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December 1977
Volume 8 — Number 6

Recording engineer/producer

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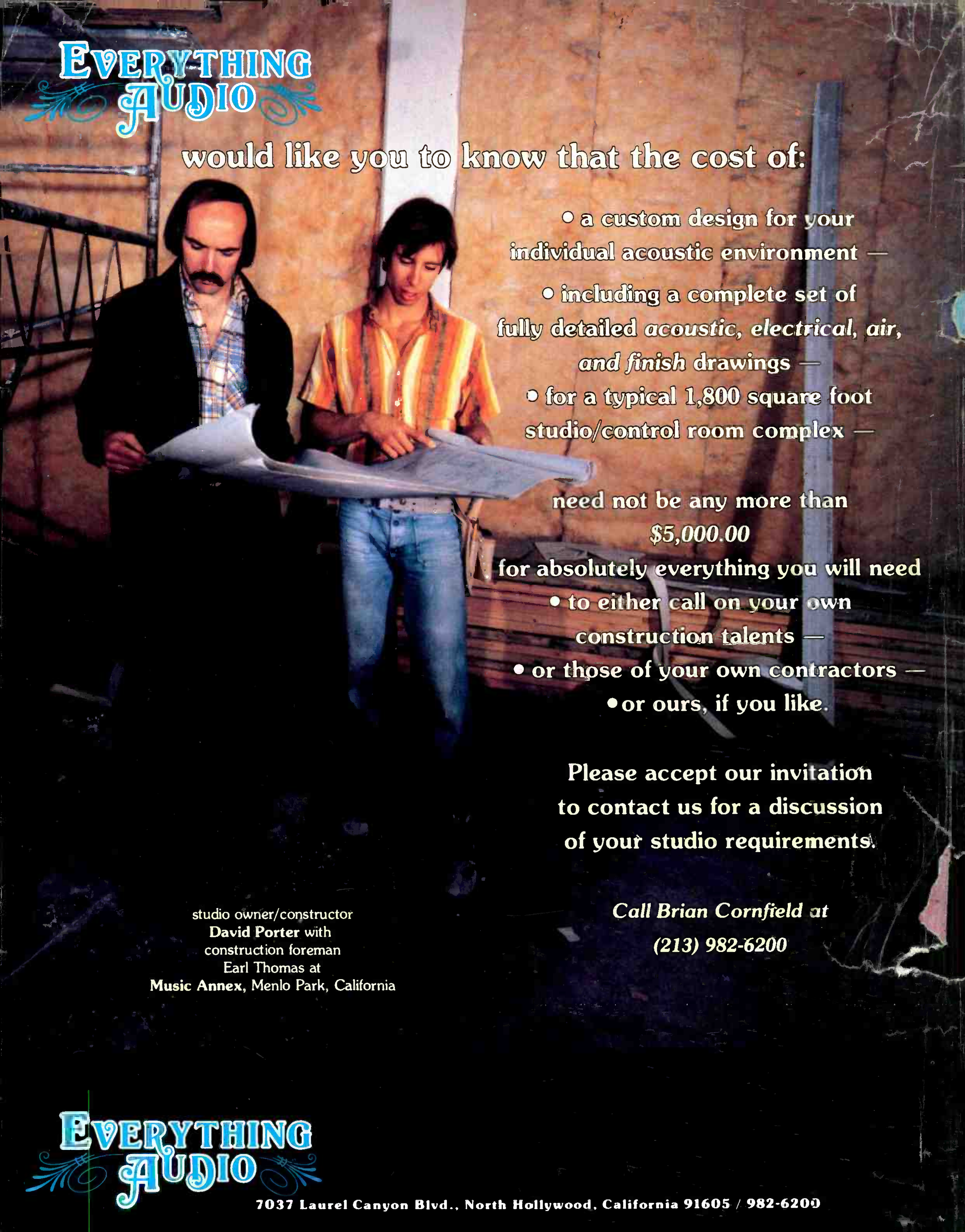

for additional information circle no. 1

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BUILDING A STUDIO CAN
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Yes, it's expensive to build a recording studio today — and, unfortunately, it's going to become

more expensive with increases in the cost of everything; materials, labor, and especially the almost hidden cost of time delays. Plumbers and electricians can't be expected to stand around without being paid while waiting for an on-site decision . . . and the costs of *"working it out when we come to it"*, can certainly be catastrophic. Once the cement truck arrives there is no turning back.

We offer the kind of planning and preparation available only from professionals who have built studio after studio. And, how many times have you heard about a room having to be rebuilt (at even greater cost) because of poor design? The lesson: experts in studio

design and construction aren't found in your local equipment store, where studio design is merely a hobby or sideline. It is impossible to build a truly professional studio from a textbook.

Excellent results can only be obtained by using proper materials in a proven plan.

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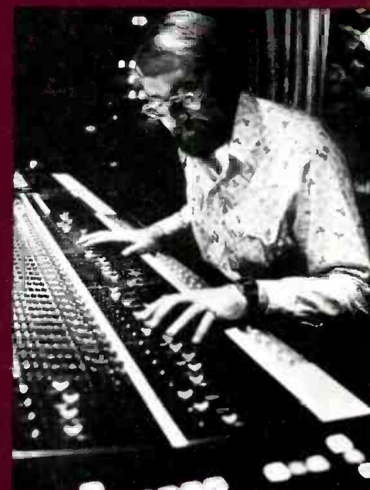

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RECORDING engineer/producer

— the magazine to exclusively serve the **Recording Studio** market . . . all those whose work involves the recording of commercially marketable sound.

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

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DECEMBER, 1977
VOLUME 8 — NUMBER 6

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cover: another of
TRICI VENOLA'S
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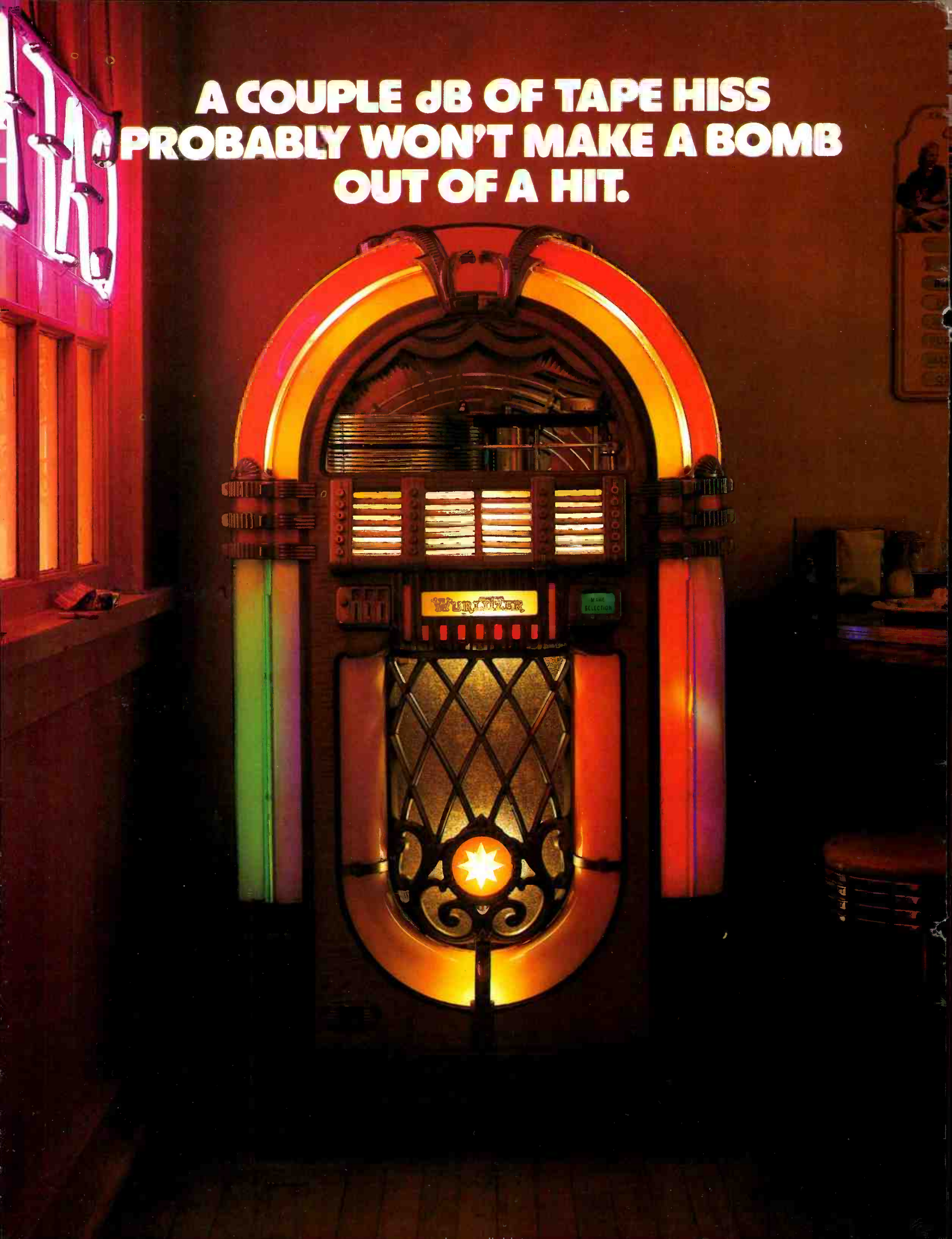
for additional information circle no. 3


A COUPLE dB OF TAPE HISS
PROBABLY WON'T MAKE A BOMB
OUT OF A HIT.
WURLITZER

BUT SOONER OR LATER YOU'RE GONNA HAVE TO CLEAN UP YOUR ACT.

WHY LIMIT YOURSELF WITH MASTERING TAPE?

You won't catch a professional racedriver putting cheap gas into his Lotus. It's just dumb.

And the same holds true in the studio. With all that heavy machinery and expensive talent, it makes no sense to compromise on your mastering tape.

TODAY'S SCOTCH 250 IS THE STATE OF THE ART.

With 250 you get far less tape noise. Considerably more high end clarity. Our exclusive oxide formulation and application reduces Mod noise. And when you add all that to dbx or Dolby it's positively the cleanest sound around.

SINCE WORDS ARE CHEAP, WE PUT 250 TO A ROUGH TEST.

Whatever the numbers or the meters say, it's your ears you should listen to. So we went to a very fussy, very fine engineer and asked him to devise a test to demonstrate the difference between 250 and our nearest competitor.

The guy first thought we were nuts.

"You're serious?" he asked. "I use 250. What if my test proves the other tape is cleaner?"

We gulped a little, and told him to go ahead. This test was bound to be expensive. But it would also be worthless, if everything wasn't above-board. That's why we chose Tom Jung of Sound 80, Minneapolis, to put it together. You may have heard of him.

THE TEST PROGRAM WAS RECORDED—ON TWIN MACHINES.

Jung, as we expected, left nothing to chance. On April 18, 1977 he recorded an original music program simultaneously on two 24-track MCI's fed by one console. One recorder was carefully optimized for 250. The other, just as carefully, for the competitor's tape.

Jung used NAB equalization at 15 ips. He really packed both tapes at 6db (370 nWb/m) over standard operating level—without a shred of noise reduction.

THE TRUTH CAME OUT FIRST AT THE AES SHOW.

It was May 10, 1977 at the LA Hilton. For playback we set up identical machines (our own M79 24-tracks, this time) with Altec 19 speakers. Then we opened our doors.

For each group of engineers we played not only the full mix, but individual tracks, first on one machine, then the other.

THERE WERE SOME WHO COULD NOT BELIEVE THEIR EARS.

"Play that bass track again," they'd say. And we'd play it.

"Are you sure both tapes were recorded at the same level?" We assured them they were.

"Lemme hear the strings with the horns."

In three days close to 600 people heard our 20-minute demo.

AND THE TRUTH IS...

We didn't find one engineer who didn't hear the difference in L.A. Ditto in Nashville, where the demo was repeated July 13 and 14.

You can simply pack more sound on Scotch 250 and still stay clean.

So the bottom line is this. Scotch 250 is cleaner tape.

DON'T TAKE OUR WORD FOR IT. BRING YOUR EARS TO NEW YORK.

We'll repeat this "head-to-head confrontation of mastering tapes" at the AES Show on November 4 and 5. Hear for yourself that heavy sounds don't have to be muddy.

3M COMPANY

CLEAN UP YOUR ACT WITH "SCOTCH" 250.

for additional information circle no. 4

Letters & Late News

from: **David Robinson**
V.P. Engineering
Dolby Laboratories
San Francisco, California

While I enjoyed reading William H. Hall's article on noise reduction in the October 1977 issue of *R-e/p*, and agree wholeheartedly with his general conclusion, I would like to point out that his remarks do not apply to the Dolby noise reduction system — lest anyone should come to the wrong conclusions.

By virtue of its design, our units have an overshoot of about 1 dB at 0 VU, and therefore do not exhibit the transient dulling effect Mr. Hall notices. Similarly, it is not necessary to make any changes in recording level when using the system, so retaining the full advantage of the noise reduction effect. The suppression of transients is one of the unique patented features of the Dolby system and thus, I am happy to say, no other system can boast the faithful transmission of transients through a practical recording system.

