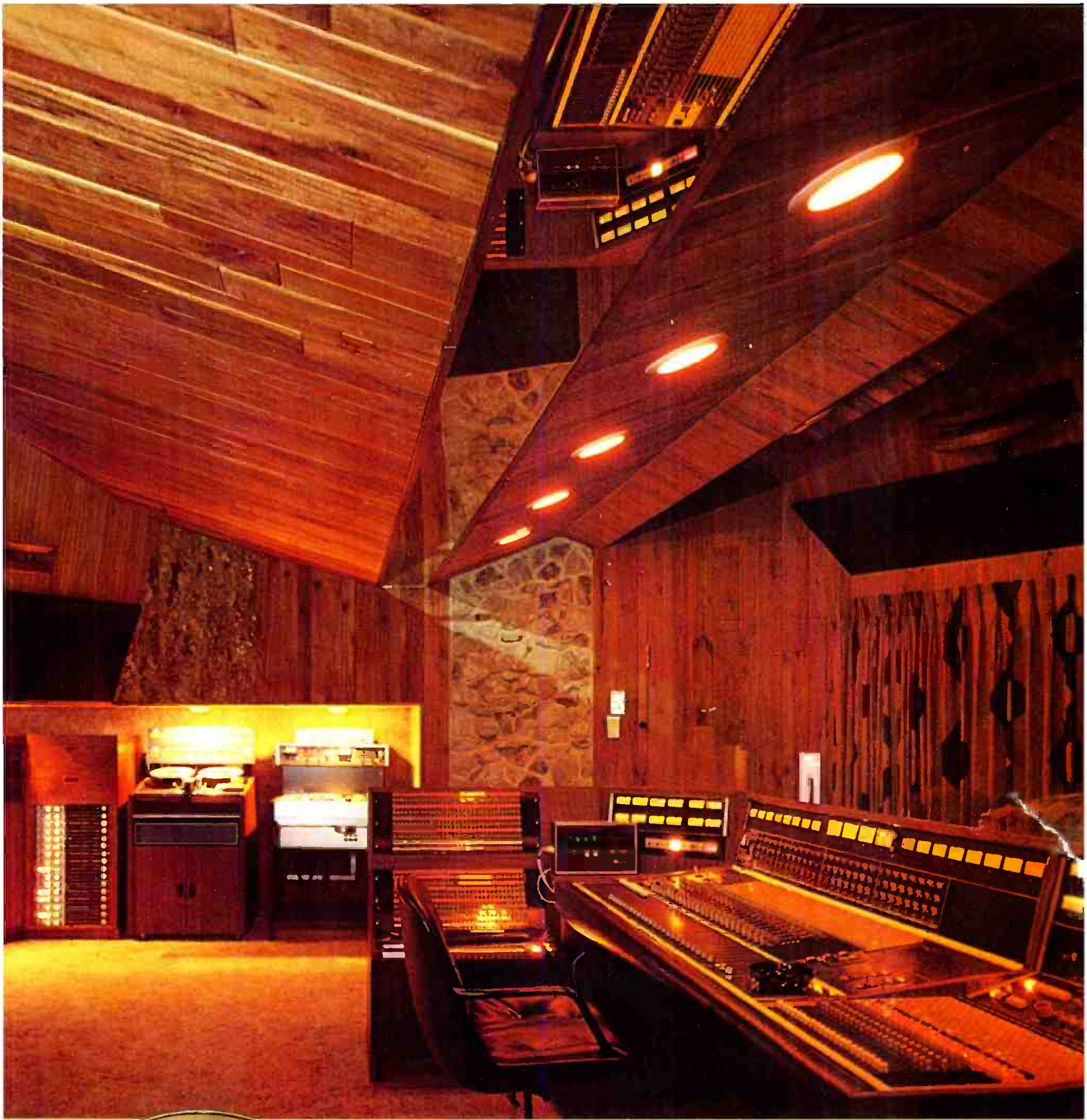
Then, if we put the right capacitor in every speaker housing, that added up to quite a few capacitors. And to be strictly correct, we really needed to put in one for each tap. Thus, if the speaker transformer had taps for 1, 2 and 5 watts, it would need a 0.1 mfd capacitor for the 2 watt tap, a 0.25 mfd capacitor for the 5 watt tap, and a 0.05 mfd capacitor for the 1 watt tap.

To make life a little easier, we would use a single capacitor for the whole horn installation. Thus, if we were delivering a total of 50 watts to a horn load, the aggregate impedance of the load, all those transformer primaries in parallel, would be 100 ohms. We needed a capacitor with a reactance of 100 ohms at 500 Hertz. This would be 3.2 mfd (that 0.25 for 5 watts was only approximate).

One way we used to do this would be to use two 6.4 mfd electrolytics back to back, and provide them with d.c. polarizing through a fairly high value resistor, so that to audio they looked like 3.2 mfd. This way the amplifier was still just as well protected, and the horns got only frequencies above 500 Hertz, so once again, everyone and everything was happy.

Well, of course, a lot of things have changed since then. Installation is easier, with those snorkel vehicles, made so the man doesn't have to hold onto the ladder, the horn, and the pole all at the same time with only two hands. And I can only assume that somewhere along the line, the full range transformers got dropped. I can imagine that with the kind of information some catalogs give, people who specified equipment could not see why they should buy a transformer costing about ten times as

Circle 23 on Reader Service Card →



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Pre-Planning: Which markets are you going after? One studio or two? What is the sound character desired? How much multi-track separation? Your budget? Future expansion? These are just a few of the questions discussed at this stage. From the answers, we'll provide you with a comprehensive proposal covering our recommendations and the cost for the entire project.

Site Evaluation: If you're planning to build from the ground up or within an existing shell, we'll help you to select the best building or location in terms of size, demolition and construction costs, zoning, noise transmission problems and those "hidden" costs. Remodeling an existing studio? We can show you how to get the best results from your space.

Design: Acoustical design is the key to the success of a studio. Westlake Audio's geometric and acoustic designs will provide you with a winning "sound." **Prior to construction,** we will guarantee the performance of your studio and control room with respect to isolation, response, decay time in various frequency bands and dispersion.

Equipment Selection: We have carefully evaluated the many products on the professional audio market to provide our clients with a broad selection of the finest recording equipment available. From consoles, automation and tape machines to monitors, microphones and support equipment, we have the lines to satisfy your requirements and budget.

Financing: We work closely with several companies whose specialty is providing clients with a lease or conditional sales plan tailored to your needs.

Construction: As the proper implementation and execution of the geometric and acoustic designs requires unique construction techniques, Westlake Audio will provide a construction foreman to oversee and work with your local construction crew and electrical and air-conditioning subcontractors.

Wiring: Our technical crew will install all the audio wiring throughout the studio complex and tie down the wiring to the microphone, headphone and tape machine panels, echo chambers, console, amplifier room and monitors.

System Interface: Our engineers will install all the recording equipment and interface the entire system to the console. A total system checkout is performed from microphone phasing to monitor equalization and balancing.

Training: Technical and mixing engineers will train your personnel in the operation and maintenance of the studio and equipment.

Studio Management: Upon request, Westlake Audio wi offer guidelines and suggestions for all facets of studio management including rates, booking procedures and controls, personnel requirements, wages, work orders, tape legends, library systems and credit.

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much as the one they needed when they both had the same rating; the only clue might be the difference in weight, if that was even given.

Something else seems to have been forgotten along the way, although part of it remains. What happens if you overload, or "underload" an amplifier, using the 70-volt line system? First, we need to define what overload or underload means.

The whole 70-volt line concept says that the maximum level, of a full-power continuous sine wave—which is not what we usually listen to—will provide 70 volts rms on the line. If an amplifier supplies 100 watts, then at 70.7 volts rms this means the impedance into which it delivers this 100 watts is 50 ohms.

If a speaker is intended to take 2 watts, then the impedance it puts across the line will be 2,500 ohms. A 5 watt speaker will look like 1,000 ohms, a 1 watt speaker will be 5,000 ohms, and so on.

Now, if you have only a few speakers connected, your 100 watt amplifier may find itself looking into a load of, say 500 ohms, instead of 50 ohms. As you probably know, that is usually okay. But what if we keep on adding speakers, until the load looks like 10 ohms, instead of 50 ohms? Now the amplifier cannot hack it.

In many modern solid state amplifiers, such an excessive current load would probably trip the protection circuits, cutting the system off. Or if the amplifier is safe with such a load, it probably means that the maximum current, which for 100 watts into 50 ohms, with 70.7 volts, would be 1.414 amps, is not exceeded.

So what the amplifier will probably do in this case is to deliver a very distorted 1.414 amps into the 10 ohm load, which will only be 14.14 volts, instead the rated 70.7. And instead of delivering its rated 100 watts, it will deliver only a very distorted 20 watts.

If you had piled on speakers until the load looked ilke 10 ohms, you were expecting the poor amplifier to deliver 1000 watts! So you will get only about 1 percent of the power you expect this way, and that very distorted.

Now, what we used to do at one time was to insert a matching transformer so at least we could do better than that. By stepping the voltage down, from 70.7 to about 22 volts (many amplifiers also have a 25 volt tap, which comes close), the proper load will now be 10 ohms, instead of 100 ohms. This means that instead of each speaker receiving only 1 percent of the rated power, very distorted, it will get 10 percent of the rated power, no longer distorted.

If cutting power down to 1/10th sounds not so good, remember two things: (a) it is a lot better than cutting it down to 1/100th, and that very distorted; and (b) 10:1 ratio, in power level, is only 10 dB.

We remember installations where making that change really saved the day. In one instance, a competitive company had installed 1000 watts' worth of speakers on a 200 watt amplifier, so the system only delivered about 40 watts worth of power, quite distorted—obviously being pushed beyond its limit.

Without our changing the installation beyond putting in a 100 watt amplifier that delivered its full 100 watts into the same speaker installation without distortion, the customer thought our 100 watt amplifier was far superior to our competitor's 200 watt job: much louder and clearer, undistorted. Actually, our level was only 8 dB up on his, and probably 30 or 40 dB down on his distortion. But with the addition of just the right transformer, his system would have been 3 dB up on ours!