---

reply from:
William H. Hall

Compromise is the art form in engineering. Every systems engineer attempts to make the best compromise between design, operational parameters and cost for every product produced. Therefore, each product is not everything to everybody as witnessed by the multitude of products in the marketplace, with many of them making claims of great advantages over their respective counterparts. Each design usually has advantages and disadvantages, the advantages usually are inputs to the advertising copy writer and the disadvantages are quietly let alone to be discovered by the user or diligent evaluator.

As stated in the letter by Mr. Robinson, it is undoubtedly true that the Dolby system does not suffer from the overshoot problem. But the system cannot achieve as large signal-to-noise improvement as the dbx system, so it can be six of one and a half dozen of the other.

What this writer is trying to say is that in this case, both systems of noise reduction can do an excellent job, but in different ways and in different circumstances. Here is where the skill of the recording engineer comes into play, that is to know the difference between the two systems, and how to utilize these differences for a superior final product.

---

from: **F. Alton Everest**
Registered Consulting Engineer
Whittier, CA

Reference is made to the letter from Paul E. Rolfes, Chief Engineer of Soundcraftsmen, in June *R-e/p*.

Mr. Rolfes is certainly to be commended for coming up with something new in high power audio amplifiers. But, perhaps, the basic idea is not so new. Let's go back 40 years or so.

In 1936 my thesis for the engineer degree at Stanford University, "A High Efficiency Grid-Modulated Amplifier Using a Saturable-Core Reactor", described a system quite similar to Paul's "Class H Amplifier". In my system the plate voltage varied with the envelope of the audio modulating signal through the action of a saturable reactor. Actually, the grid bias of the amplifier being modulated was also varied so that the ratio of plate voltage to controlled part of the grid bias remained constant and equal to the of the tube. Thus low audio modulating signals were cared for by the quiescent plate and bias condition and audio bursts increased plate and bias voltage sufficient to care for the bursts.

This project was suggested to me by my major professor, Dr. Frederick E. Terman. News of this work was somehow picked up by the late J. N. A. Hawkins, an editor of the now defunct RADIO magazine and later noted for the audio innovations in Fantasia at Disney Studios. Johnny Hawkins was simultaneously working on essentially the same idea! Instead of applying for a patent, he suggested that we make the idea public through publication, which both he and we did (Ref. 1). I worked on the idea later with a graduate student at Oregon State where I was teaching. (Ref. 2).

It is amazing how old ideas reappear in contemporary garb. Although the original application was modulating an RF amplifier, the basic idea was equally applicable to audio amplifiers except there was little concern in those days

for efficiency in audio amplifiers and, of course, 250 watts RMS per channel in audio was unheard of for the "hi-fi" buff of that day. Modern solid state circuits are light years ahead of saturable reactors, but one uses what is available!

This is certainly not intended to denigrate Paul's fine contribution to the science and art of modern amplifiers, but only to set it in historical perspective.

Ref. 1: Terman, F. E. and F. A. Everest, "Dynamic Shift, Grid Bias Modulation", RADIO NO. 211, July 1936, pp 22-29, 80-82.

Ref. 2: Everest, F. A. and F. H. Dickson, "Applying The Dynamic Shift Principle", COMMUNICATIONS, July 1941.

Ref. 3: Hawkins, J. N. A., "A New High-Efficiency Linear Amplifier", RADIO, May 1936, pp 8-14, 74-76.

Ref. 4: Hawkins, J. N. A., "Expanding Linear Amplifier Notes", RADIO, June 1936, pp 63-67, 83-84.

Editor's Note: Mr. Everest is the author of the very popular HANDBOOK OF MULTI-TRACK RECORDING, of which more than 700 copies have been ordered by R-e/p readers.

reply from:

Paul E. Rolfes
Chief Engineer
Soundcraftsmen
Santa Ana, CA

Thank you for sending me a copy of Mr. Everest's letter in which he refers to work that he did using saturable reactors in an RF amplifier. I wish to congratulate Mr. Everest and Dr. Terman for their innovative thinking.

As we all know, the advancement of technology in the electronics industry is so rapid that, since we are all bound by the same physical principles, one repeatedly sees similarities in ideas, even though they are indeed very different.

Again, congratulations to Mr. Everest for his progressive thinking. We need more people like him in this world of electronics.

from:

Gordon Menard
Professional Audio Sales
Mincom Division
3M Company
St. Paul, MN

Enjoyed very much your extensive (to say the least) article covering tape deck maintenance. Your experience will, I am sure, help other people in performing these transport checks and repairs.

Concerning runout of the 3M capstan, we typically see about 200 microinches and find problems can occur when it's over 300 microinches.

Your statement about having good quality measurement tools is very important, and would without question make these tests useless.

from:

Steven B. Fuller
Department of Electrical Engineering
Washington University
St. Louis, MO

In reading Leon E. Weed's article in the October issue of *R-e/p* I was struck by several errors and other items requiring clarification. Taking them in the order they appear I would first question the use of the term "Time Domain". This term applies to a method of analysis of pulse circuits as opposed to "Frequency Domain" which is more often used for C. W. circuit analysis.

Secondly, I would point out that the mathematical analysis is not nearly so complex as we are led to believe. In fact, one of my freshman students has worked these out in three dimensions in

addition to the two dimensional analysis I presented in the April 1976 issue of *R-e/p*.

Thirdly, the M-S and X-Y systems are not equivalent as stated by Mr. Weed. The synthesized left and right microphone patterns are not cardioid as shown in the illustration but rather hypercardioid thereby differentiating this system from the X-Y system.

Fourth, I would like to see Mr. Weed justify his statements on cable length as it would take many hundreds of feet difference in cable length to cause a phase difference in the audio region. In addition, if these cable induced phase differences were to become significant relative to errors induced by the physical spacing between the capsules, the difference in cable lengths would be even greater.

Lastly, it should be noted that the microphones used should have the same polar patterns at all frequencies. Mr. Weed's choice of the U-87 is poor as its cardioid pattern varies greatly with frequency. A much better choice would have been KM-86's.

Those being my major points I would close by asking why one would synthesize a pair of figure eight microphones with a similar pair as shown in the illustration?

reply from:

Leon E. Weed
Division of Science
& Mathematics
State University at Albany
Albany, NY

I would like very much to hear from Mr. Fuller again and others after they have tried out this recording technique. I would suggest, however, that it should be plugged into an audio system rather than a computer.

Letter to:

Ampex Corporation
Audio-Video Systems Div.
Redwood City, CA

I have seen your latest advertising campaign in Billboard and *R-e/p* magazines extolling the virtues of your new and exciting ATR-100 Tape Recorder.

I am quite certain that you are unaware of the fact that this ad has caused quite a bit of controversy in our industry and I thought I would write to see whether it would not be to your own best interests to put this to rest. I will not quarrel with the possibility that the ATR-100 may be the best audio TAPE recorder in the world, but I must take some issue with your leaving out of the word TAPE.

I am sure you are well aware of the direct-to-disk industry which has sprung up in the past couple of years, and as the importers of the Neumann Disk Cutting Equipment with which a predominance of these disks are cut, I must tell you that I am quite certain that any A/B test of your ATR-100 against our Disk Cutting System would clearly establish that the direct disk is far superior to any tape machine, including the Telefunken recorders we are presently introducing here.

I would like to prevail upon you to reexamine your ad in the light of these facts, trusting that you had really not even thought of recording machines other than on tape when you made up that ad.

Stephen F. Temmer
President
Gotham Audio Corporation

DIGITAL AUDIO STANDARDS COMMITTEE ESTABLISHED

The Digital Audio Standards Committee, a joint effort of the Audio Engineering Society and the Joint Committee on Inter Society Coordination (JCIC) of the Electronic Industries Association (EIA), the Institute of Electrical and Electronic Engineers (IEEE), the National Cable Television Association (NCTA), and the Society of Motion Picture and Television Engineers (SMPTE).

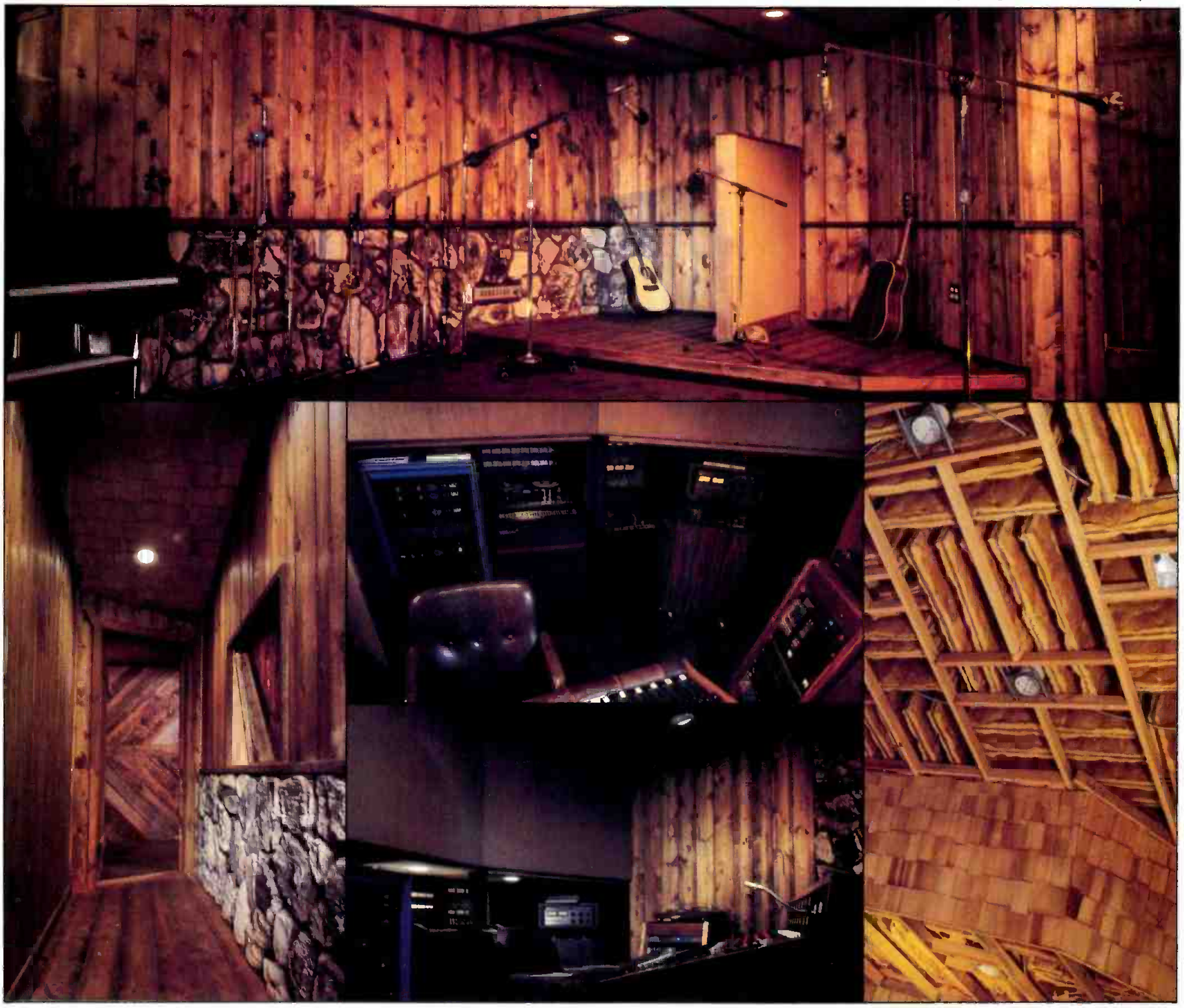
The initial meeting, held in Salt Lake City, Utah on December 1 and 2 was chaired by John G. McKnight of Magnatic Reference Laboratory.

The purpose of the meeting was to establish communications between various manufacturers and users of digital audio systems for all applications, and to discuss the standardization possibilities and problems.

The discussions centered on the sampling frequencies now used by the present digital audio studio systems, and those used by Japanese consumer systems using videotape and laser disc storage media; and the needs for interchangeability with television and motion picture systems. Frequencies discussed included 44.05594 kHz., 50 kHz., and 54 kHz. Because none of these frequencies meets all of the requirements for all applications,

. . . continued on page 88 —

Another full-service custom studio by Express Sound Co., Inc.



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THE EXPRESS SOUND CO.

for additional information circle no. 10

How to get a three-motor, direct-drive, isolated-loop deck. And save $5,500.

"Ingenuity of design can be fascinating for its own sake, but when it results in a product of demonstrable excellence, as with this tape recorder, one can only applaud..."

The review is from *Modern Recording*. The tape deck is Technics RS-1500US. And the ingenuity of design that *Modern Recording* and *Audio* have praised in recent issues is Technics' advanced "Isolated Loop" tape transport with a quartz-locked, phase-control, direct-drive capstan.

By isolating the tape from external influences, Technics has minimized tape tension to an unprecedented 80gms. Eliminating virtually all signal dropout. While reducing modulation and wow and flutter to a point where conventional laboratory measurement is seriously challenged. A considerable achievement when you realize Technics RS-1500US is priced substantially below its professional counterpart. $5,500 below.

Electronically, too, Technics has provided the ultimate in professional control and performance. A separate microphone amplifier. Record amplifier. Mixing amplifier. And three-way bias/equalization. While IC full-logic function controls permit absolute freedom in switching modes.