All of which says, it's worth knowing what you are doing! ■

NEW PRODUCTS AND SERVICES

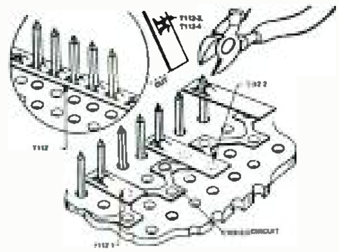
VARIABLE PITCH/DEPTH SYSTEM

● Designed to improve the performance of all Neumann and Scully lathes, regardless of vintage, VP-1000 provides a completely digitally computerized variable pitch/depth system for disc recording lathes. The control system employs a specially designed wide-range, low noise, high output dual 300 milli-second audio delay line, eliminating the need for any preview channels.
Mfr: HAECO
Circle 63 on Reader Service Card

MIXER KIT

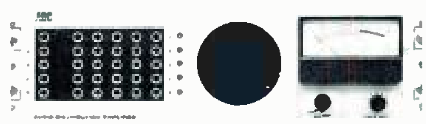
● Prokit II mixer contains slide attenuators and pan pots on all six inputs. Each input is switchable from mic to line input. The mic preamp gain is switch-selectable; each mic input has a switchable 20 dB pad. Professional vu meters are backed up by led overload indicators. Other features include optional transformer output and optional +48 volt microphone powering. Available in kit or factory wired.
Mfr: Gately Electronics
Circle 64 on Reader Service Card

SOLDERLESS BUSSING



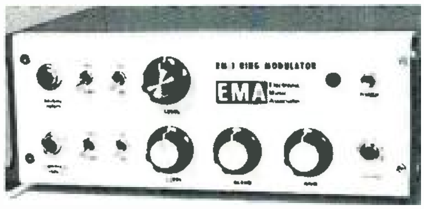
● Tiny barbs in T112 series phosphor-bronze bus links secure the strips to 0.025 inch square wrapped-wire posts without initial soldering. For quick, easy bussing or interconnections, the strips are merely pressed over the posts to provide continuity. The square hole firmly orients the strips on ninety degree increments to prevent accidental mispositioning. The links come in 12, 0.2, and 0.3-inch lengths with hole spacings of 0.1, 0.2, and 0.3-inches. The shorter clips can be used to temporarily secure DIP sockets to a circuit board until soldering affixes them permanently.
Mfr: Vector Electronic Co., Inc.
Price: $1.20-$12.75 per pack, depending on length.
Circle 65 on Reader Service Card

FIVE CHANNEL AUDIO DISTRIBUTION AMPLIFIER



● Independent level adjustment for five separate 600 ohms audio channels is possible with this audio distribution panel. Each of the five input channels has five isolated outputs. A switchable meter/monitor speaker circuit is used for signal adjustment and monitoring. Amplifier gain is adjustable from 0.0 db to 30 dB at a frequency response of 30 Hz to 15 kHz ± ½ dB. Input and output impedance is 600 ohms, with transformer coupled input and outputs.
Mfr: ADC Telecommunications
Circle 66 on Reader Service Card

RING MODULATOR



• Laser trimmed integrated circuits permit the critical internal adjustments on RM-1 ring modulator to be permanently set by precision laser beams. The unit does not need trimming or internal adjustment, even after severe vibration or drastic temperature change. Designed to serve as an interface between acoustic and electronic musical instruments, the modular unit can operate as an amplitude modulator (vca), a balanced (ring) modulator, or a frequency doubler. Universal input circuitry enables the user to modulate almost any combination of two input signals, such as microphones, electric guitars, oscillators, radios, synthesizers, etc. An a.c./d.c. switch on each input allows the use of slow d.c. control voltages as well as a.c. audio signals. The two signal inputs are designated *signal* and *carrier*. The *blend* control adjusts the percent of modulation by selecting the desired mixture of the signal and the signal-carrier modulation products. Using a sum and difference principle, the unit can modulate complex harmonics, such as the human voice. If no carrier is provided, the signal is automatically routed to the carrier input, creating a frequency doubler effect.

Mfr: Electronic Music Associates
Circle 57 on Reader Service Card

TWO-CHANNEL INTERCOM SYSTEM

• Specifically designed for two-channel communications in the high decibel levels encountered in the entertainment field, the CCX-200 system features a main station, which is interconnected with two-conductor microphone cable to lightweight remote belt packs. Up to thirty remote stations can be combined in any combinations, permitting both communications from the main station to remotes or internal communication among remotes. Master gain controls for each channel allow for independent level section. The main station is also available in rack mount.

Mfr: Clear-Com (Lumiere Productions)
Circle 58 on Reader Service Card

LIGHTWEIGHT STEREO HEADPHONES

• Weighing only 2.3 ounces, DT302 "open aire" headphones provides improved comfort. Ear cushions are made of soft acoustical sponge in color. The frequency response range is from 20-20,000 Hz. The impedance is 2 x 600 ohms; the headphone can be connected directly to either high or low impedance outputs. Also available is the DT301, a single-earphone for use in special studio applications and with cassette recorders, transistor radios, and dictating machines.

Mfr: Beyer (Revox)
Price: DT302: $29.95
DT301: $15.00

Circle 59 on Reader Service Card

AUDIO SPECTRUM DISPLAY

• A real time presentation of the spectral energy of an audio signal in applications where the conventional vu meter provides insufficient information is available through the use of this compact device, the Audio Spectrum Display, which incorporates a solid-state display and uses a digital time-sharing technique. The display divides the audio spectrum into ten octave-width segments, each octave displayed on a vertical column of ten light emitting diodes. Included in the device are two accumulative memories, each with a MEMORY ENTER switch and an ERASE button. A SAMPLE button allows short duration (effectively instantaneous) samples to be memorized. Four audio inputs can be accommodated and selected in any combination by front panel push buttons. A step type attenuator varies the input sensitivity in 2 dB increments to affect a wide range of audio levels.

Mfr: Amber Electro Design Ltd.
Circle 55 on Reader Service Card

SIGNAL GENERATOR

• Five selectable frequencies (50 Hz. 100 Hz, 1 kHz, 10 kHz, and 15 kHz) and a controllable level of signal output from −74 dBm to +4 dBm are offered by model 800 signal generator. The solid state generator is designed so that it may be turned on or off without introducing transients into the circuitry. All controls are located on the front panel; the unit measures a compact 3½ x 1½ x 2⅞ inches.

Mfr: Spectra Sonics
Price: $84.00
Circle 56 on Reader Service Card

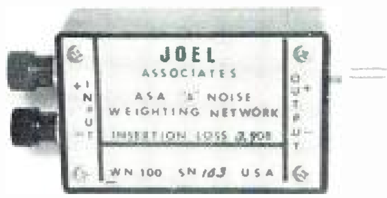
DUAL MASTER TRANSPORT

● Need for a conventional bin loop system is eliminated in the Dual Master transport. The unit is basically two high speed tape drives, each of which contains the same source material and operates in sequential mode. While transport "A" is playing, transport "B" is in parked mode; when "A" runs toward the end of the program, "B" will start pre-rolling and recording will switch over from "A" to "B" when "B" reaches a stable speed. Program changes can be made in less than thirty seconds per master. Enough tape length is provided for C-90 and C-120 tape duplication with 7.5 ips master. Available in ¼ inch, ½ inch and 1 inch versions, the transport is usable for cassette, 8-track, or open reel types.
Mfr: Recortec, Inc.
Circle 60 on Reader Service Card.

PARAMETRIC EQUALIZER

● Two versatile new equalizers, models 621A/621B, offer four cascaded equalization sections, each with non-interacting, continuously adjustable center frequency, bandwidth, and amount of boost or cut controls. Each section provides up to 16 dB peak and dip to minus infinity, permitting use as a notch filter to eliminate hum and other fixed-frequency interference. Each section tunes over a 20:1 frequency range, with broadly overlapping coverage for maximum flexibility. Bandwidth is continuously adjustable from approximately ¼ octave to 3 octaves. An overload light monitors each section, indicating the presence of peak clipping in any part of the equalizer; overloads can be corrected with the integral gain control. The equalizers, offering up to 12 dB gain, can be used at either line or intermediate-level patch points.
Mfr: Orban/Parasound
Circle 62 on Reader Service Card.

NOISE WEIGHTING NETWORK

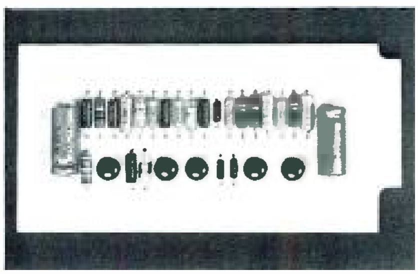


● This unit is designed to be placed between the device whose noise is to be measured and a high impedance audio volt meter. The curve obtained is based on the "A" curve, ASA standard S1.4-1961, which was adopted by the NAB in 1965 and is recommended by many tape recorder manufacturers. The purpose of a weighted noise measurement is to give a response curve similar to that of the ear at low volume levels, indicating more subjectively the signal-to-noise ratio than unweighted measurements. The unit measures 1⅛ x 3¼ x 2⅛ inches.
Mfr: Joel Associates
Price: $34.50
Circle 61 on Reader Service Card.

TAPE TENSION GAGE

● Magnetic recording tape of all widths can be measured for dynamic or static tension with hand-held model T2 Tentelometer. Tape tension on video machines can be accurately set while the machine is operating; head wear is optimized and transient problems can be detected. The unit is self contained, requires no external power, operates in any orientation and reads both in ounces and grams of tension. Models are available in 5, 12, or 20 ounce full scale ranges, complete with carrying cases.
Mfr: Tentel
Price: $179.00 up
Circle 53 on Reader Service Card

AUDIO AMPLIFIER

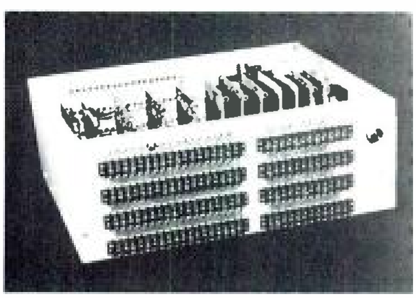


● Transformers are eliminated in solid-state model 110 audio amplifier, replaced by an integral active isolation device. A single design satisfies all amplifier requirements within an audio control system, performing the functions of microphone pre-amplifier, booster amplifier, mixing amplifier, line amplifier, and other such functions required up to line level. Contained on a modular printed circuit card, it features silicon transistors and diodes. Impedance is 600 ohms; frequency response extends from 10 Hz to 200 kHz. Size of the unit is 2½ x 5 x ½ inches; weight is two ounces.
Mfr: Spectra Sonics
Price: $72.00
Circle 54 on Reader Service Card

8-IN/4-OUT MIXER-CONSOLE

● Model MX-20 8-channel in/4-channel out mixer/console features eight balanced low impedance microphone inputs and eight unbalanced line inputs, with four fixed and four variable outputs. Microphones used with MX-20 can be optimized for each particular recording environment through the 5-position mic input equalization switches on channels one through six. Channels seven and eight contain pan pots which can direct input for distribution to any position between channels one and two. At each channel there is a 4-position mic attenuation switch. Each channel is also equipped with its own straight-line input control, using a vu meter and mic/line selection switch. Other features include a straight-line master level control, headphone output with volume control and channel selector, a varied patch panel, and a first stage fet pre-amplifier. The unit is light enough for portable use. Facilities are built-in for cascade connection of two mixers, providing up to sixteen channels of input.
Mfr: Superscope, Inc.
Price: $1,050.
Circle 48 on Reader Service Card

AUDIO CONTROL CONSOLE



● Separation of the control circuitry from the basic audio electronics is achieved in Series 35 controller. The controls can be placed in any location in the studio while the electronics are mounted in a rack for easy maintenance. The unit can be ordered for rack, cabinet, or custom applications with a choice of rotary, slide or push-button mixers, automated or manual mixing, in any mode. Series 35 offers eight channels: mono, dual mono, stereo, dual stereo, or combinations. It has quadrisonic capability, fail-safe internal power supply, and plug-in interchangeable cards.
Mfr: Ramko Research
Circle 49 on Reader Service Card

Circle 30 on Reader Service Card

OSCILLOSCOPE CONVERTER

● Almost any oscilloscope may be converted into a real time audio analyzer with the ARA-412 audio response analyzer. Although it is not an impulse analyzer, it can be used to analyze all types of sound which have a less significant time framework. The device covers the audio frequency range from 40 Hz to 16 kHz, with 27 parallel bandpass filters. It has an input impedance of 100,000 ohms high level and 250 ohms microphone level.

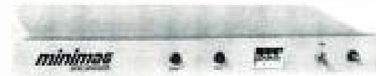
Mfr: Communications Company, Inc.
Price: $1,295.00.
Circle 50 on Reader Service Card

SLIDE FADER CONSOLES

● Flexibility is the aim of this slide fader console. The user may have his choice of slide fader brand, subject to availability of the mixer and depth provided under the panel. Featured in the unit are four switchable inputs per fader, user adjustable for mic or high-level; four muting relays with programming board for feedback-free origination from four locations; 104 dB mic-to-program output; shielded PC mixing bus; built-in cue/talkback system; built-in headphone amplifier. The console is available in 8, 10, and 12 channel versions in dual mono and stereo configurations.

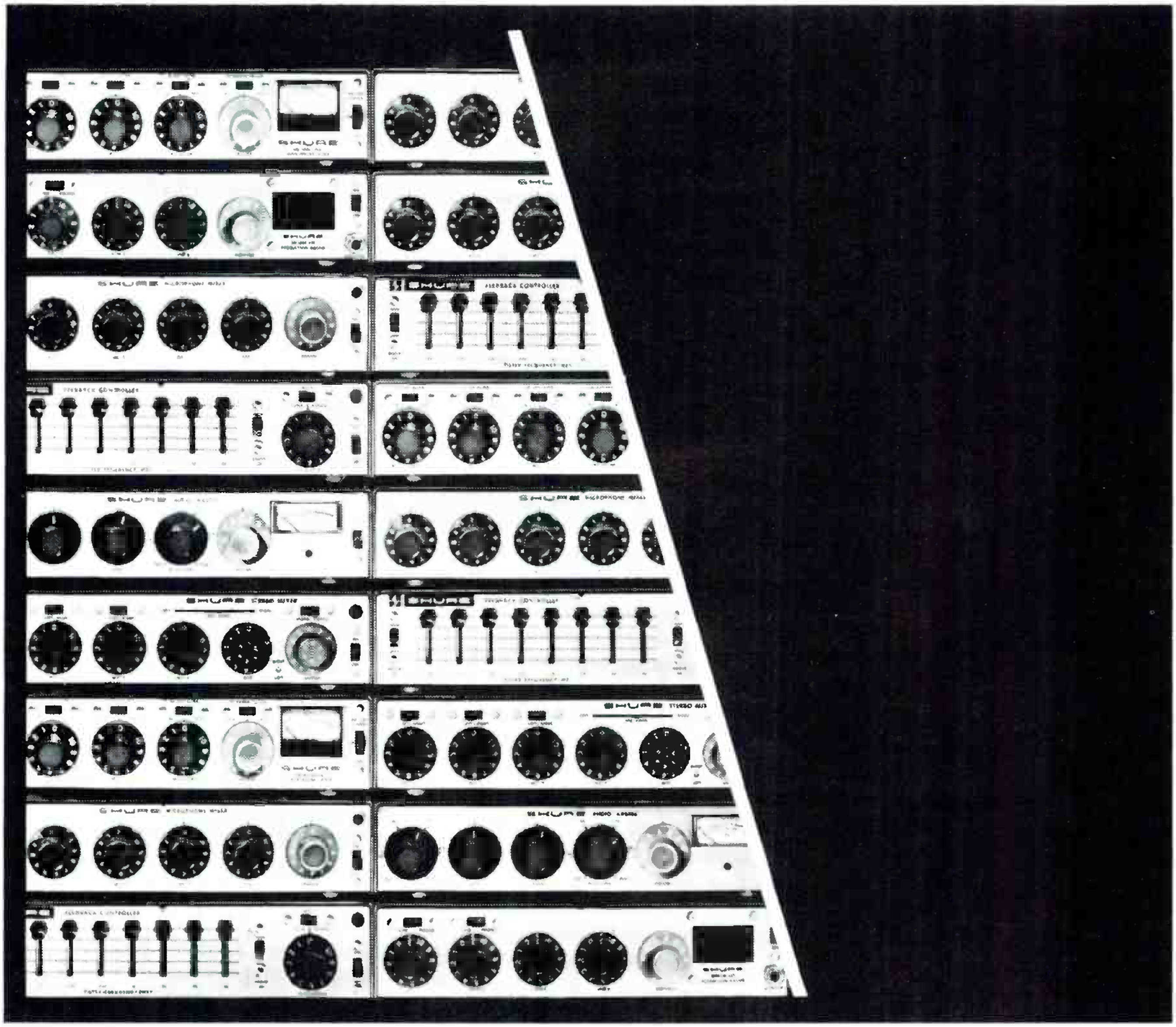
Mfr: Ampro Corporation
Price: Start at $2,695 (8 channel dual mono board)
Circle 52 on Reader Service Card

ELECTRONIC SYNCHRONIZER



● All essential functions required for the synchronization of magnetic tapes are provided by the compact Minimag®, measuring only 19 inches wide x 1¾ inches high x 12 inches deep. The unit will interlock any two tape machines, video, multi-channel, sprocketed or unsprocketed audio; two or more Minimags can be used to synchronize additional tape machines. Designed for rack mounting, Minimag is supplied with a built-in code generator. It has a capture range of ±50 seconds and will maintain sync, or variable offset for any length of time regardless of tape stretch or shrinkage. If tapes are within 50 seconds of sync, it will adjust motor control voltage automatically until they are in perfect sync, with or without offset.

Mfr: Automated Processes
Price: Under $2,000.
Circle 51 on Reader Service Card



It's a mod, mod, modular world.

Simplify, simplify! Instead of paying more for bigger, bulkier audio control components, *pay less* for compact Shure modular components that — singly or in combination—handle critical functions flawlessly. Cases in point: (1) the *M67 and M68 Microphone Mixers,* the original high-performance, low-cost mixers; (2) the *M610 Feedback Controller,* the compact component that permits dramatically increased gain before feedback; (3) the *M63 Audio Master,* that gives almost unlimited response shaping characteristics; (4) the *M688 Stereo Mixer,* for stereo recording and multi-source audio-visual work; (5) the *M675 Broadcast Production Master,* that works with our M67 to create a complete production console (with cuing!) for a fraction of the cost of conventional consoles; and (6) the *SE30 Gated Compressor/Mixer,* (not shown above) with the memory circuit that eliminates "pumping." For more on how to "go modular," write for the Shure Total Communications Components Catalog No. AL280.

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SHURE®

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R. S. MINTZ

Noise Considerations in Audio Amplifiers

This article discusses some of the ramifications of input noise in audio amplifiers, as well as the measurement, effects, and elimination. A microphone amplifier is also described.

BACKGROUND HISS has always plagued the professional sound recordist. Over the past decade or so, great strides have been made in the supression of this unfortunate aspect of high-fidelity sound reproduction, but the unequivocal laws of nature state that there will always be some noise in any sound recording or reproduction with its basis in electronics. The reason for this fact lies in the nature of electrons and matter.

Take a simple resistor. It doesn't even have to be a resistor in the usual sense—anything that is made up of matter and that passes electrical current to some extent may be termed a resistor. When electrical current flows through this resistor, collisions between *electrons* (the name we give to fictitious particles bearing an electric charge of the *negative* variety) produce a random voltage which can be detected as *white noise*. This white noise has the characteristic that it can be thought of as having components of all frequencies—it is what you would get if you were somehow to connect a million signal generators together, each one generating a signal 1 Hertz (cycle per second) greater than the last one. In an audio amplifier, an input signal, composed of an electric current flowing through its source, generates some amount of white noise (or hiss) due to the bombardment of electrons on the matter that makes up the resistance of the generator (source).

Electrons collide with themselves at room temperature even without an external current passing through the matter in which these electrons exist. This self-induced electrical noise becomes greater as the temperature of a body increases, and it is due to the fact that when a body is warm, the molecules in it vibrate in a random manner. When you touch something hot, you can feel the energy of vibration of its molecules in the form of heat.

Scientists have been able to calculate with some degree of precision just how much energy is contained in a warm body. Of importance to audio engineers is an expression of this thermal energy in a body, such as a resistor, in terms of voltage or current.

As we mentioned earlier, as a body gets hotter its thermal energy increases. In electrical terms, its self-generated *noise voltage* increases.

When a body has no heat content at all, it is said to be at a tempertaure of absolute zero. This temperature corresponds to 0 degrees on the Kelvin scale and to −273 degrees on the centigrade scale. Fortunately, the Kelvin scale and the centigrade scale have units of the same magnitude, so that one may easily convert from °K to °C by adding −273 degrees to the Kelvin temperature.