Compare specifications and prices. Then you'll realize there's no comparison. TRACK SYSTEM: 2-track, 2-channel recording, playback and erase. 4-track, 2-channel playback. FREQ. RESPONSE: 30-30,000Hz, ± 3dB (−10dB rec. level) at 15ips. WOW & FLUTTER: 0.018% WRMS at 15ips. S/N RATIO: 60dB (NAB weighted) at 15ips. SEPARATION: Greater than 50dB. RISE TIME: 0.7 secs. SPEED DEVIATION: ± 0.1% with 1.0 or 1.5mil tape at 15ips. SPEED FLUCTUATION: 0.05% with 1.0 or 1.5mil tape at 15ips. PITCH CONTROL: ± 6%. SUGGESTED RETAIL PRICE: $1,500.*

Technics RS-1500US. A rare combination of audio technology. A new standard of audio excellence.

*Technics recommended price, but actual retail price will be set by dealers.

Technics *Professional Series*
by Panasonic

For additional information circle no. 11

fact or fiction?

CONSOLE NOISE SPECIFICATIONS —

by Paul C. Buff

Having recently completed the design and specifications for an ultra-low noise amplifying device (see companion article ***"Towards Microphone Transparancy"***, Page 60), I found it necessary to measure noise levels within a fraction of a dB of absolute theoretical minimum. When I consistently came up with readings well beyond those limits, I knew something was wrong. Although I would have loved to advertise an amplifier with *negative noise,* I chose to take a safer route — correct the test proceedure to reflect the true characteristics of the device. To make a long story short, I found the measurement discrepancies, one by one, and can now measure and specify noise readings accurate to plus or minus ¼dB.

At this point you're probably saying, "So you didn't know how to measure good, and now you do. Big deal."

Well, let me go on a little bit further. You see, the impetus for my research into ultra-low noise electronics was from one of my pet customers. He had just traded in his 8 year old Quad Eight console on a new shiny "state-of-the-art" board, whose manufacturer I won't go into. When my customer started using his new toy, guess what he heard when he turned up his mikes? Right! Noise!

Referring to his handy dandy spec sheet, he found a mike pre-noise spec of -128 dBm E.I.N., a similar figure to that of his old console. Since his ears refused to believe the consoles were the same, he called me in to verify the amount of noise present in the new console.

Upon taking on the assignment, I first pondered just how both console makers managed to specify *noise performance well beyond that which is possible on this earth!* Ignoring this paradox, for the moment, I proceeded to make measurements of the second console, while simultaneously educating myself in the art of measuring *properly.* What did I come up with? Hang onto your hats — -118.5 dBm E.I.N. Nearly 10 dB below the manufacturers specification.

Yet I have in my files a letter from the chief engineer of the console company (one of the largest in the world) stating his measurements indicate an E.I.N. of -130.7 dBm, *a figure 6 dB better than the theoretical limit of -124.8 dBm.*

The reasons for the discrepancies in both the spec and the measured performance are twofold:

1. The manufacturer is using a misleading method of specification which does not relate to audible noise.
2. The manufacturer is uneducated in the proper measurement techniques required to express noise in either the misleading form or in the industry accepted correct form.

What is even more startling is the fact that most U.S. console makers are also guilty of one or both of the above allegations.

There are exceptions, of course, one notably being Mr. Deane Jensen, who manufactures transformers for a good many of today's consoles. I have read his specs and tested his transformers in detail and find his literature impeccably defines the performance of his product.

Getting back to the problem, what are the methods of specification, how are they applied (and mis-applied), and what do they mean in terms of audible signal-to-noise ratios?

THEORETICAL MINIMUM NOISE

The theoretical minimum noise level which can be obtained by any amplifier is determined by the thermal noise of its source resistance. Thermal noise, by definition, is the noise produced by the thermal agitation of electrons within any electrical conductor. Thermal noise is precisely predictable by use of the formula:

$$\text{Thermal Noise Voltage } E_t = \sqrt{4KTRB}$$

or

$$\text{Thermal Noise Power } P_t = 4KTB$$

(in watts)

Wherein:

$K = 1.38 \times 10^{-23}$ (Boltzmans constant)
T = Temperature in Kelvin
R = Source resistance in ohms
B = Absolute bandwidth in Hz.

CONSOLE NOISE SPECIFICATIONS —

Two facts are evident in the formula; that thermal noise is a constant quantity of power, regardless of the source resistance, and the noise is white in nature (equal noise power per cycle of bandwidth).

In a microphone amplifier, the source resistance is, indeed, the resistance of the microphone element itself. Thus, the mike element is the limiting factor in determining the Theoretical Minimum Noise performance of the system.

Applying the formula to *any microphone* of *any impedance*, over a 20 Hz. to 20 kHz. bandwidth, at room temperature (300°K), the Thermal Noise generated is precisely -124.8 dBm, and it is this figure which governs the minimum amount of noise which may be present.

To those of you who have been induced to believe that your mike pre-amps are producing noise levels equivalent to -129 dBm at their inputs, I'm sorry, you lose. Anything better than -124.8 dBm is fantasy — *absolutely impossible!* (-121 dBme is typical in today's equipment.)

I realize that this revelation may seem a bit hard to swallow in view of the mounds of "technical" publications which refer to E.I.N.'s of -127 to -130 dBm. All I can offer is that you get yourself a good electronics text book and an accurate meter and find out. In order to make the rest of this text crystal clear, I think it wise to define the meaning of the terms used here-in.

DEFINITIONS

dBm — A measurement of *power* referred to 1milliwatt @ 600 ohms. 0 dBm is characterized by a voltage reading of .7745 vrms. across *600 ohms*. Thus, a dB reading meter whose 0 dB point represents .7745 vrms. may be directly employed to read dBm if, *and only if*, the impedance of the circuit point under measurement is 600 ohms.

dBme — A measure of *power* referred to 1milliwatt @ any other impedance. Example: .7745 vrms. across 150 ohms = 4 mw = +6 dBme.

dBv — A measure of *voltage* generally accepted in the U.S. to be referred to .7745 vrms. A very common mistake is to think of dBv as being the same as dBm. While this is true at 600 ohms, there is no direct relationship at other impedances, since dBv is voltage, and dBm is power. (See nomograph Figure 1.) Example: 0 dBv (.7745 vrms.) across 10,000 ohms = .06 mw. = -12.2 dBme.

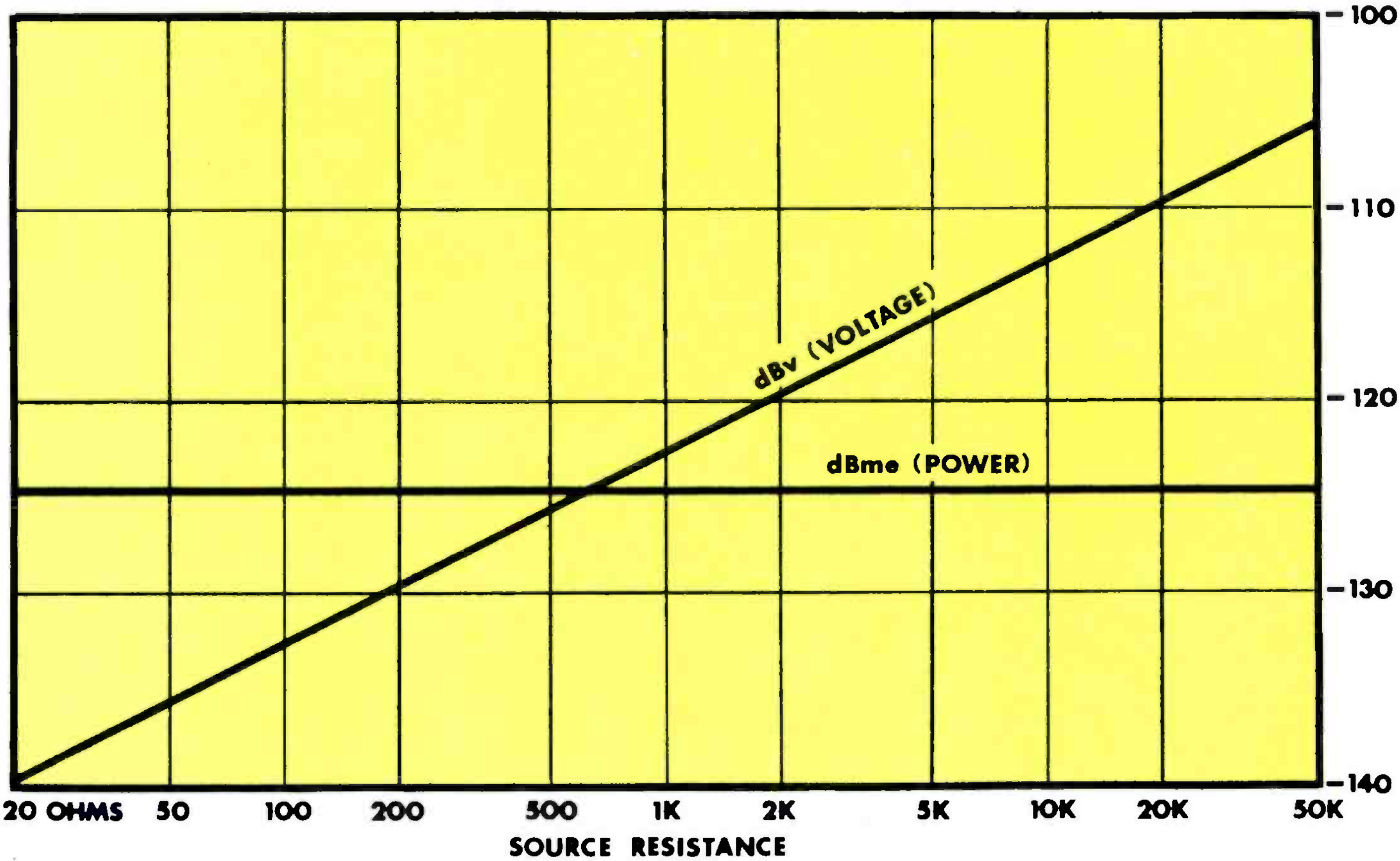
dBV — (Capital "V") Same as dBv, except referred to 1 vrms. (European)

EQUIVALENT INPUT NOISE (E.I.N.)

Since any amplifier may perform a useful function only when connected to some signal source, it is necessary to include the Theoretical Minimum Noise thermally generated in that source if the specification is to bear any meaningful relationship to the circuit's noise performance in actual use. This philosophy is applied almost universally throughout the electronics industry. Thus, the E.I.N. of any amplifier can never be better than the thermal noise of the source. As stated earlier, this amounts to -124.8 dBm,

. . . continued on page 24 —

Figure 1:
Theoretical Minimum Noise Voltage (dBv) and Power (dBme) v.s. Source Resistance
(Absolute Bandwidth = 20 Hz. to 20 kHz.)



Sierra Audio

Sometime before the sun comes up, you'll learn to love your MM-1200.

Ampex gives you a lot of reasons to buy an MM-1200 multichannel audio recorder: capacity, fidelity, flexibility and reliability. The performance specifications alone have made this the best seller among professional multitrack recorders.

But you'll find your own reasons for loving the MM-1200 you buy.

Some morning, after a long session that dragged through the wee hours, you're going to realize just how many MM-1200 features you used to keep things moving. How you used the electronic tape timer, plus search-to-cue, to save precious minutes of time. How you used the quick-change head feature to switch back and forth between 8, 16 and 24 track work. The ease with which you employed the optional accessories such as the pure video layback head, synchronizer and variable speed oscillator to finish the audio portion of a video production.

The same professional drive for perfection that kept *you* going, kept *our* engineers working to refine an already-respected design.

That's why you see so many MM-1200s in professional recording studios. Sooner or later, before the sun comes up, everybody learns to love an MM-1200.

AMPEX

Ampex Corporation, Audio-Video Systems Division, 401 Broadway, Redwood City, California 94063, 415/367-2011.

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—continued from page 20 . . .

20 - 20 kHz.

If it is desired to express this figure in terms of voltage (dBv), it is necessary to reference the voltage to the source impedance so that a conversion to power may be made. For instance, Theoretical Minimum at 50 ohms may be expressed as -135.6 *dBv* r.e. 50 ohms, but *not* as *-135.6 dBm*, or *-135.6 dBme*. (See Figure 1.)

A person relatively unaware of the finer points of mathematical deciphering, as typified by the average console buyer, might tend to look at the figures (-124.8 dBme r.e. 150 ohms) and (-135.6 dBv r.e. 50 ohms) and make the deduction that the latter represented less noise. In reality, -135.6 dBv r.e. 50 ohms represents exactly the same amount of noise (3.31×10^{-13}mw) as -124.8 dBme re: 150 ohms, and will yield exactly the same noise performance.