No resistor on earth is ever at absolute zero (−273°C.), but if one were to be at that temperature then there would be no thermal noise generated in it. With resistors found in the sound studio (including the resistance in microphone coils, transformer coils and the like) a thermal noise voltage is generated. This noise voltage exists even when nothing is connected to the resistor (or conductor) and its magnitude depends on the ambient temperature, the resistance, and on the bandwidth of interest. It is this noise voltage, together with amplifier-induced noise voltage, that prevents noiseless systems. We will examine noise in terms of the voltage that is impressed on the input of an audio amplifier.

First we must have an expression for the noise voltage generated due to the temperature of a body (resistor) alone.

H. Nyquist in 1928 came up with an expression that seems to fit the observed phenomenon, namely

$$E_n = \sqrt{4 \cdot k \cdot T \cdot R \cdot B} \qquad (1)$$

Noise voltage, (E_n), is equal to four times "k" times the temperature of the body times its resistance times the bandwidth of interest, all taken to the ½ power. Each

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R. S. Mintz is with Custom Sound Productions in New York City.

term affects the noise voltage in a resistor, as we shall see.

The noise voltage is dependent on temperature: as T increases so does E_n. The noise is dependent in the same way on the value of the resistance; as R increases, so does E_n. Naturally, since E_n is a random noise, it is distributed over all frequencies. As you increase the bandwidth you are considering, you will be considering more noise voltage. Therefore, as B increases, so does E_n. T is in °k, R is in ohms, and B is in Hertz (cycles per second). k is *Boltzman's Constant,* a constant relating energy to temperature, and is equal to 1.38 x 10^{-23} Joules per degree Kelvin. It just so happens that the amount of noise voltage generated in any resistor due to temperature is equal to the square root of the product: four times Boltzman's Constant times the absolute temperature times the resistance times the bandwidth of interest.

You now have a "resistor" connected to an amplifier. In a typical case, the "resistor" might be the secondary winding of the transformer in the microphone that is picking up the voice of Brian Jones. You want the vocalist's voice and nothing else, but the amplifier doesn't know or care anything about your wishes and desires. As far as it's concerned, there is a voltage across the resistor, and it is going to amplify that voltage as well as any other voltage that appears across that resistor. So if the amplifier itself is perfect, generating no noise (or hiss) of its own, you are going to get some. How much? From EQUATION 1,

$$E_n = \sqrt{4KTRB}$$

Let us assume typical values for a recording studio situation. The ambient temperature is 20°C, even though the producer is sweating. The resistance of our microphone is 80 ohms and it's connected directly to a perfect amplifier without picking up any voltage from induced currents along the wires to the control room. We are interested in a bandwidth of 0 Hz to 20,000 Hz. Substituting these values in our formula, we get in a noise bandwidth of 2 x 10^4 Hz:

$$E_n = \sqrt{4 \times 1.38 \times 10^{-23} \times 293 \times 80 \times 2 \times 10^4} \quad (2)$$

(We use exponential notation for ease of calculation)

$$E_n = \sqrt{258777.6 \times 10^{-19}} = \sqrt{2.58 \times 10^{-14}} \quad (3)$$

$$E_n = \sqrt{2.58} \times 10^{-7}$$

$$E_n = 1.6 \times 10^{-7} \text{ or } .00000016 \text{ volts}$$

This is equal to 0.16 μV, or approximately −136 dBV.

Thus, under these conditions, even a *perfect* amplifier would "see" a noise voltage of −136 dBV; no amplifier could have a lower input noise voltage. Unfortunately, no amplifier is perfect—not only do real amplifiers have a certain amount of hiss which they generate internally, but they also send out a (hopefully small) amount of current which travels through the source resistance and generates even more noise in it (which the amplifier sees, too, and amplifies). So we must add to the theoretical minimum a value corresponding to the amplifier's own generated noise voltage and also a value of the noise voltage generated in the resistance as a result of the voltage drop across it due to the noise current flowing through it.

Thus, the total amount of actual input noise in real amplifiers can be expressed as

$$E_T = E_{n(a)} + E_n + I_nR_s \quad (4)$$

The total noise voltage E_T is the sum of the amplifier's self-generated noise voltage $E_{n(a)}$ plus the thermally-generated noise in the source resistance E_n plus the product of the noise current times the source resistance I_nR_s.

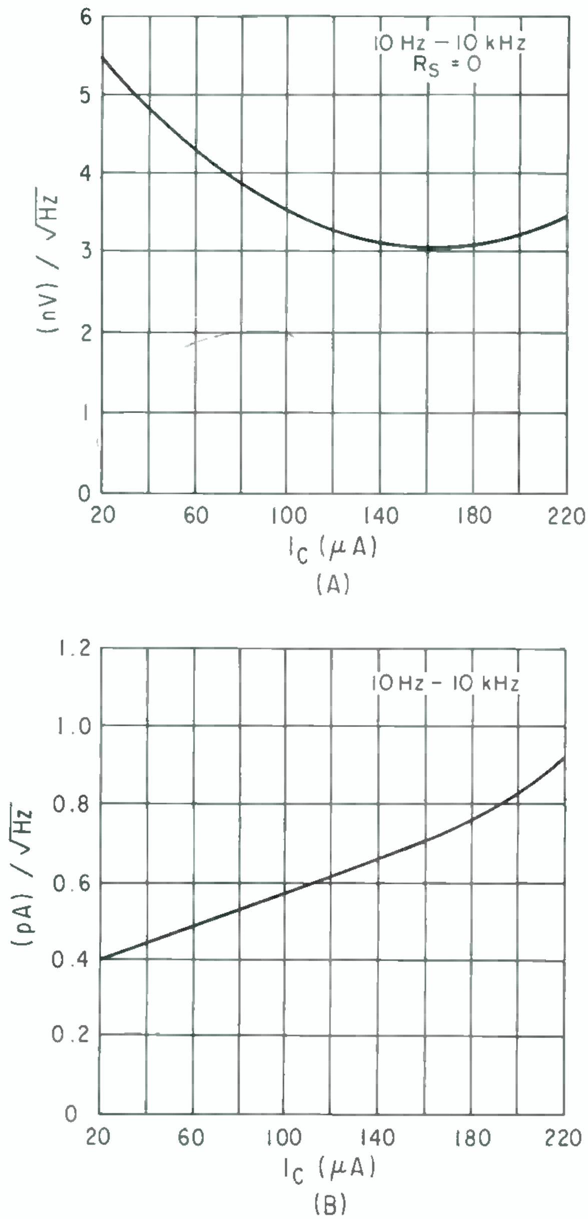

Figure 1. At (A) wideband equivalent input noise voltage versus collector current is shown. At (B) it is current that is shown.

Now that we have the theory under our belt, let us take a look at the real world in terms of what is in the control room:

FIGURE 1 shows the E_n and I_n specifications for a currently available audio amplifier of the integrated-circuit variety. The National LM381AN was chosen because the specifications are published by the manufacturer and because the i.c. exceeds the level of input noise reduction that has been achieved with discrete transistors.

As shown in FIGURE 1, the device's input noise current increases as the collector current of the input stage is raised. However, the input noise voltage *decreases* as the collector current of the input stage is raised to a value of 170 microamperes and then begins to increase again. Clearly, the optimum input noise voltage occurs at an input-stage bias level of 170 microamperes. If the source resistance is less than 3,000 ohms, the best overall noise performance will be achieved with an input-stage bias current of 170 microamperes.

Let us consider what the total input noise voltage will be with our source resistance of 80 ohms in a control room with the thermostat set at 20 degrees C., following the manufacturer's recommendations regarding input biasing and single-ended use of this typical low-noise i.c. amplifier:

From EQUATION 1 we know that the noise generated thermally in the source resistance will be, in a 10kHz bandwidth.

$$E_n = 4kTRB \quad B = 10{,}000 \text{ Hz}, \; R = 80 \text{ ohms}, \; T = 293°K$$

$$E_n = \sqrt{4 \times 1.38 \times 10^{-23} \times 293 \times 80 \times 10^4}$$

$$E_n = \sqrt{129388.8 \times 10^{-19}}$$

$$E_n = \sqrt{1.29 \times 10^{-14}} \text{ or } .000000113 \text{ volts}$$

This is equal to 0.113 microvolts, or approximately −138.9 dBV.

In order to find out the actual total equivalent input noise voltage under these conditions for an operating LM381AN-type amplifier, we must add the amplifier's self-generated noise voltage plus the product of its noise current times the source resistance. Then the total equivalent input noise voltage for this amplifier is, from EQUATION 4

$$E_T = E_{n(a)} + E_n + I_n R_s$$

From FIGURE 1, we can substitute values for $E_{n(a)}$ and I_n:

$$E_{n(a)} = e_n\sqrt{Hz};\; e_n = 3 \times 10^{-9} V/\sqrt{Hz} \qquad (5)$$

$$I_n = i_n\sqrt{Hz};\; i_n = 7.2 \times 10^{-13} A/\sqrt{Hz} \qquad (6)$$

Plugging in these values, we obtain for an expression of the system's total wideband equivalent input noise voltage:

$$E_T = \sqrt{[(3x10^{-9})^2 + (7.2x10^{-13}x80)^2]x10^4 + 1.29x10^{-14}} \qquad (7)$$

$$E_T = \sqrt{(9x10^{-18} + 3.31x10^{-21})x10^4 + 1.29x10^{-14}}$$

$$E_T = \sqrt{9x10^{-18}} + \sqrt{.00331x10^{-18}x10^4} + \sqrt{1.29x10^{-14}}$$

$$E_T = \sqrt{9x10^{-14}} + \sqrt{.003x10^{-14}} + \sqrt{1.29x10^{-14}} \qquad (8)$$

$$E_T = \sqrt{10.293x10^{-14}} = \sqrt{10.293}x10^{-7}$$

$E_T = 3.21\ 10^{-7}$ V or 0.321 microvolts. 0.321 μV is approximately −129 dBV.

A comparison between this value and the theoretically perfect amplifier shows a difference of 0.2 microvolts.

EQUATION 8 shows the contribution of the noise term associated with the amplifier itself, the source resistance voltage drop due to input noise current (a function of the amplifier design) and thermally-generated noise in the source resistance. Clearly, the larger the resistance the greater will be the noise terms associated with it, namely I_nR_s and E_n. Threfore, it is convenient for low-noise operation to keep the source resistance as low as possible, in general.

In the previous case, the amplifier noise-voltage was optimized by increasing the input stage bias current. If the source resistance is greater than 15,000 ohms, the standard input bias current of 18 microamps will yield the best noise performance. See FIGURE 1.