EQUIVALENT INPUT NOISE
(Misleading Forms)

In preparing this text, I personally called upon a good cross section of U.S. console makers, in an effort to interpret their methods of arriving at mike pre-amp specs. One fact was clear from this study: *There does not exist a standard method of specification of E.I.N. in the U.S. console industry. In most cases, due to improper, misleading or downright ignorant specmanship, U.S. consoles cannot be compared, for noise levels, on the basis of comparing specifications.*

Of the methods in common use today, I have assembled four categories, three of which are out and out misrepresentations, and only one being properly stated. Here are the categories:

A. — Transducer Gain Theory — A method which utilizes an old formula of undetermined origin which allows one to correctly determine the E.I.N. of an amplifier, in dBme, then *arbitrarily add 6 dB* to the result, thus making Theoretical Minimum appear to be -130.8 dBm. I have yet to find an industry expert who is able to explain why the author of the formula chose to give the industry a 6 dB handicap over reality.

B. — Amplifier Only Noise — A method which excludes the thermal noise of the source from the calculation. I say calculation since an amplifier cannot be measured, at a given impedance level, without an equivalent source resistance. Even if it could be, such a specification bears no direct relationship to the actual signal-to-noise level present under use, and as such, is of little value to the user.

C. — The Erroneous Method — Here is a classic case of adding apples to get oranges. If I told you the precentage of "engineers" in our industry who commit this crime I believe you'd be amazed. Here's how this one goes: You measure the Equivalent Noise *Voltage* at the amplifier's input, in dBv, and then you simply refer to it as *power* and state it as dBm. The lower the specified input impedance, the better your numbers. In current advertising, one console builder claims his equipment is "3½ times quieter than the competition's published specs", and specifies 50 ohm source impedance. Now in some of the cited cases, the competition's specs already exceed theoretical limits by several decibels.

Indeed, the advertiser's console may have a lower *equivalent input noise voltage* due to the lower measurement impedance, but a 50 ohm microphone requires 4.8 dB more gain than a comparable 150 ohm microphone, for a given SPL. The end result is that while the input *noise voltage (dBv)* is lower, it must be amplified more. The *noise power (dBme)*, noise figure and effective signal-to-noise ratios are not improved. (See Figure 1.)

D. — A Correct Method — Some of the manufacturers I reviewed state the E.I.N. of their equipment in terms of voltage (dBv) with reference to the specified measurement impedance. This method is basically honest and acceptable. I did, however, find certain discrepancies among some users of this specification method, relating to inaccurate bandwidth measurement techniques, exclusion of RMS correlation factors, failure to include mike loading losses and failure to include the shunt effect of the amplifier input impedance across the microphone in stated impedance levels. These infractions are relatively minor in relation to methods A, B, and C, but can, and in some cases do, add up to mis-statements of up to 3 dB from actual noise levels.

NOISE FIGURE
(A Reliable Specification)

Let's face it, the prospective purchaser of a mike amplifier is not interested in applying thermal noise formulas, converting terms, referring to charts or going through a host of calculations in order to decipher a manufacturers specification. When it comes to noise, he (she) wants to know one thing: By how much does the amplifier degrade the signal-to-noise ratio available from the microphone itself?

This sort of specification is widely used throughout the rest of the electronics industry, and is called "NOISE FIGURE". Its use implies that the specifier has considered all factors commonly omitted (as in #D above). NOISE FIGURE is simple in concept and simple to apply.

If an amplifier adds no noise to the source, and does not reduce the potential of the source by loading effects, it has a NOISE FIGURE of 0 dB. If an amplifier degrades the potential signal-to-noise ratio of the source, either by amplifier generated noise, or through loading effects by, say 5 dB, then its NOISE FIGURE is 5 dB. The specification is directly comparable to the anticipated results, and is hard for a manufacturer to fudge on, unless he just lies.

FACTORS DETERMINING MEASUREMENT ACCURACY

THE RMS FACTOR — Noise measurements are almost universally made with average responding meters. Such a meter will read the TRUE RMS value of a sine wave, and only a sine wave. An average responding meter will normally indicate a 1.05 dB error when applied to white noise. This amount must be added to the actual measured noise, unless a TRUE RMS meter is used.

NOISE BANDWIDTH — The thermal noise formula shows that noise is proportional to the absoulte bandwidth under measurement. Most noise measurements are made with a first order (6 dB/octave) passive low pass filter. Textbooks state that such a filter with a 3 dB point of 12.7 kHz. will yield an effective absolute bandwidth of 20 kHz. While the book is correct, it pre-supposes that the device under measurement has an infinite bandwidth. Typical audio electronics, of course, do not have an infinite bandwidth. The use of first order filters on a mike pre-amp exhibiting a roll off at, say 80 kHz. can introduce an error of around 1 dB in the measurement. This error must either be added to the measured result, or a higher order filter (over 36 dB/octave) used. If the device to be measured has frequency response variations within the measurement

bandwidth, they must be equalized out, either on paper or with hardware.

SOURCE LOADING LOSSES — If we are to accurately define the loss of signal-to-noise ratio from that which is potentially available from the source itself, we must consider the power that is lost in the input impedance of the amplifier. If, for instance, a circuit designed to accommodate a 150 ohm mike has an input impedance of 150 ohms, 1/2 of the potential mike signal will be lost in the amplifiers input resistance, thereby reducing the potential signal-to-noise ratio by 3 dB, before even entering the amplifier. It is obviously manditory that this loading loss be included in the specification.

PUTTING IT ALL TOGETHER

Now that we've assembled all the necessary methods needed to make a correct noise measurement, let's put down a procedure to do it properly.

1. Calibrate and verify all test equipment.
2. Examine the frequency bandwidth of the circuit to be measured.
   A. This must be done with an oscillator which exhibits the characteristic impedance specified for the test. (Figure 2.) Failure to do this can result in an untrue bandwidth measurement due to irregularities in the source impedance of the amplifier under test.
3. If the device to be measured exhibits frequency response variations within the range of 10 times the upper measurement bandwith:
   A. Use a high order (over 36 dB/octave) filter, or,
   B. Determine a correction factor to be added to the measured noise.
4. Accurately establish the effective gain of the circuit, allowing for source loading losses, using the following procedure:
   A. Use a test oscillator (1 kHz.) with an output impedance equal to the specified test impedance. (Figure 2.)
   B. Set the oscillator for a level within the linear operating range of the amplifier under test while *disconnected* from the circuit and feeding only a high impedance dBv reading voltmeter. (In this configuration the oscillator represents the full potential of a microphone of equal impedance.)
   C. Without re-adjusting the oscillator output level, connect it's output to the input of the circuit under test.
   D. Measure the voltage (dBv) at the amplifier output.
   E. Subtract the open circuit oscillator voltage (4B) from the amplifier output voltage (4D) to arrive at the effective gain of the amplifier.

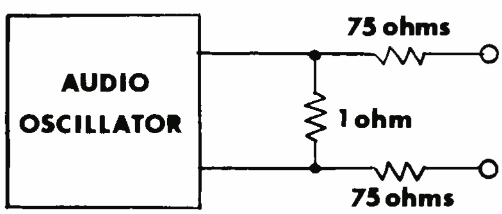


Figure 2:
Recommended Configuration of Audio Oscillator for Testing Microphone Inputs
(Circuit values shown for 150 Ohm impedance.)

5. Remove the test oscillator and terminate the input with a resistor equal to the specified test impedance.
6. Measure the noise at the amplifier output with a filter as described in #3 above. (It is wise at this point to monitor the noise under measurement both audibly and with a scope to insure that you are measuring noise and not hum or garbage.) If hum is a problem and you are trying to determine hiss level only, a 400 Hz. high pass filter may be used without serious error introduction.
7. Apply any correction factor indicated in #3.
8. Unless your meter reads TRUE RMS, add 1.05 dB to the reading. (I.E. –73 dBv + 1.05 dB = –71.95 dBv)
9. Subtract the effective amplifier gain (4E) to arrive at the E.I.N. of the amplifier in volts (dBv).
10. Convert this to dBme, based on the value of the specified test impedance, using the following formula:

$$\text{dBme} = \text{dBv} + 10\log_{10}\left(\frac{600}{R}\right) \text{ wherein:}$$

dBv = #9 above
R = Specified test impedance

The result of this calculation will yield the correct E.I.N. of the amplifier in dBme, including thermal source noise and loading loss.

11. Calculate the thermal noise of the source, at the specified bandwidth, using the following formula:

$$\text{Thermal Noise Power (dBme)} = 10\log_{10}(B \times 1.656 \times 10^{-17})$$

Wherein B = Specified test bandwidth

12. Subtract #11 from #10 to arrive at the NOISE FIGURE of the amplifier.

SUMMATION

As you might have guessed by now, making a correct noise measurement on an input amplifier is a bit more complex than hanging a meter on the output and writing down the answer.

It is of acute importance that the manufacturers of such devices fully understand the meaning of the specifications they publish, and educate themselves in the proper methods of measurement and calculation.

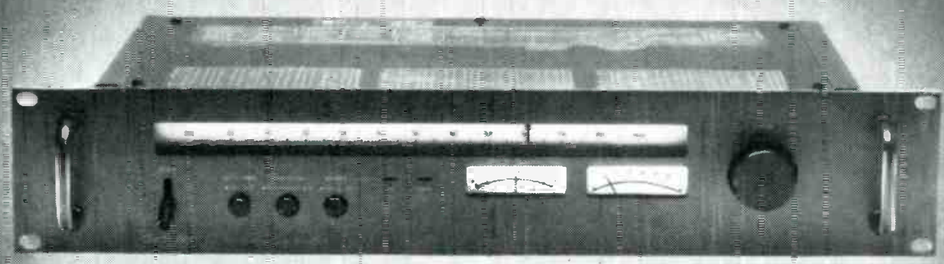
I strongly urge that you, the users of such devices demand the industry adoption of a standardized method of measurement and specification which includes attention to all of the details which I have mentioned in this text.

I should be more than happy to act as a self-appointed chairman for this cause if I receive sufficient feedback from the industry indicating your desire for such a standard.

Please feel free to direct any comments on this subject to me, c/o Recording Engineer/Producer, P. O. Box 2449, Hollywood, CA 90028.

In closing, let me say this: While I have heaved some rather heavy stones at the U.S. console industry, most representatives I spoke with showed a sincere desire to clean up their act, but stated that everyone else was guilty of mis-stated specmanship. The feeling was that if they made proper specifications, their equipment would look bad in comparison to the others. I suppose they have their point. It's up to you dear customer, what'll you have?

Introducing the Technics ST-9030 tuner. Purists would feel better if it cost over $1,000.



To some, tuners that offer 0.08% THD, 50 dB stereo separation, a capture ratio of 0.8 dB and waveform fidelity should demand a price tag of over $1,000. But with the ST-9030 this performance can be yours for under $400.*

That's quite a feat for a tuner. But then the ST-9030 is quite a tuner. It has two completely independent IF circuits: A narrow band, for ultra-sharp selectivity. And a wide band, for ultra-high separation and ultra-low distortion. It even selects the right band depending on reception conditions, automatically.

Both bands give you the same extended flat frequency response. Because, unlike conventional tuners, the ST-9030 utilizes an electronic pilot cancel circuit that cuts the pilot signal, without cutting any of the high end. It's ingenious. And a Technics innovation.

The Technics ST-9030 has one of the quietest, most sensitive front ends of any tuner. With an advanced linear frequency 8-ganged tuning capacitor and 3 double-tuned circuits, plus dual gate MOS FETs in the 2-stage RF amplifier and balanced mixer circuit. What's more, there's a servo tuning circuit that locks into the tuned frequency, regardless of minor fluctuations. The result: Negligible drift distortion and maximum stereo separation.

Technics ST-9030. Compare specifications. Compare prices. And you'll realize there's really no comparison.

THD (stereo): Wide—0.08% (1kHz). Narrow—0.3% (1kHz). S/N: 80 dB. FREQUENCY RESPONSE: 20Hz—18 kHz +0.1, −0.5 dB. SELECTIVITY: Wide—25 dB. Narrow—90 dB. CAPTURE RATIO: Wide—0.8 dB. Narrow—2.0 dB. IF, IMAGE and SPURIOUS RESPONSE REJECTIONS (98 mHz): 135 dB. AM SUPPRESSION (wide): 58 dB. STEREO SEPARATION (1 kHz): Wide—50 dB. Narrow—40 dB. CARRIER LEAK: Variable −65 dB (19 kHz). Fixed −70 dB (19 kHz, 38 kHz). SUGGESTED RETAIL PRICE: $399.95*

Technics ST-9030. A rare combination of audio technology. A new standard of audio excellence.

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Technics *Professional Series*
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For additional information circle No. 15

The IE-30A Audio Analysis System from IVIE

- 1/3-octave Spectrum Analyzer
- Full-octave Spectrum Analyzer
- Precision Sound Level Meter
- True RMS AC Voltmeter
- RT_{60} (Reverberation)*
- THD (Harmonic Distortion)*

For the first time, a real time analyzer and precision sound level meter have been combined into a portable audio analyzer "system" with features and accuracies rivaling the best laboratory instruments available.

The fully digital IE-30A comes standard with a precision laboratory microphone calibrated in dB-SPL and remoteable up to several hundred feet, a test probe with three precision attenuator settings for calibrated dBμV measurements (true rms, average or peak), nickel cadmium batteries with charger, and a hard shell, foam lined travel case.

*Using optional accessories

Other features include selectable detector responses, gated mode operation for measurement of reflections and time delay events, dual involatile memories that store or accumulate data that can be recalled to the IE-30A display up to weeks later.

The IE-30A was designed to accommodate an inexpensive new family of optional accessories. The IE-17A measures RT_{60} (reverberation time) in 1/3-octave bands up to 99.99 seconds with 10 millisecond resolution. The IE-15A measures total harmonic distortion (THD) to less than .01%.

SPEC BRIEFS

- 1/3 octave or full octave bands 30 filters on ISO centers 25Hz to 20KHz.
- Highly selective three pole-pair filters exceed ANSI S1.11-1966 Class III, B.S. 2475-1964, DIN 45652, and IEC255-1966.
- 1/3 octave display weighted A, C or Flat.
- LED array 30 x 16. Resolutions of 1, 2 or 3 dB for display ranges up to 45dB.
- Precision SLM has *Fast*, *Slow*, *Impulse* or *Peak* responses with A, C or Flat filter weightings.
- 4 digit/0.1dB resolution readout with display hold mode.
- Meets requirements of:
  ANSI S1.4-1971 TYPE S1A, S1C,BS 4197-1967
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- 30 to 149 dB SPL re $20\mu N/M^2$.
- Microphone is omnidirectional condenser Type 1 (Precision). Flat 10Hz to 20KHz.
- Signal outputs for recorders, oscilloscope displays and voice prints.

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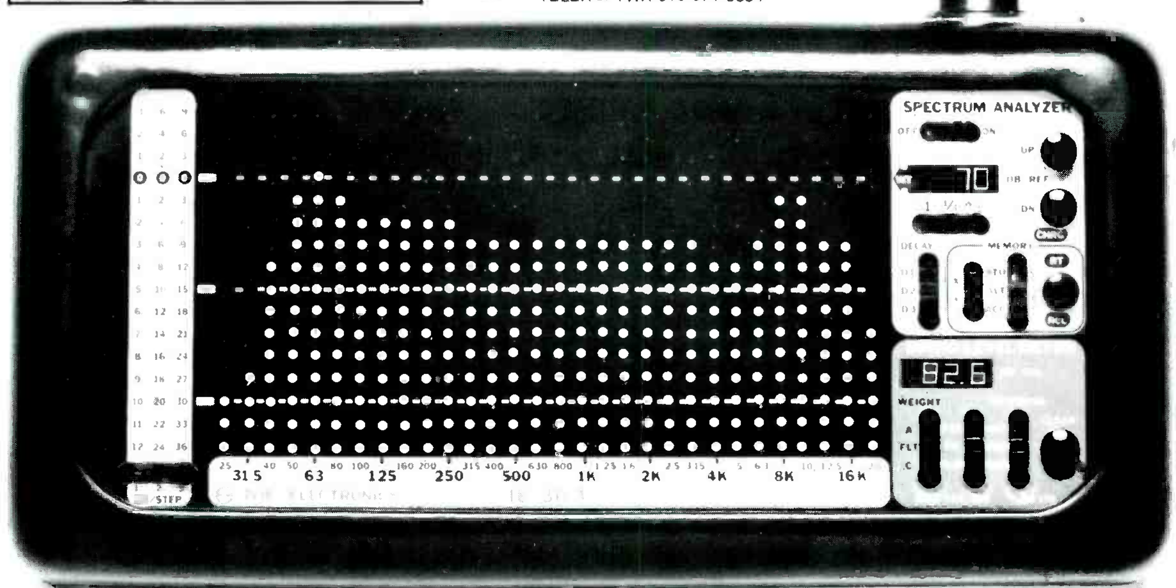

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for additional information circle no. 16

The CPR-16

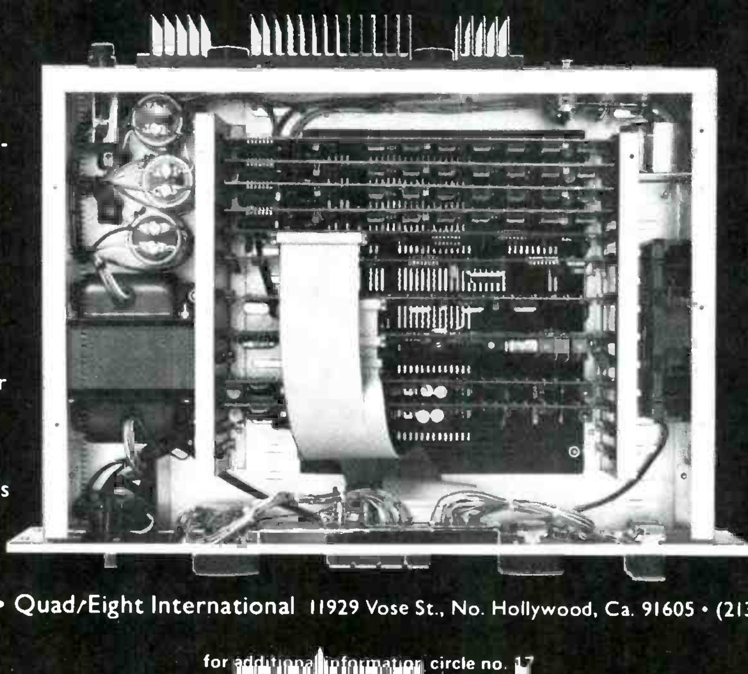
Computer Programmed Reverberation

The Quad Eight CPR-16 represents a revolutionary breakthrough in the application of advanced computer technology for the professional audio marketplace.

Two years in development, the CPR-16 is the first product to embody advanced digital technology in a configuration which will allow an unprecedented degree of control over the reverberant field by signal processing.

It offers the user a flexibility beyond the now ordinary mechanical methods; every possible aspect of the reverberant field is capable of alteration by the engineer. Reverberation time can be changed from zero to twenty seconds in sixteen steps, even during operation, without signal degradation. High and low frequency damping rates can be controlled over a wide range which previously was only achieved by time-consuming and clumsy re-arrangements of complex arrays of absorption splays in live chambers or rooms. The simulation of "room size" can be modified with a single control which adds a variable delay before the first echo or reflection signal.

And, most importantly, the CPR-16 allows the prominence and density of resonant modes to be altered. Thus, the density and diffusion rate of echoes can be tailored to match any room, electromechanical device, or whim.



The CPR contains two individual reverb programs and one "open" program provision for future custom sound processing effects. Program 1 is analogous to a live acoustical chamber, and Program 2 creates a simulation of an artificial plate. Program 3 switching facility is pre-wired for a plug-in programming module and Program 4, ECHO allows a "Tape-Echo" simulation that is unobtainable by present mechanical methods.

The CPR-16 incorporates control over every critical aspect of the reverberant field. If you have a special application which requires a previously unavailable sound processing effect, information on custom programs is available from the factory.

Write us or call for a detailed brochure on the revolutionary CPR-16. The age of affordable digital processing for audio has arrived. If you've been waiting for technology to catch up with your imagination, your time has come.

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for additional information circle no. 17

A 20 kHz Low Pass Filter for Audio Noise

In making routine audio system noise measurements, one is often presented with the problem of obtaining a realistic figure describing the white noise voltage appearing at the output terminals of a wide band audio system.

Simply connecting an AC voltmeter across the system output terminals is an unsatisfactory approach because the voltmeter, being a very wideband device, measures and reads the value of all the noise occurring from frequencies of a few Hertz to several megaHertz. The error incurred in including noise energy above the range of audibility results in an overly pessimistic evaluation of the noise performance of the system. Many times recognition of this fact has prompted the use of hastily fabricated R-C filter networks having response knees located somewhere in the ten to thirty kiloHertz region. The filter is inserted in the measurement circuit immediately prior to the voltmeter used to indicate the noise voltage. The results obtained in this way may be at considerable variance from subjective impressions of the system noise performance. Since the nature of the filter used has a very significant effect on the observed value of the noise voltage, and the characteristics of the filter are dependent on the components chosen, a great deal remains to be desired as far as accuracy of the measurement and comparability of results is concerned.

A popular approach uses a single pole filter (6 dB per octave) constructed with a series resistor and a shunt capacitor with the –3 dB point F_c at 12.7 kHz. This frequency is derived by dividing the 20 kHz. bandwidth by —/2. This will yield the same reading of white noise as a sharp cut-off filter at 20 kHz. if the system under test is flat to well beyond 100 kHz. But since the system is not usually that flat, errors of 1 dB or more usually result.

Measurements

by Deane Jensen

The filter circuit described here is offered as a solution to the noise measurement filter selection dilemma. It consists of four cascaded stages whose net response can be described as an eight pole Butterworth response with a 3 dB break at 20 kHz. The schematic for the filter circuit is given in Figure 1.

The Butterworth filter itself is comprised of stages A3 through A6 and their associated components. The reader is urged to select and/or trim component values to within 1% of those given for the filter stages themselves as the accuracy of the response characteristic of each individual filter stage will affect the overall response of the system and thus the accuracy of the measured results obtained using the filter.

A number of convenience features have been included in the circuit that may bear some explanation. Input and output ports of the filter are accessed through either GR binding post-type terminals or standard BNC co-axial connectors. The low-pass filter characteristic may be switched in or out for comparison of band-limited readings versus wide-band noise indications.

Stages A1 and A2 are provided for additional gain should it be necessary to boost the noise signal to obtain a readable deflection on the indicating voltmeter. Stage A1 can be selected for a gain of either 20 dB or 0 dB. Stage A2 has a fixed gain of 20 dB. Its output may be selected for input to the filter circuit in addition to whatever gain is selected for the input stage A1. We have, then, the option of adding 20 or 40 dB of voltage gain to the incoming noise signal.

Finally, an 800 Hz., single-pole, high-pass R-C filter is provided to minimize the effects of 60 Hz. hum that may be

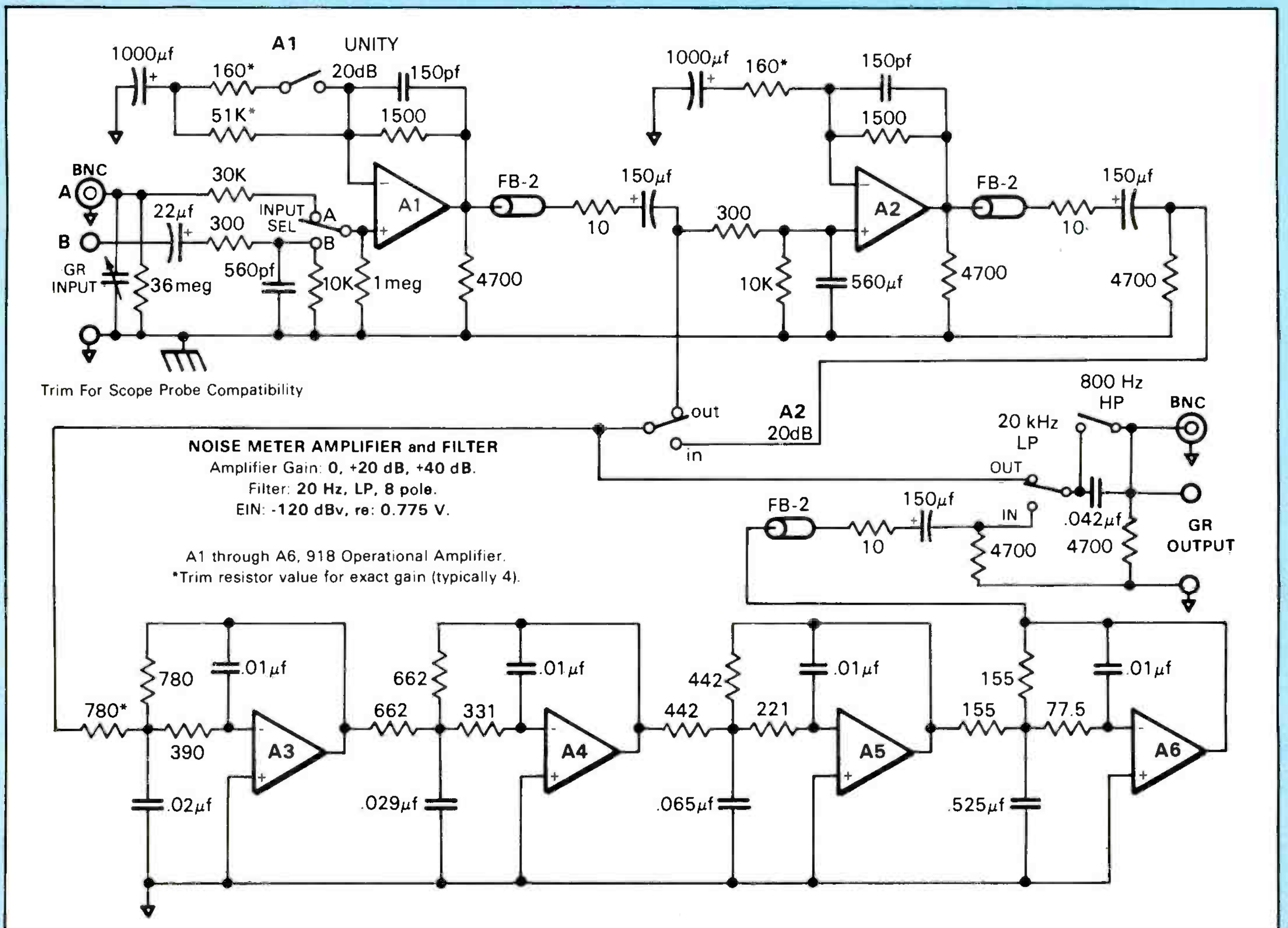


Figure 1: **Schematic diagram of the noise filter/noise amplifier. The gain elements are Pacific Recorders & Engineering 918 amplifiers. Power supply connections have been deleted for clarity.**

present in the audio system output signal. The 800 Hz. filter attenuation at 60 Hz. is about 22.6 dB. Use of the 800 Hz. high-pass should not affect the observed indication of a white noise spectrum more than about 0.1 dB of its reading without the 800 Hz. filter. If the noise signal being observed has a significant 1/f character, the effect will be greater. The user should be aware that use of any but a high input impedance voltmeter will be likely to adversely load the filter output circuit.