There are a number of transformers on the market today which can provide gain prior to the first active stage of the amplifier. The gain of the input transformer is equal to its effective turns ratio. However, as the turns ratio increases so does the secondary coil's resistance, especially in miniature transformers. Aside from any other drawbacks (such as distortion and transformer noise) the input noise will increase as the R_s terms increase in the E_T equation. But due to the added voltage gain of an input transformer, the *amplifier's* gain to produce the same output level that the amplifier would produce without the input transformer is *less*. Let us take a look at a typical case:

An amplifier operates at a gain of 100. With a source resistance of 80 (corresponding to a typical 600-ohm microphone transformer secondary) at an ambient temperature of 20 degrees C., we have, as before, a total input noise voltage in a 10 kHz. bandwidth, of

$$E_T = \sqrt{(e_n^2 + i_nR_s^2 + 4kTR_s)\,B} = 0.321\ \mu V \qquad (9)$$

With an amplifier gain of 100, the output noise voltage will be 32.1 microvolts. Now we add the step-up transformer. Let us pick a typical miniature input transformer with a turns ratio of 1:10. The secondary coil resistance of this transformer, a Beyer TR/BV 35570, measures 10,000 ohms. Its voltage gain is 10, so we will turn the amplifier's gain down to 10. We will use the standard biasing of our op amp, since we wish to obtain optimum performance for a 10k source. Our total input noise is now

$$E_T = \sqrt{[(5x10^{-9})^2 + (4x10^{-13}x10^4)^2 + 4x1.38x10^{-23}x293x10^4]x10^4}$$

$$E_T = \sqrt{(25x10^{-18} + 16x10^{-18} + 1617.36x10^{-19})x10^4}$$

$$E_T = \sqrt{(25x10^{-18} + 16x10^{-18} + 161.736x10^{-18})x10^4}$$

$$E_T = \sqrt{202x10^{-18}x10^4} = \sqrt{202x10^{-14}} = \sqrt{202}x10^{-7}$$

$E_T = 14.2x10^{-7}$ 1.42 microvolts

Since the gain now has been reduced to 10, the output noise voltage is 14.2 microvolts. Clearly, here is an improvement as far as the output of the amplifier is concerned. The best results would be obtained with an input transformer that could supply sufficient gain to drive the input stage to a point near its clipping point (in this case, 300 millivolts) while maintaining the lowest possible secondary coil resistance.

It is hoped that the reader has been given a basic insight into the problems and solutions regarding the elimination of noise in audio systems. Specifications alone do not tell the whole story unless one follows through with an analysis of them. For example, in the last case, where an input step-up transformer was used to achieve better overall noise performance, the thermal noise contributed by the transformer itself was six times greater than the noise voltage term associated with the amplifier itself. The thermal noise of the input transformer was the major contributor of noise, so that we may say that the amplifier itself was almost "perfect." ■

STEPHEN H. LAMPEN

Build a Super Window for Your Studio

Here's a fairly simple studio construction job that can pay dividends in the improved communication that will result between the control room and the studio.

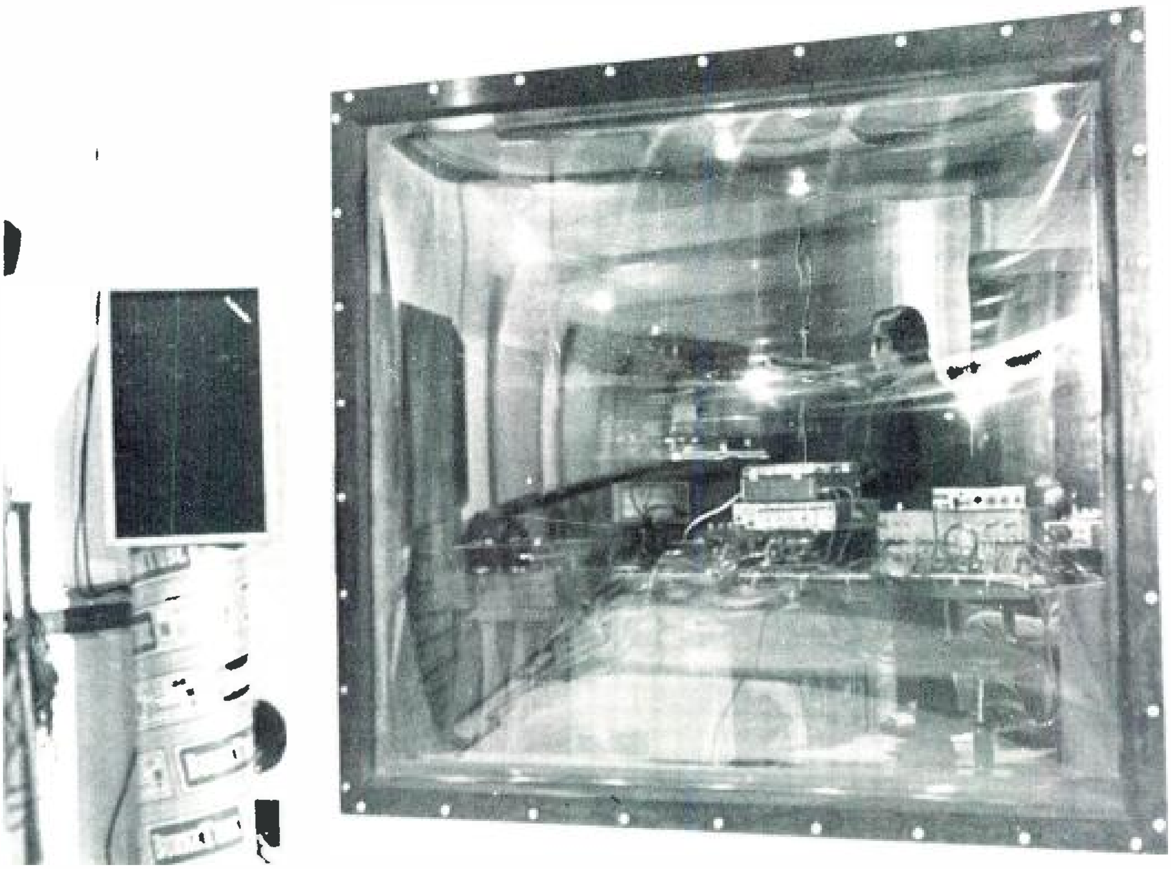

Figure 1. The window as seen from the studio. The view is into the control room.

When Stephen Hill, owner and chief engineer of *Celestial Sound*, was completing work on his new studio in San Francisco, the problem of a studio window came up. "I was looking for something futuristic and spacy," he said. Drawing on his extensive background in architecture and inspired by the award-winning Cuban exhibit at the Montreal Expo, he designed and installed an unusual and, for recording studios at least, unique window. (Figure 1)

Formed from two pieces of acrylic plexiglas, and shaped

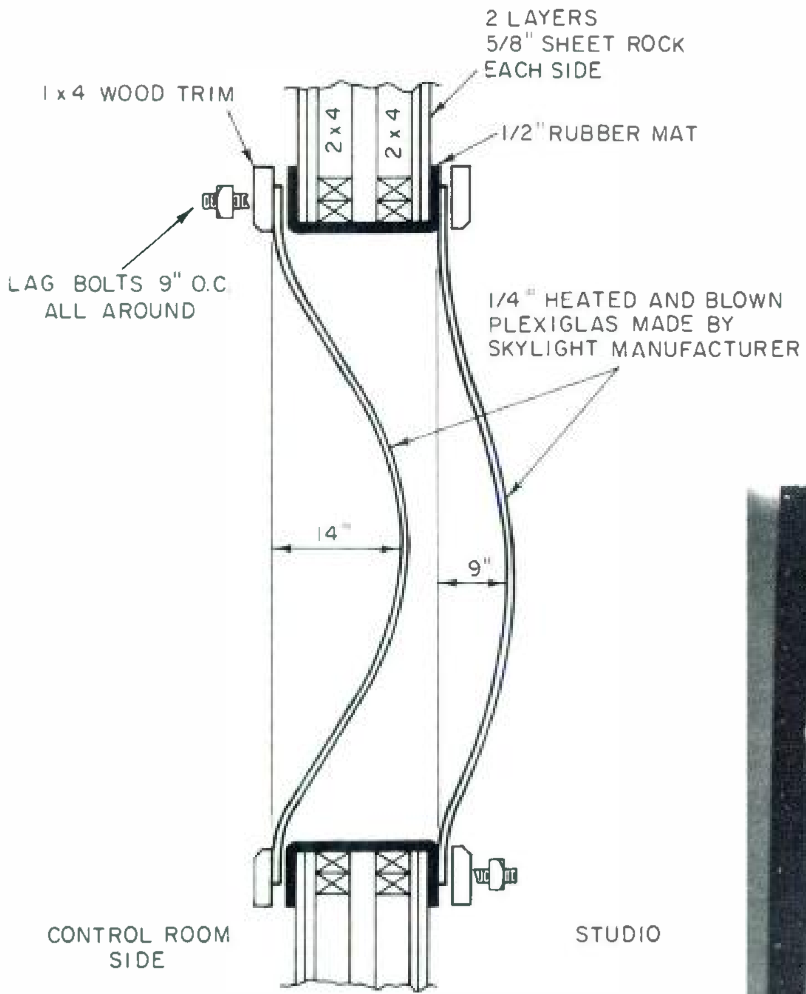

Figure 2. The construction drawing side view. From the front, the window specified is 6-feet, 4-inches across, and 5 feet, 4-inches top to bottom. The window has good dispersion and high rigidity:weight ratio. The windows cost $85 each and total cost including construction was $200.

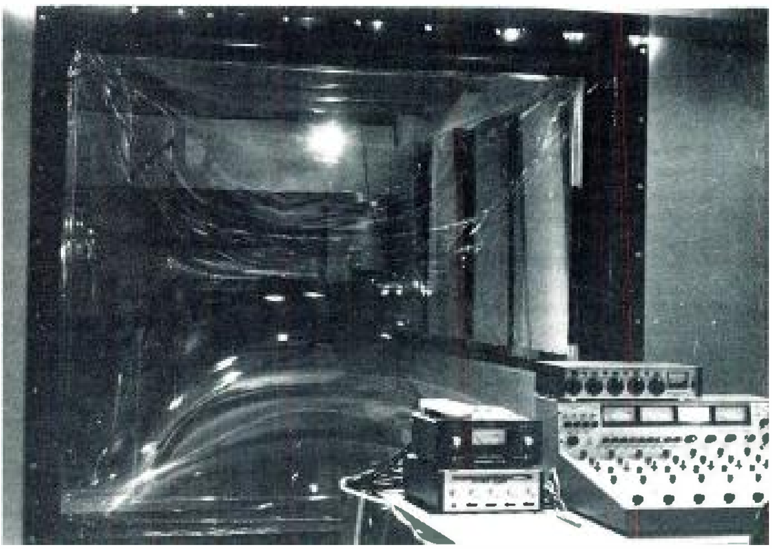

Figure 4. The view into the studio from the control room. The entire room is visible.