A few construction tips are offered as an aid to the reader in obtaining good results from this filter construction project.

The first of these is to verify the frequency response of each filter stage individually using the response curves given in Figure 2. As mentioned previously, it is very important that the responses of each stage very accurately conform to the theoretical curves if the overall response of the filter is to agree with the composite curve. The composite curve was chosen so that

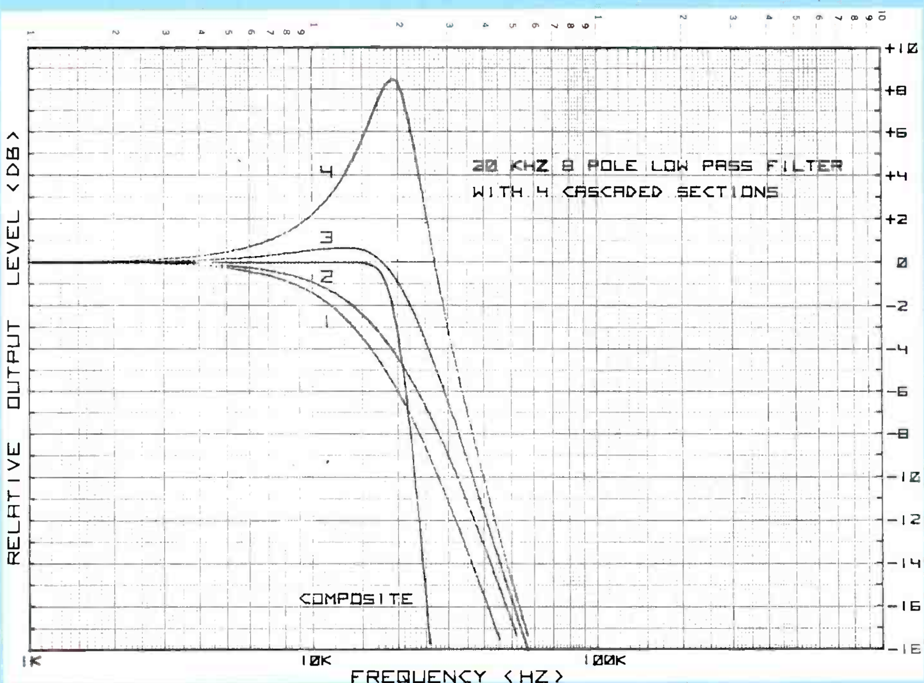


Figure 2: **Individual frequency response curves for filter stages A3 through A6. Curve 1 corresponds to stage A3, curve 4 to stage A6. See text for details.**

the high frequency roll-off characteristic of the system under test, if it lies beyond about 30 kHz. as will most generally be the case, will not adversely affect the observed noise measurement as will be the case with

simple noise filters having single-pole characteristics.

The break point of the 800 Hz. hum filter should also be determined with care and verified by direct measurement. The effective cut-off of the 800 Hz. single pole corresponds to an infinite slope cut-off filter at 500 Hz. If this value of effective cut-off is not maintained, the noise measurement error may be greater than about 0.1 dB or the amount of hum rejection will be decreased.

The input BNC connector can be used to accept the input of a 10:1 oscilloscope probe if the reader desires. This is particularly handy for isolation of noise amplifier stages. It is probably most desirable to add a small trimming capacitor across the 36 megOhm resistor so that a selected scope probe will respond to the noise filter input in the same way as it does to its customary oscilloscope.

The GR binding post input is intended to accept signal from low impedance sources such as operational amplifiers. The noise performance of the filter itself will be impaired and the measuring circuit will be excessively loaded if the binding post input is used for observation of a high impedance source.

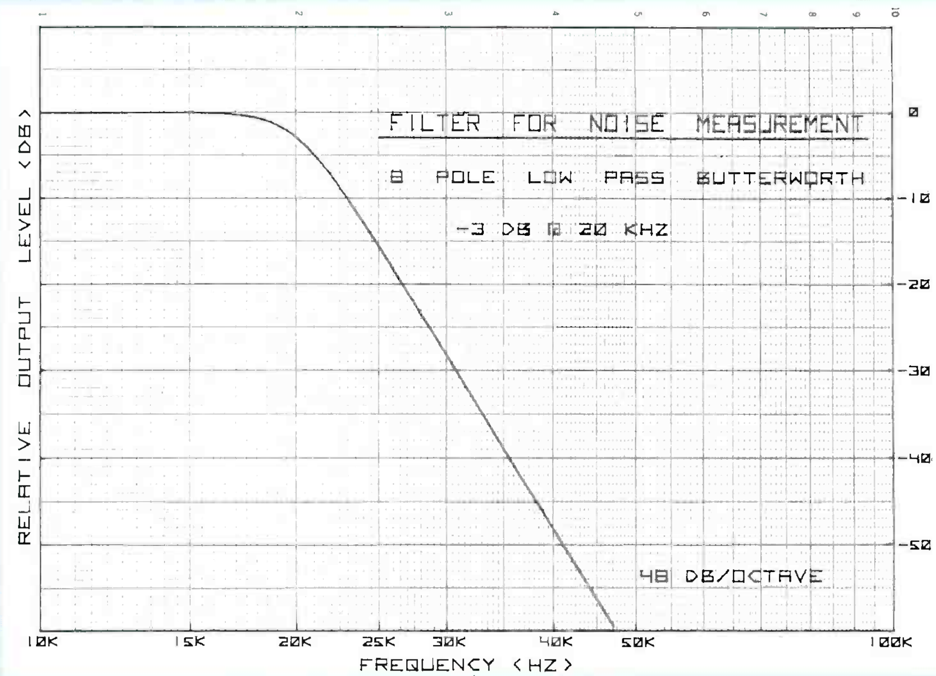


Figure 3: **Expanded plot of the 8-pole filter amplitude transfer characteristic showing response to -60 dB.**

The ferrite beads on the amplifier output leads are specials available from Jensen Transformers as ferrite bead number FB-2. Its purpose is to isolate the capacitive load from the amplifier output by raising the load impedance seen by the gain element at very high frequencies.

The 780 Ohm resistor at the input of filter stage A3 should be trimmed for an overall gain of unity for the filter stages

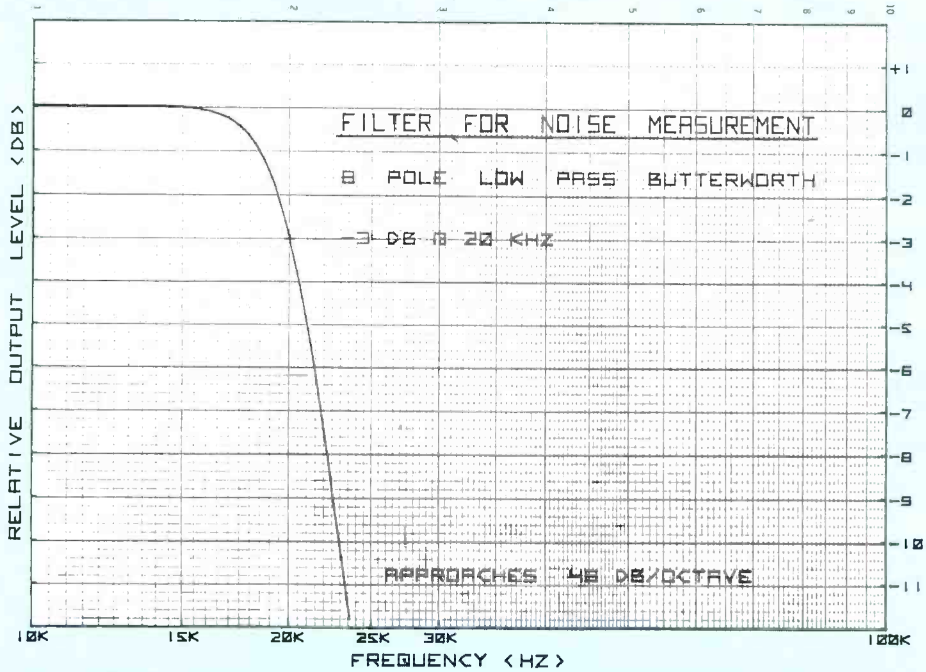


Figure 4: **Detailed plot of the 8-pole filter amplitude transfer characteristic showing details of response about the knee. The net response of any filter constructed should be compared to these theoretical curves for proper results using the filter.**

A3 through A6 taken as a whole, and not for unity gain of stage A3 only. This is important in the validity of the comparisons between readings taken in the filter in and filter out switch positions.

The gain stages themselves are Pacific Recorders & Engineering 918 discrete amplifiers. An alternative that may be acceptable from the standpoint of gain stability and noise performance is the Signetics 5534 IC. Signetics will soon be offering this device generally. The 5534A model device is specified as having a lower input noise current rating than the 5534. It measures noisier in this application because the input noise voltage of the -A chip is higher than the 5534. Because the application of the gain element here is noise voltage rather than noise current dominated, the un-suffixed chip is to be acceptable.

As far as the power requirements for the system are concerned, an external, low-noise, low ripple, regulated 15 Volt bi-polar supply will generally suffice. An optional alternative to an external power supply would be a bi-polar battery supply of ±15 to ±22.5 Volts. The Eveready 762S (NEDA 709) is a possible choice.

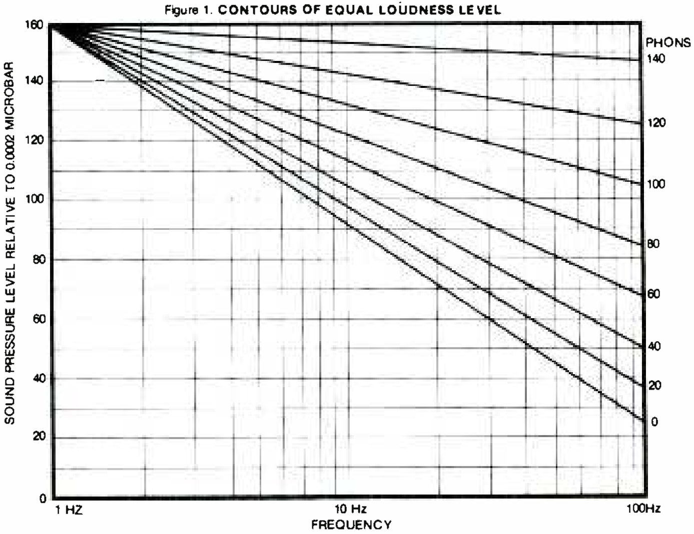
The use of this filter can contribute to a more repeatable measurement of electronic white noise in the audible spectrum. Comparison of noise measurements taken using it will lead to more valid judgements of system noise performance. Last, but not least, the use of this filter will permit measurements of noise that agree more closely with theoretical expectations than do the hasty R-C networks chosen for convenience or accepted for availability.

The Case of the Missing Low Notes

by Michael Rettinger

Audio men are generally well aware that the recording and reproduction of tones below 50 Hz. are faced with many difficulties. In disc recording they are confronted with the problem of groove width for lows with significant amplitudes, and in tape recording special reproduce amplifiers with a high gain at the low notes are required (this is for the reason that the standard ring-type reproduce head responds to the rate of change of the magnetic field, which rate becomes proportionately less with frequency.) There is no reason why engineering cannot find a solution to these **recording** problems.

Figure 1. CONTOURS OF EQUAL LOUDNESS LEVEL



However, it is in the **reproduction** of these low notes where sophisticated technology is of little help. The reason for this is that what is needed consists simply of an increasingly larger air volume displacement of the speaker cone or diaphragm as the frequency is lowered. Theoretically the moving part — usually likened to a piston — must displace a certain amount of air before a given sound pressure level, SPL, can be established for a given frequency at a given distance from the source. The lower the desired note, the larger the piston excursion must be. Its travel from its mid or rest position is, however, inversely proportional to the square of the frequency for the establishment of a wanted SPL. Thus, at 25 Hz., the cone must travel four times as far as at 50 Hz. to achieve an equal SPL.

But even that is not all. The human ear becomes increasingly less sensitive to the lower frequencies, so that for an equal loudness level in phons the cone must travel more than four times the distance it is required to move for the next higher octave tone; the exact additional travel being a function of the desired note and amplitude.

It is the purpose of the following to illustrate graphically the required diaphragm or piston excursions for frequencies below 50 Hz., including the infrasonics below 16 Hz. not normally audible (if perceptible sensorially in another way) to humans.