Figure 3. The bulging of the window can be seen in this illustration. The studio equipment has been lit to reflect into the window.

into a partial "bubble," the window bulges out from the control room into the studio. The two original sheets were blown into shape by a skylight manufacturer who can blow a piece of acrylic up to a half-sphere. In this case, the control room sheet was blown to a center depth of fourteen inches, the studio sheet to nine inches. There is a slight amount of visual distortion in the window. "That distortion," said Steve, "is due to the fact that the original sheets had flaws in them. If they had started out with optically perfect sheets, the window would also be perfect."

Construction of the window and its installation was quite simple. (FIGURE 2). Two-by-fours were cut and double-butted to already existing two-by-fours. Two layers of ⅝-inch sheetrock were placed on each side, a ½-inch rubber mat stretched around the edge. The blown plexiglas was then laid on each side of the mat (which framed the entire window) and fastened on with lag bolts on 9-inch centers through a frame of one-by-four with a beveled edge for a frame effect. The finished dimensions of the window were 6'4" by 5'4" but windows of almost any dimension can be produced due to the high rigidity/weight ratio of plexiglas.

The effect on the viewer or performer is uncanny. In the control room the window draws the eye out into the studio. In this way, both rooms seem like one and the performer does not feel totally isolated. The window ceases to be a partition between the rooms and tends to amalgamate them. "You don't feel any change of environment from one room to the other," said Hill.

From an acoustic standpoint, the window tends to disperse sound in the studio while, in the control room where it is concave, the focusing effect on the sound can only be sensed when one is a few inches away from the window. "The studio has been made to look like an airline with the window as a porthole," continued Mr. Hill.

This feeling is increased by the use of many curved surfaces, flat grey side panels and by airline seating for guests. Steve does not claim the seating idea to be original. "I got the idea from the *Record Plant* in Sausalito. They got the last three-seat sets, so I got a two-seat one." At $10 per seat, they are probably the most comfortable seating one could think of—if you can find them. And, naturally, they come with built-in ashtrays, seating adjustments and plush padding.

The guests, artists, producers—or whomever—can sit in comfort looking out into the studio through this gigantic porthole. And, considering the costs involved ($85 per sheet of plexiglas, total expenditure of around $200), it makes one want to chuck his old, flat plate-glass windows for something really modern. ■

MICHAEL RETTINGER

Recording Studio Acoustics, part 2

In this installment, the author discusses monitoring room acoustics from both the practical and theoretical viewpoint. Graphs for calculation are included.

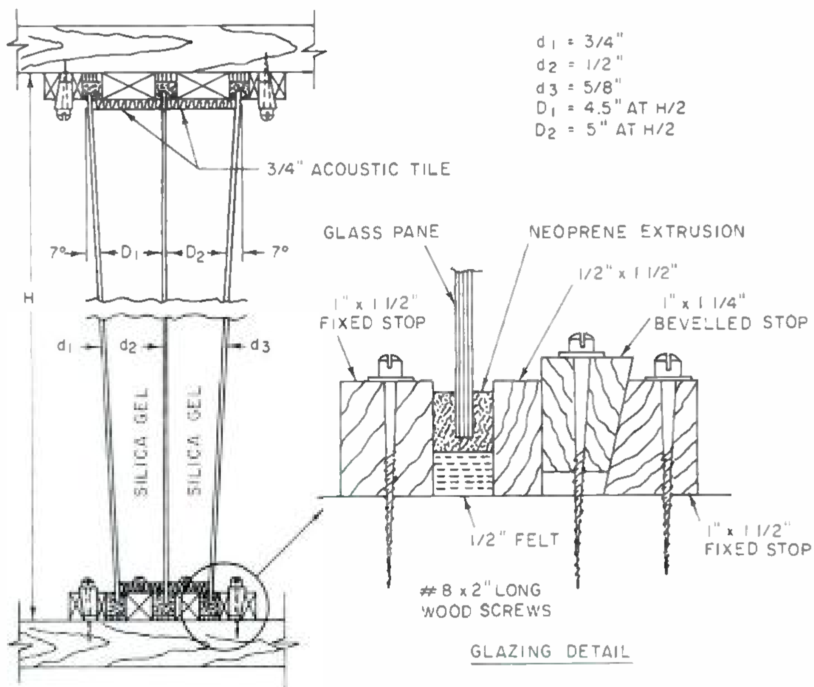

Figure 1. Details of construction of triple-pane sound-retardant window used in monitoring room. The values for the glass thicknesses shown on the figure refer to a large (8 feet x 16 feet) window in a Hollywood motion picture scoring stage.

A COMMON ELEMENT OF RECORDING STUDIOS is a monitoring room or monitoring rooms. For the large scoring stages in motion picture productions, when simultaneous recordings are to be made of the instrumental music on the stage, a vocalist in his vocal booth, and a chorus in a group enclosure, with each performer watching the pertinent scene on a large picture screen and each mixer observing his performers, three monitoring rooms have been used simultaneously to reduce the very expensive recording time to a minimum and to achieve optimum results.

A monitoring room is intended for checking the quality of the music played in another room, either adjoining it or linked with it over closed-circuit television. The operation in the room may involve level controls of various band sections, the addition of a reverberatory note to portions or all of a rendition, tonal modifications of passages, or the use of compression, limiting and expansions of a part of the program. It is an enclosure which is intended not to stamp its own acoustical characteristic on the reproduced signals, but one which has a nearly flat reverberation *time characteristic*, that is, variation of reverberation time with frequency.

The achievement of a frequency-invariant reverberation

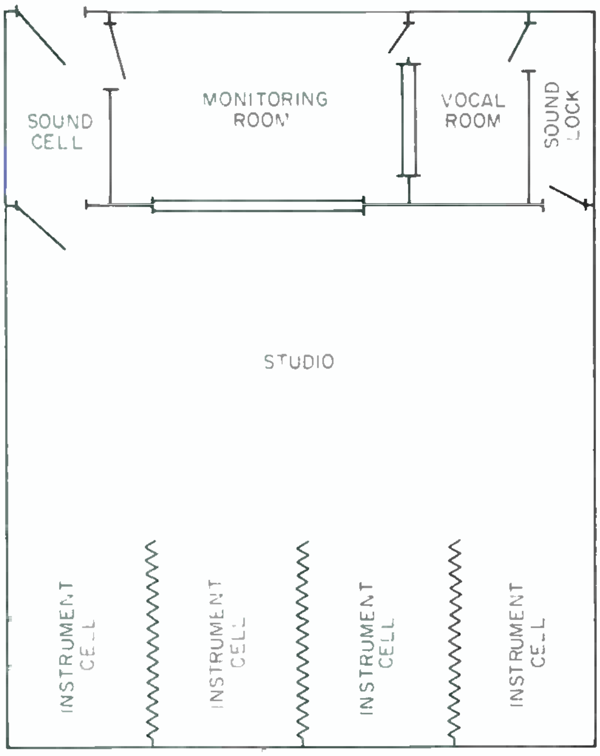

Figure 2. This illustration shows sound-locks, or vestibules, for noise control, and folding partitions for the quick establishment of instrument cells.

time in a monitoring room is frequently made difficult because not enough space has been alloted for this enclosure, so that according to statistical acoustics, too few normal modes come to exist in any given low-frequency interval. Unfortunately a clear discussion of this phenomenon requires lengthy mathematics, and will not be given here. (The interested reader will find the subject described in the writer's *Acoustic Design and Noise Control*, available through the book sale page of **db**.)

Considering the fact that the product of most recording studios is reproduced in a living room, it stands to reason that this enclosure should not be much smaller, or have a floor area less than 400 square feet. So large a room will permit the concealed installation of medium-sized two-way speaker systems, not only in front of the mixer but also at the locations of the quadriphonic units. And if and when video-disc reproduction becomes an economic operation, a space of this size will likewise accommodate a television set or two.

In the acoustic design of a monitoring room the following details of construction deserve special consideration:

- The floor of the room should be made higher than the studio floor to allow a better view over the performers and to hide the multiplicity of cables entering the mixing console. It should be sturdy and consist of at least 2 inch x 6 inch T&G wood, with hot wax smeared in the joints to prevent squeaks later when the timber has dried out. When carpet is desired, it should be of the non-electrostatic type, preferably with a fine stainless steel net interwoven with the plastic or wool, instead of a "treated" material which tends to lose its anti-static quality.

- Monitoring room windows must follow the same acoustic mass law which pertains to walls, partitions, floors, ceilings, doors, and other common sound barriers, in that for a given construction greater insulation is secured by increasing the surface density, that is, the mass per square foot, of the space-divider. Hence, when sound-transmission losses in excess of 50 dB at 500 Hertz are to be achieved for a window, a triple-pane unit should be employed. For the same surface density, it allows for thinner panes than a double-pane unit with 4-inch separation between panes. Thus, if, say, two 1-inch thick panes are required for a double-pane window, a triple unit may be constructed with three ⅝-inch lights, with the following advantages: (1) the price of ⅝-inch thick sheets is less than half that of 1-inch ones; (2) there is much less risk involved in cutting a ⅝-inch sheet of glass than a 1-inch thick one; (3) the extra air-space provides a somewhat greater sound-insulation, particularly when every pane is properly floating in a soft Neoprene surround and the walls between the lights are treated with 1-inch thick sound-absorbent. To avoid like resonance frequencies on part of the lights, it is desirable to make them of a slightly different thickness while yet retaining the required overall mass per square foot. FIGURE 1 shows details of construction of a triple pane window employed for a motion-picture studio sound-recording complex.

- The sound-insulation of a door is not unlike that of any other solid barrier, like a window—except for the inevitable sound-leaks about the perimeter of the unit—in that its 500-Hertz sound-transmission loss must be closely equal to

$$TL = 23 + 14.5 \log M$$

where M is the surface density of the door.

Two medium density doors, weighing in the order of five pounds per square foot each separated by a *sound-lock* or vestibule, are at all times preferable to one high-density door weighing fifteen or even twenty pounds per square foot. There is simply no substitute for the structural isolation of such a double door which almost, but not quite, permits the arithmetic addition of the individual door's sound-transmission losses (rather than requiring the logarithmic addition of the TL's, where double the surface density achieves only a 4.35 dB increase in the insulation). FIGURE 2 shows studio with sound locks.

There are no magic sound-insulators, patented super sound-barriers which do not have to obey the acoustic mass law stated previously. There are, however, very useful trade-offs when the boundary represents a composite barrier, such as a wall with a window and door. It is sometimes quite economical to over-insulate a wall, that is, to give it a sound-transmission loss greater than required, so that other elements in the wall may have a smaller TL.

As an example, consider a monitoring room wall which is to exhibit a noise reduction of 50 dB at 500 Hertz (when the ratio of S/A* is unity, (S) being the area of the wall and (A) the total absorption in the monitoring room). Such a wall, according to the acoustic mass law, has to have a surface density of 73 pounds per square foot, as is represented by a relatively inexpensive 6-inch thick concrete block wall. A like surface density for even a small window—say, 10 percent of the total wall area—would require a total glass thickness of 5-inches, assuming glass weighs 15 pounds per square foot 1-inch thick. However, by using a slightly thicker concrete block wall, so that its surface density comes to 100 pounds per square foot and its TL rises to 52.5 dB at 500 Hertz, the window may exhibit a TL 9.5 dB less, or 43 dB at 500 Hertz, for

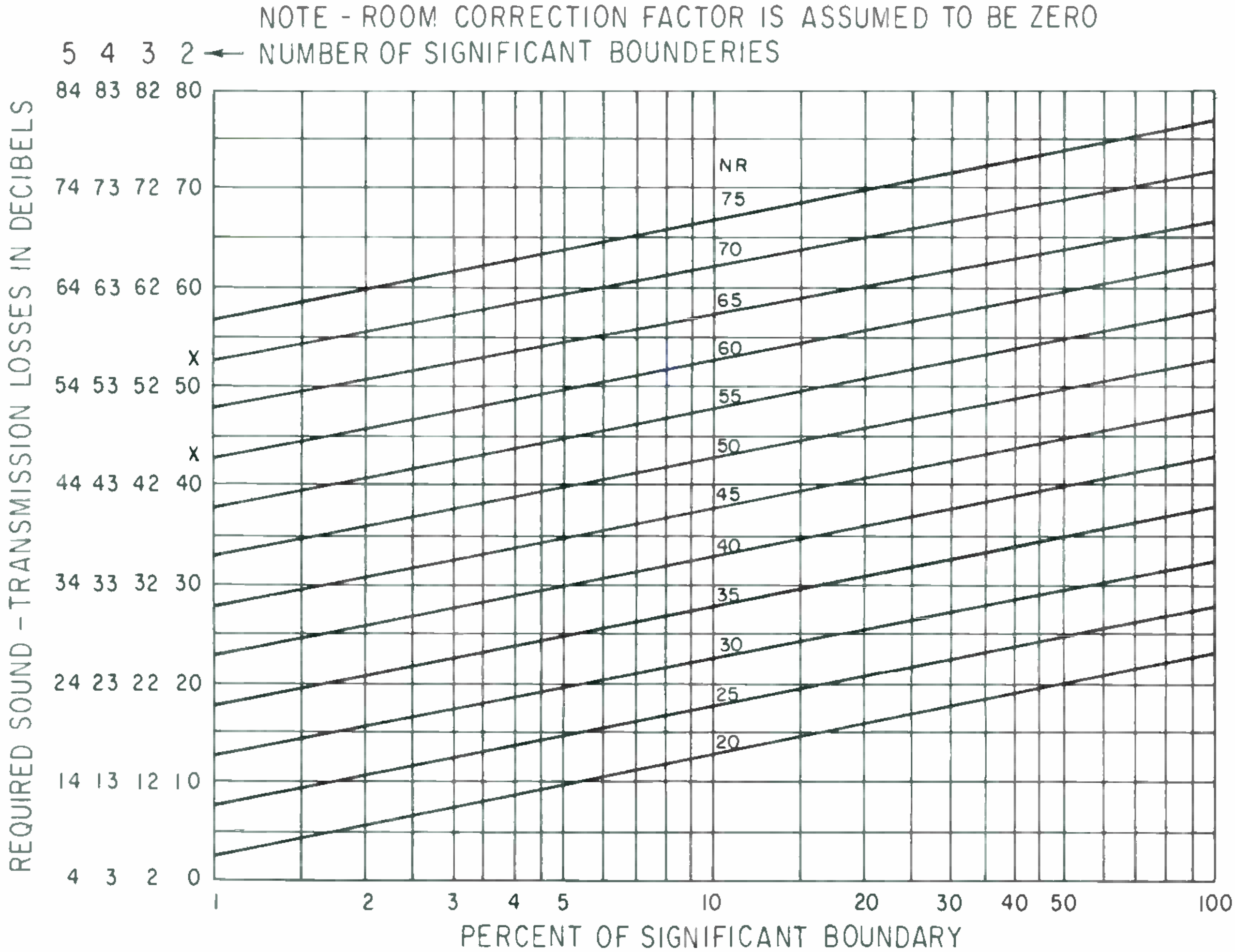

Figure 3. This graph provides quick determination of the required sound transmission losses of elements in a composite barrier to provide various noise-reductions. The abscissa refers to the per cent area of an individual element.

the composite barrier to exhibit a sound insulation of 50 dB at 500 Hertz.

The equation which links the various TL's of a composite barrier is given by

$$TL_i = 10 \log k_i + NR + 10 \log N$$

where k_i = percent of total significant boundary area (0.1 for window in example)

N = number of insulators of wall (2 for example)

NR = required noise reduction at frequency f (500 Hertz in example)

FIGURE 3 shows a chart for the rapid determination of the required TL's of a composite barrier; the dots and X's refer to example.

*Often called room-correction factor

From numerous loudspeaker response measurements made in monitoring rooms, this writer has often observed a lack of low-frequency support which is not apparent from the response curve furnished by the manufacturer. Considering that most mixers are at least five feet from the emitters, and therefore well within the reverberant sound field, the reasons for this lack of bass may be one or more of the following conditions:

- The room walls, floor, and ceiling are inadequately stiff and too light-weight, allowing diaphragmatic action and large-scale low-frequency absorption.
- The loudspeaker cabinets are not in an infinite baffle but radiate into a solid angle greater than 6.28 steradians, which often leads to pronounced interference effects. For instance, when the distance between front of cabinet and the wall in back of the cabinet comes to four feet, a dip in the response will occur at 70 Hertz and a peak will occur at 140 Hertz.
- The directivity factor Q of the mid-range and high-frequency loudspeakers is too large, that is, the sound is confined in too narrow a beam, producing too high a sound-pressure level at these frequencies at the mixer's position.

A monitoring room of 4,000 cubic feet should have a reverberation time of 0.4 seconds at all audio frequencies. While the side walls should be kept moderately live, the wall behind the mixer should carry a highly absorptive treatment since all wall, ceiling, and floor reflections terminate there. For this reason, several Hollywood monitoring rooms have applied anechoic fiberglass wedges to this boundary, adequately protected by screens and railing. The panels of the mixing console should carry vibration-damping compound inside the instrument, and the partition between monitoring room and stage should have a sound-transmission loss at 500 Hertz of no less than 55 dB. ■

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TASCAM REVERBS—$500; Tascam mixing consoles—$2,350; Tascam ½-inch recorders—$2,750; Tascam 8-track recorders—$4,600. All shipped prepaid/insured, including *free* alignment/equalization/bias/calibration. **Music & Sound, Ltd., 11½ Old York Rd., Willow Grove, Pa. 19090. (215) 659-9251.**

AMPEX RECORDERS, mint condition; in consoles. One AG440B-1, $1,500; two AG440B-2, @ $2,000; other items. **Audio International. (201) 322-7269.**

PHILLIPS MD MIXING CONSOLE, 16-in/4-out; switchable sensitivity line/mic; vertical slide attn's, echo send and return levels; post- and pre-echo select, 8 equalizers; tackback, monitor switching, mono output, all connections UA type connectors. Contact **T. Kulchar, TPC Inc., 445 Melwood St., Pittsburgh, Pa. 15213. (412) 682-2300.**

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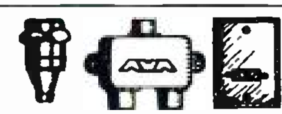

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CRYSTAL SYNC OSCILLATOR . . . $165.00. Accuracy: .0005% from —22 degrees to +140 degrees F. Fits internally in Nagra III and IV and other tape recorders. Use of state-of-the-art technology makes this kind of accuracy at this price possible. **Audio Scientific, Inc. (212) 258-1687.**

AMPEX SPARE PARTS; technical support; updating kits, for *discontinued* professional audio models; available from **VIF International, Box 1555, Mountain View, Ca. 94042. (408) 739-9740.**

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STUDIO SOUND—Europe's leading professional magazine. Back issues available from October '72. $1 each, postpaid. **3P Recording, P.O. Box 99569, San Francisco, Ca. 94109.**

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WISCONSIN: RM SOUND & CO. has a few Electro Sound ES505 tape recorders in stock and ready for delivery. See for yourself. Quality at a lower price. **RM Sound & Co., Box 72, New Berlin 53151.**

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GO 24-TRACK! Buy this 24-16-8-track machine and don't spend the big bucks. With one machine. you can offer all three configurations for what you'd expect to pay for a 16-track alone. This is the original MCI with individual meter modules. Complete with auto-locator, digital timer, and Kendun modifications to eliminate punch in/out clicks and reduce hum. Reason for sale: going to a machine with film lock. Price: $22,000. Many extra cards, motors, other spares, available. A real bargain for a one-machine studio; maintained in A-1 condition by **Kendun Recorders/Kent Duncan. (213) 843-8096.**

CUSTOM TAPE DUPLICATION, 8-track and cassettes. We specialize in small runs. **Dick Walen, Custom Audio Sound Service, 4226 Robert St., Red Wing, Minn. 55066.**

SOLID-STATE AUDIO MODULES. Console kits, power amplifier kits, power supplies. Octal plug-ins—mic. eq., line, disc, tape play, tape record, amplifiers, Audio and tape bias oscillators. Over 50 audio products; send for free catalog and applications. **Opamp Labs, Inc., 1033 N. Sycamore Ave., Los Angeles, Ca. 90038.**