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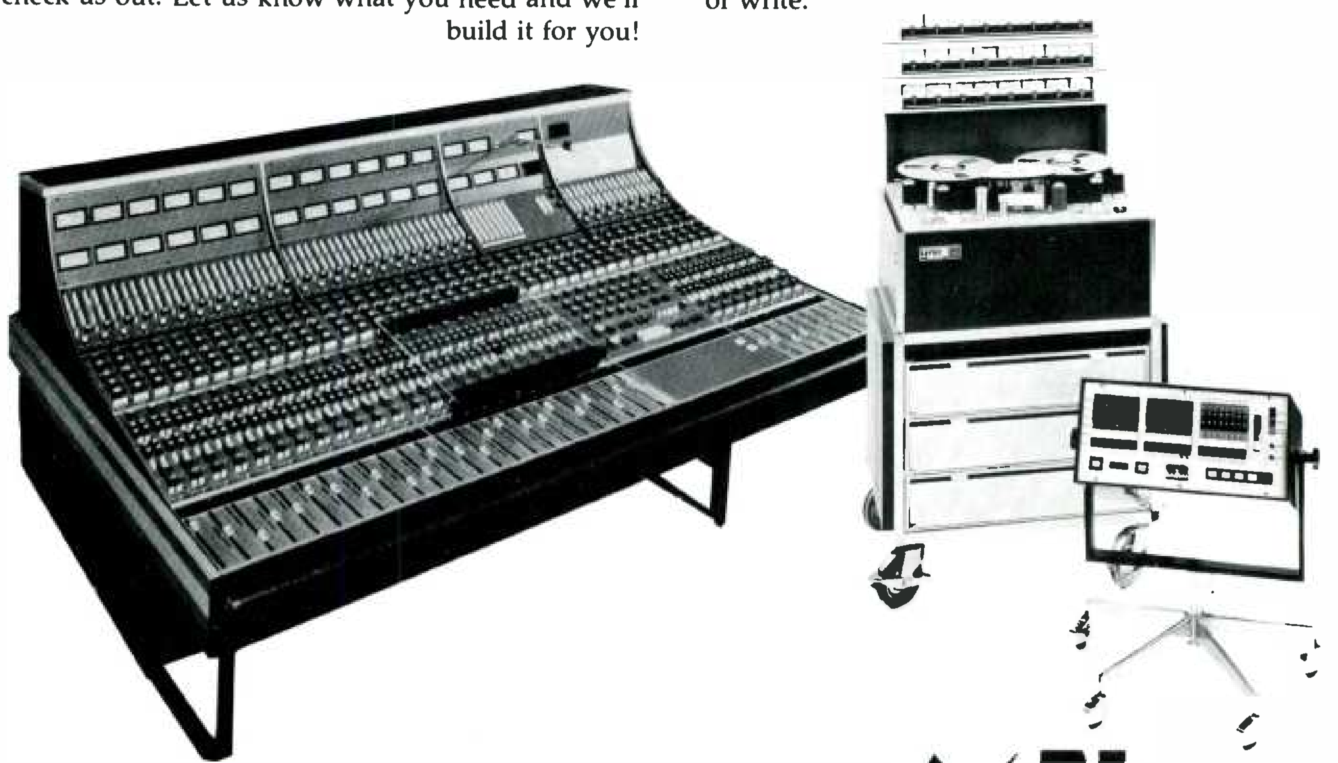

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Figure 1 shows somewhat idealized contours of equal loudness level in phons for signals below 100 Hz. in binaural free-field listening conditions. As an example, for a loudness level of 100 phons at 25 Hz., the SPL has to be 121.5 dB, as shown by a dot on the graph. This means that the tone will sound as loud as a 1,000 Hz. note of 100 phons (although its SPL has to be only 100 dB).

For a loudness level of 100 phons, we may write:

$$SPL_{100} = 160 - 27.5 \log f \quad (1)$$

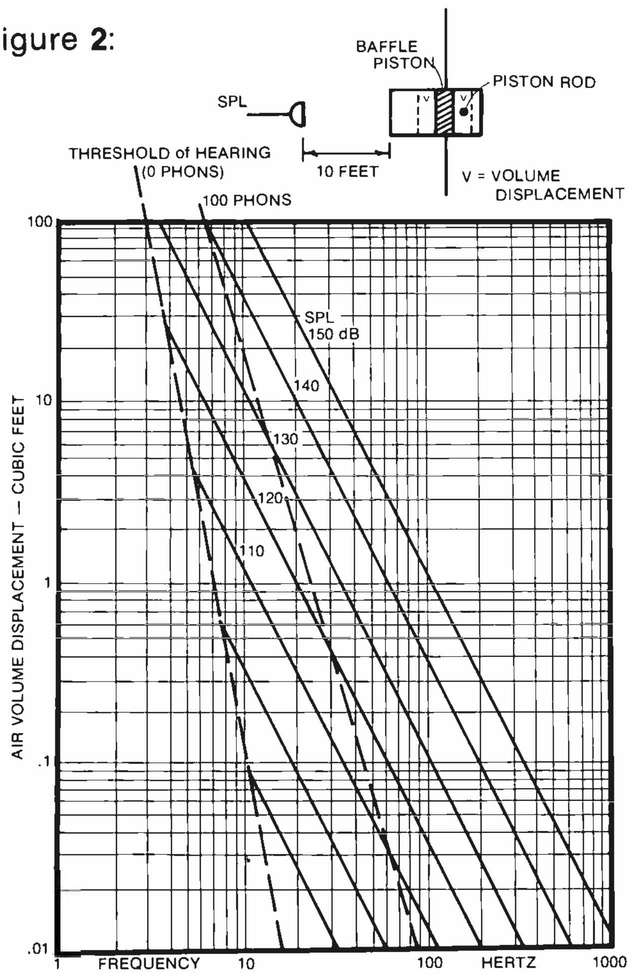
It should be noted that all loudness contours, whether for frequencies below or above 100 Hz., are not representative of a constant condition. Generally, at the end of a day, a person's hearing may be as much as 10 dB less sensitive to certain tones than early in the morning.

On Figure 2 are shown the required air volume displacements by a piston as a function of frequency to establish various SPL's at 10 feet from the pistonphone. As an example, at 20 Hz., for a loudness level of 100 phons, when the SPL at this distance has to be 124.3 dB, the cyclic air volume displacement by the piston from its mid position has to be 1.64 cubic feet, as indicated by a circle on the figure, and is equivalent to 2,834 cubic inches.

The equation for the required air volume displacement by a piston 10 feet from an observer is given by:

$$V = \frac{0.0004 \times 10^{SPL/20}}{f^2} \quad (2)$$

Figure 2:



By substituting (1) in (2) we get:

$$V = \frac{40{,}000}{f^{3.375}}$$

It is seen, therefore, that for a loudness level of 100 phons, the required air volume displacement varies inversely with nearly the fourth power of the frequency.

Since a 16" diameter cone has a cross-sectional area of 200 square inches, and a possible maximum travel of .5 inches from rest position (1 inch total), it will take 2,834/200 x .5 = 28 woofers to establish an SPL of 124.3 dB at 10 feet from the source!

As may be readily calculated, for a frequency of 40 Hz., the corresponding air volume displacement has to be only .16 cubic feet or 277 cubic inches. This can be accomplished with only 277/200 x .5 = 2.75, or only three 16" diameter woofers.

But how many loudspeaker cabinets can boast of even two woofers? It stands to reason, therefore, that the case of the missing lows is one of missing woofers.

When the loudspeakers are indoors, the cone excursions requirements are not quite so severe as they are outdoors, because the reflected sound provides reinforcement, particularly in the reverberant field of the enclosure. Another way of stating the situation is to say that, for the same acoustic power output by the woofer, the SPL at a given distance in a room will be greater than in free space.

As an example consider a room in which the "critical" distance is 10 feet. By that is meant the distance where the direct sound energy equals the generally reflected one. It is given by the equation:

$$D_c = .0315 \sqrt{\frac{V}{(1 - a)\,T}}$$

where A = average absorption coefficient of the room; T = reverberation time at a given frequency; and V = volume of enclosure, in cubic feet.

At the critical distance, the SPL will be 3 decibels higher than in free space for a woofer with a directivity factor of unity. This means that only half the acoustic power is required to establish a desired SPL at that location. Since the cone excursion is proportional to the square root of the acoustic power, the cone travel for half the power can be 30 per cent less, or 30 per cent fewer woofers are required (two instead of the three previously calculated for the 40 Hz. tone with SPL of 124.3 dB at 10 feet). Of course, for the reproduction of 20 Hz. under the same circumstances, there would still be 20 woofers necessary.

At 30 Hz. — the lowest organ note — we would still need an air volume displacement of .4 cubic feet or 691 cubic inches (for a loudness level of 100 phons at 10 feet), which requires seven 16-inch woofers in the open and five in the hypothetical room previously discussed.

It may be noted that for sound effects such as wind noises and earthquake rumbles, the reproduction of 20 Hz. and even lower notes is desirable, as was done in the Universal Studios "Sensurround" System. This was accomplished by the installation in the theatre of additional woofers in segmented horns.

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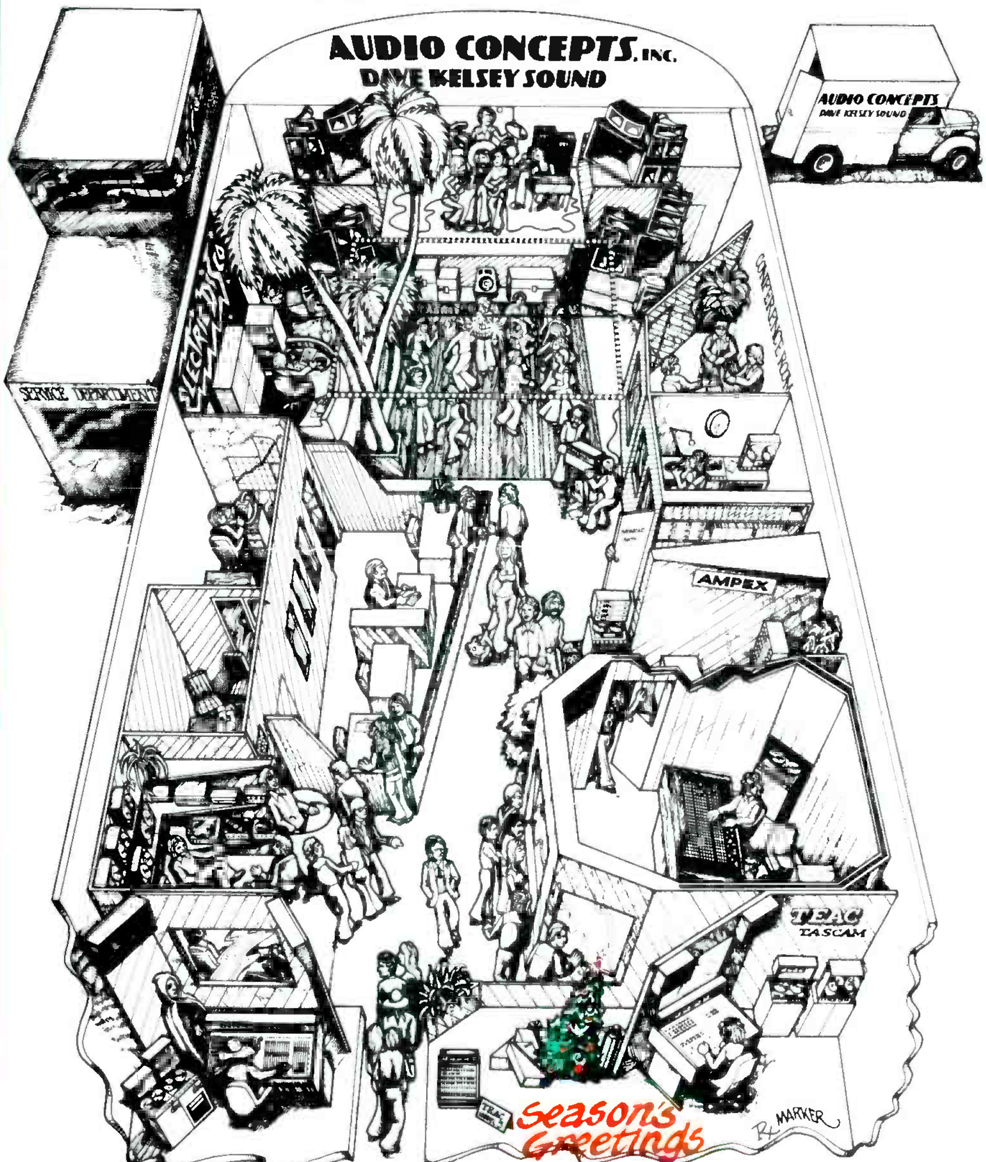

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an overview . . .

RECORDING the GRAND PIANO

by Paul Laurence

The piano that one hears with one's ears is so difficult to capture on tape that many engineers have yet to record it to their own satisfaction. Engineer Tom Perry (Boz Scaggs, Helen Reddy, Dean Martin, Glen Campbell, Crackin') among them, calls it, "personally the most frustrating instrument" to record. "I've tried everything from . . . I've tried every microphone in the world on a piano . . . I've tried a different number of mikes, different positions — underneath, above — lid completely off — mikes high, mikes low, mikes on the bridge, mikes on the sounding board, mikes in the back of the strings, three mikes, four mikes, two mikes, one . . ." Many confess to an uncertain relationship in general with the instrument. Perry again: "I don't know if it's just me or what, but for some reason I'm never particularly happy. I don't know what I even want to hear . . . somehow it never seems to sound like when you're just standing next to a piano."