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WANTED: EXPERIENCED RECORDING ENGINEER/PRODUCER with own following to join with new fully equipped, top-quality recording facility in the Los Angeles area. Salary and percentage. Send resume to **Box 101, 1120 Old Country Rd., Plainview, N.Y. 11803.**

KLUM-FM, the 40,000 watt voice of Lincoln University, announces the following openings: MAINTENANCE ENGINEER (first phone), and PROGRAM MANAGER. Address inquiries to **Dr. A. H. Ottley, Office of the President, Lincoln University, Jefferson City, Missouri 65101.**

EXPERIENCED RECORDING ENGINEER, seeking a career position with dynamic recording company. Will accept a trainee position to meet company requirements for the desired position or for a related field. Credits include classical, jazz, remote location recordings, and sound reinforcement setups. Resume available. Contact: **Lee Mitchell, 4243 E. South Shore Drive, Erie, Pennsylvania, 16511. (814) 889-2672.**

PEOPLE, PLACES, HAPPENINGS

● The **SMPTE Toronto Conference,** at the Four Seasons Sheraton Hotel, is slated for November 10-15. **Alex MacGregor** has been appointed Local Arrangements chairman with overall responsibility for conference arrangements. Session topics will include television systems, photo instrumentation, films for television, motion picture systems, small format, theater design and projection, satellites in broadcasting, cable television, television and film in education, laboratory practices, and sound recording and reproduction. A 78-booth exhibition will showcase professional motion picture and television equipment. For information, write to SMPTE, c/o Conference Coordinator, 862 Scarsdale Ave., Scarsdale, N.Y. 10583.

● Six one-and-a-half day conferences, geographically scattered throughout the country, have been arranged by the **National Association of Broadcasters.** In addition to the customary offering of news in broadcasting techniques and trends, there will be an informal evening question-and-answer session prior to each conference. Experts available will include personnel from the **Radio Advertising Bureau,** the **Federal Communications Commission, Chuck Blore Creative Services,** as well as broadcast equipment manufacturers and top NAB personnel involved in legal, regulatory, management, public relations, and engineering activities. The conferences will be held at the following times and places: New York, October 22-23, Waldorf Astoria; Atlanta, October 28-29, Hyatt; Chicago, October 30-31, Hyatt O'Hare; Dallas, November 14-15, Fairmont; Denver, November 18-19, Brown Palace; Las Vegas, November 20-21, Sands.

● **Westlake Audio** of Los Angeles, has opened an office in Nashville, Tennessee. The office will be headed by **John W. Gardner,** formerly chief engineer of **Bearsville Sound Studios.** While construction of new quarters in Nashville is under way, the company will be temporarily located at 3250 Dickerson Road, Suite 206.

● **Collin C. Chamberlain** has joined **Media IV Advertising** as account executive. Formerly trade shows and exhibits manager and advertising supervisor with **Ampex Corporation,** Mr. Chamberlain will represent full-service accounts for the agency, which is located in Sunnyvale, California. The new appointment is part of Media IV's plan to broaden their service, moving from specialized graphics/production to a wider range of advertising services.

● Two additional associates have been added to the staff of **Gilfoy Sound Studios** of Bloomington, Indiana. **Gary Carrelli,** formerly with **Omega Sound,** has joined Gilfoy as systems engineer and **Kirk Butler** has assumed the position of mixer. Mr. Butler was formerly with **Ray Stevens Sound Laboratory.** Gilfoy Studios has been conducting recording studios seminars in conjunction with **Indiana University; Jack Gilfoy** is currently working with the IU School of Music developing a new Associate of Science degree to be called **Audio Technology.** If you are interested in this program, information can be obtained either from Mr. Jack Gilfoy, Gilfoy Sound Studios, 1130 W. 17th St., Bloomington, Indiana 47401 or Dean William Christ, School of Music, Indiana University, Bloomington, Indiana 47401.

● **Otari Electric Company,** Japanese manufacturers of professional tape recording and duplicating equipment, has entered the American market with the formation of **Otari Corporation,** of San Carlos, California. According to **Brian Trankle,** marketing manager of Otari Corporation, the company was founded in 1965 by a group of former TEAC engineers, among them **Masayuki Hosoda,** now president of Otari Electric. Otari's product line includes ¼ and ½-inch professional recorders, high speed cartridge and cassette duplicators, in-cassette duplicators, duplicator quality-control monitoring reproducers, tape winders, and cassette tailoring machines. Distribution in the United States will be by professonal distributors or by manufacturer's representatives. The new American operation will be headed by **Mitsuo Takekawa.**

● Adding to the four-channel picture is the UD-4, newly developed by **Nippon Columbia,** Ltd. The discrete quadriphonic system, which was introduced on September 2, 1974 in Tokyo at a large press and trade showing, is claimed to solve the difficulties found in both the matrix and discrete systems by providing superior localization plus excellent sound quality. Nippon Columbia will shortly release ten albums of UD-4 recordings and three models of stereo units equipped with UD-4 devices, as well as a UD-4 demodulator.

● The New York City chapter of the **Society of Broadcast Engineers** is sponsoring a convention to be held at the Hilton Inn, Tarrytown, N.Y. on October 25th and 26th, from 9:00 a.m. to 9:00 p.m. both days. Exhibits, workshops, and seminars are planned. For information, contact **John Lyons** at (212) 335-1600 or **Lyn Snyder** at (212) 347-2940.

● A separate sales organization for the sale and distribution of the new Ferrograph Studio 8 professional recorder, Ferrograph Super 7 recorder, Ferrotester audio test equipment, Editall tape editing equipment and Thorens transcription turntables has been created by **Elpa Marketing** of New Hyde Park, N.Y. **E. L. Childs** will serve as marketing manager of the professional products division and **John P. King** will be technical director. Three regional sales representative firms, covering eastern, southern, and western locations, have been named to service the new organization.

● The first commercial record release in the **dbx** encoded "noiseless" format was made recently by **Hal Powell** of **Klavier Records,** a solo recital by harpist **Susann McDonald,** an interesting application of the process previously used only for professional studio tape recording. dbx processing compresses the signal upon recording and expands it on tape playback at the disc mastering stage, achieving some 35 dB reduction in tape and background noise along with improving transient response and preserving the full dynamic range of the original performance. A dbx decoding device is used in playing the record.

● **Fanshawe College** of London, Ontario has appointed **Tom Brennand** to the post of engineer and instructor in Recording Engineering and **Allan Allbutt** as producer and instructor in Music Production. Mr. Brennand has engineered popular albums through **Eastern Sound Studios.** Mr. Allbutt comes from **Leeds Music, Ltd.** Both men are part of the faculty of Fanshawe's program in Creative Electronics.

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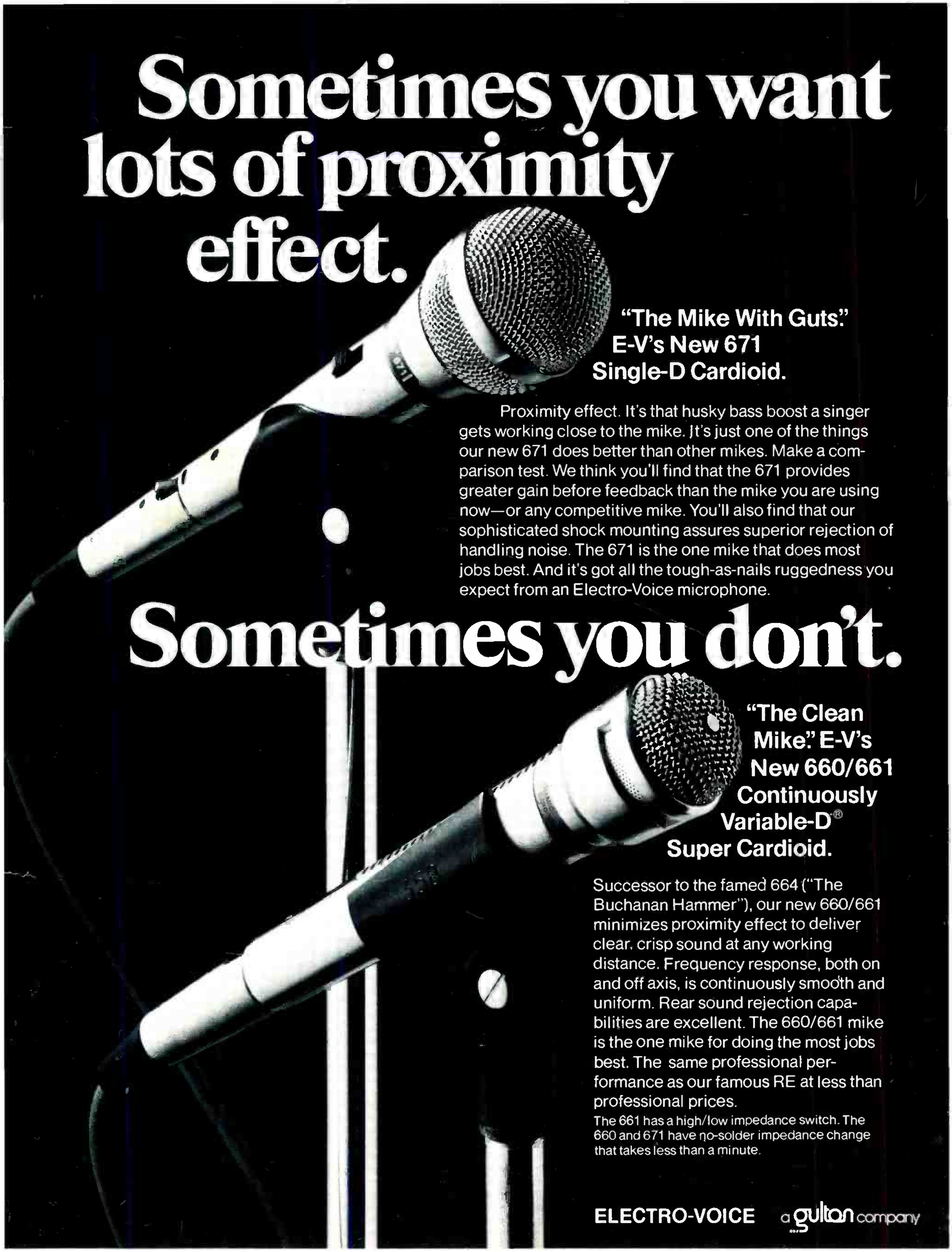

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