Perhaps its elusiveness in even being *photographed* (much less *painted*) on tape can account for why there is not a whole lot of fooling around with the sound of the acoustic grand piano among the more mature engineers.

Mack Munich (David Bowie, the Electric Light Orchestra, Led Zeppelin, the Rolling Stones, Donna Summer) is one who typifies this position. After having taken considerable liberties with the instrument during his *experimental* phase he has come to eschew what he calls the *bent* or heavily processed piano sound. He describes the natural sound he likes as "bright, hard and clear", a good example being the Electric Light Orchestra's "A New World Record" album.

Bob Margouleff (Stevie Wonder, Billy Preston, Minnie Ripperton, Lothar and the Hand People) describes the natural piano sound he wants to hear as a "full-bodied piano sound with a lot of bottom, stereo, a big spread. A natural sound."

However, unnatural piano sounds which have contributed to hit records have to be acknowledged as well. Among the best examples which come to mind are those by the Beatles who have probably done more with the piano sound than anyone else. "Sexy Sadie", "Birthday", and John Lennon's "Imagine".

With the preceeding as an introduction it would seem to be constructive to take a look at several components of the problem as an aid to possible individual solutions.

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THE PIANO

To the best of everyone's knowledge, the first piano was built around 1709 by Bartholomew Cristofori of Florence, Italy.

Most of the instruments, as we know them today, have 88 keys, each of which, when depressed, activates a felt-tipped hammer which in turn strikes from one to three unison-tuned strings, producing a note. The piano also has dampers — small rectangular pieces of wood whose undersides are covered with felt — which modify the sound by moving closer to, or father away from the strings. The dampers are controlled by three independent foot pedals: the left (speaking from the player's point of view. His left hand is left, his right hand is right) or *soft* pedal, which shifts the entire harp to the right so that the hammers hit one less string per key. The middle pedal, called *"sostenuto"* or *sustaining*, when engaged takes the damper off whatever keys are depressed, sustaining that chord. The right pedal, known as the *damper* or *sustain* pedal, takes the dampers off all of the strings, not only sustaining whatever notes were last played, but also letting all the other strings vibrate sympathetically. The whole instrument is sustaining, in other words. Two or more pedals used simultaneously produce a more or less cumulative effect.

There are two main physical permutations of the acoustic piano. First and foremost there is the grand piano, it being the form used for the majority of piano recordings. The grand piano extends horizontally and measures anywhere from 5 to 9 feet long, the smaller ones being called *baby* grands and longer ones *concert* grands. Generally speaking, (for less than piano virtuoso solo recordings) the optimum size for contemporary recording purpose is said to be a 6 to 7½ foot grand. Beneath 6 feet the sound tends to be described as too *boxy*, too contained, and beyond 7½ feet the sound becomes too rich in harmonics, sometimes referred to as being *ring-y*.

The smaller, box shaped *upright* piano is typified by foreshortened strings anchored to a vertical harp, and is about 2 feet long.

BRAND AND SOUND

The difference in sound among the various brands of grand pianos are as a rule fairly subtle. They are pretty much all going for a similar purity of the tone. The main difference is really in the touch. Still, according to comment, there are identifiable subtleties: Steinway & Sons is very much the standard, and is known for its uniformity of tone across the keyboard, full but not too-rich sound, and percussive qualities. Those are Steinways on the Beatles' "You Like Me Too Much", John Lennon's "Imagine", the Allman Brothers' "Jessica", and the Rolling Stones' "Melody".

Yamaha has lately become very popular for studio use, and is considered to be among the brightest sounding. That's Musicland's (Munich) 6 foot Yamaha grand on the Stones' "Hot Stuff" and "Memory Hotel".

Baldwin has recently shown an increase in popularity as well. The instrument is known for its heavy action and full, "big" sound.

Mason-Hamlin is perhaps the loudest and richest of the lot, and is known also for its heavy action. Neil Diamond swears by them.

The Bluthner grand is celebrated for its uniformity, a sound that is clear and strong at both top and bottom, as well as for its light action. That's a Bluthner that Paul McCartney is playing at the Twickenham Sound Stage for the "Let It Be" album.

Bechstein is a loud, bright, and percussive piano, and finds great favor with classical people. Trident Studios' Bechstein is heard on Elton John's "Your Song" and "Take Me To The Pilot", and Queen's "Dear Friends".

The Bosendorfer grand has 92 keys and is know for its amazing action and mellow sound. Al Stewart's "Year of the Cat" cut features a Bosendorfer.

TUNING

Obviously, the sound of a piano will depend on its tuning as well as any individual treatment or adjustment it may have undergone. The practice of tuning sharp — anywhere between 441 and 444 (the European middle "A") is still very much in evidence. Still quite valid is the paragraph relative to tuning from the article appearing in R-e/p during 1971 (*Recording The Piano — A Forum, Vol. 2., No. 2, March/April 1971, pages 11-17).* "... the reason for tuning sharp is to get the whole band changing ... nobody could lay back, or they would get lost."*

"While there is quite a bit of sharp

**Photocopy reprints of "Recording the Piano" from the March/April 1971 issue are available. Send $2.00 to R-e/p, P. O. Box 2449, Hollywood, California 90028.*

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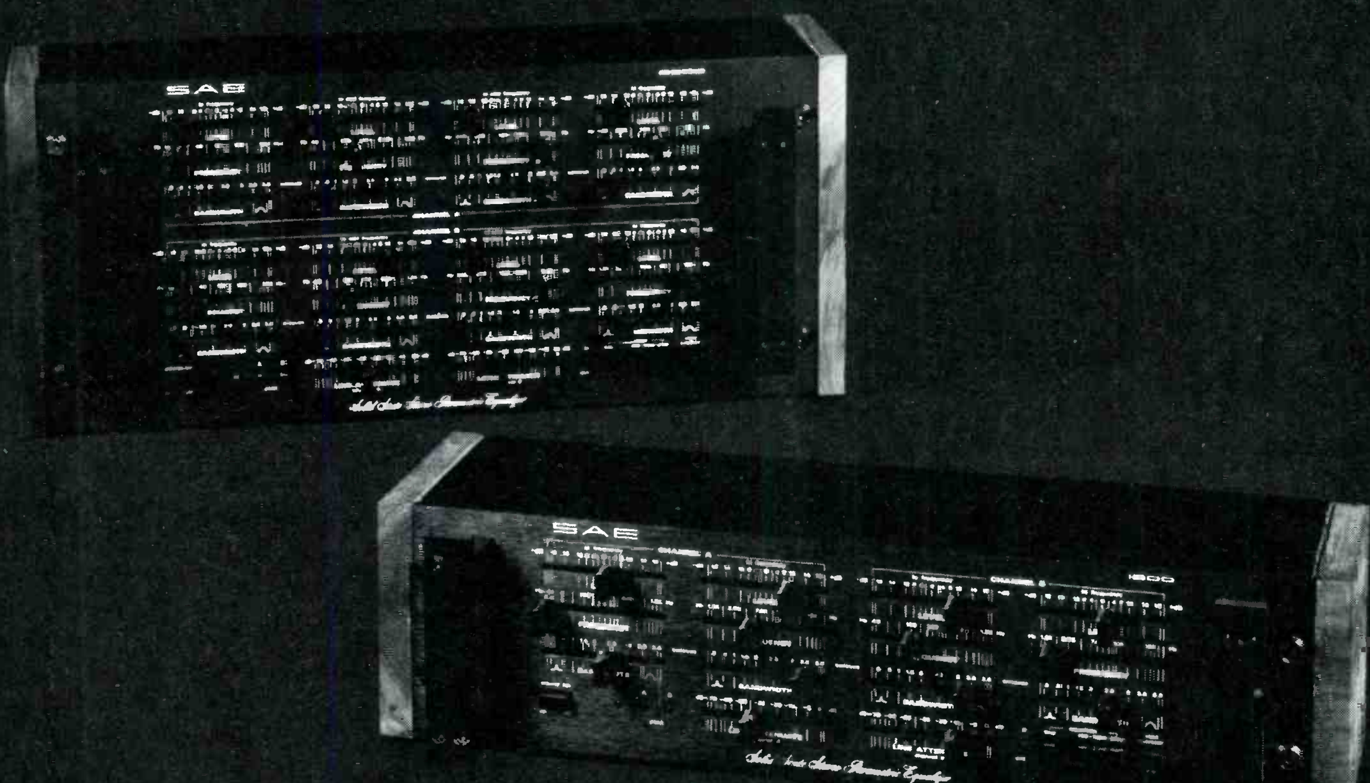

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tuning on straight-ahead band sessions, the technique just doesn't work at all well when sharp piano is played against electronic instruments which cannot change intonation to get up with a sharp piano, or sharp strings."

The brightness of a given piano can be adjusted by a tuner manipulating the pins imbedded in the hammers. Pulling them out more will add more attack and brightness, pushing them in will do just the opposite.

However, regardless of specific make or tuning the piano is a true binaural instrument, a large instrument with the sound emanating from a broad surface area. The instrument produces what might be seen as three essential types of sounds. First, of course, are the actual musical tones. The piano has a fabulous frequency range, from about 27.5 Hz. to 4,096 Hz. for the root notes, with the (useable) harmonics going up to about 12 kHz. Secondly, there are the percussive or mechanical noises attendant to producing the musical sounds — the actual sounds of the hammers hitting across the strings, and when the piano is miked from the player's side of the keyboard, the sounds of the keys hitting the wood. Lastly, the piano emits considerable unwanted and extraneous information consisting of both low tones and high harmonics.

These factors help explain why it is oftentimes claimed that piano is an instrument difficult to get into perspective compared to others. So, inasmuch as there are the usual different notions of just what a well-recorded piano should sound like it seems that only one criterion meets with any sort of general acceptance. *Consistancy* is certainly that order; where no notes either *jump-out* or *get lost* because of difference in tone or level. Ideally, it is agreed to by most that the instrument should sound smooth and even, from octave to octave, across the keyboard. Likewise, very few go for a particular occasional abereration in such perspective where one or more treble notes seem to be coming from the bass region, or vice-versa. Beyond consistancy, there is some temptation to further generalize that clear articulation of notes is a minimum requirement. But, no, there seem to be many who eschew close miking, opting for the more distant, open, airy, and "fuzzier" sound.

PLACEMENT AND ISOLATION

The grand piano, because it is a large acoustic instrument, and particularly when it is not recorded using very close miking, tends to pick-up considerable leakage from other instruments. In the great majority of cases this is considered a problem, but every once in a while that very same leakage can be an asset, and the instrument wants to be positioned out in the recording environment to take advantage of its sonic magnetism. With this in mind it might do, now, to look at several of the techniques used by various studios to position the piano.

Among the most sophisticated means for isolation are the completely separate, trapped, piano rooms installed at many studios. At Caribou they have built an acoustically-lined piano cut-out in the studio wall, which yields the dual advantages of absorbing almost all of the piano sound, at the same time keeping the instrument and its player in the same environment as the other players. At Caribou the microphones are hooked onto the ceiling of the cut-out. Elton John's "Rock of the Westies" LP was cut at Caribou as an example of how the cut-out sounds.

Another sophisticated approach to isolating the piano in the center of the recording environment is the piano box Gus Dudgeon had built for Elton John. The box was built to match the piano's configuration, 4 feet tall. The interior is damped with fiberglass, all except the ceiling of it which is left untreated and is a reflective surface. Using this box, Gus has five feet (including the 12 inches from the top of the piano down to the strings inside the piano) to work with for his piano sound. In this way, he has found it possible to get away from the unwanted sounds too close to the strings, without any loss of isolation. The sound Gus gets for Elton is described as a fairly distant, binaural, but clear piano sound.

By far the most commonly used tools for controlling the piano's sound is the combination of position in the studio, the position of the grand piano's adjustable lid, as well as the use of a variety of damping materials such as furniture blankets and other draperies. Essentially, how the lid and/or damping materials are arranged, in other words, how the piano is allowed to breathe, determines its ambience. Those who like the lid down often refer to a "tighter sound", "better isolation" and a "stronger bottom end". Those devotees of the open lid are heard to criticize the "too confined sound", "a lot of unwanted mechanical noises", not the least of which being bottom build-up . . . boomy-ness. Admittedly, a consistant problem for the up-lid-ers is that of the piano player singing a live vocal.

Among those who do away with the lid completely is Bob Margouleff who prefers to record piano completely unfettered, and the only instrument in the main room.

ARRANGING THE PIANO

As we have seen the acoustic piano is an extremely expressive and versatile instrument. From a recording standpoint, after guitar, it is probably the instrument most represented on American and English pop music records of the last twenty years. The piano is extremely melodic — much more so than rhythmic — it lends itself easily to orchestration, and all things considered is probably the instrument most closely associated with modern-day English language pop songwriting.

From an arranging standpoint, the acoustic piano can do just about anything and can therefore be used just about everywhere. As a song's featured instrument, as *color*, just *trimming*, as a

. . . *continued on page 48* —

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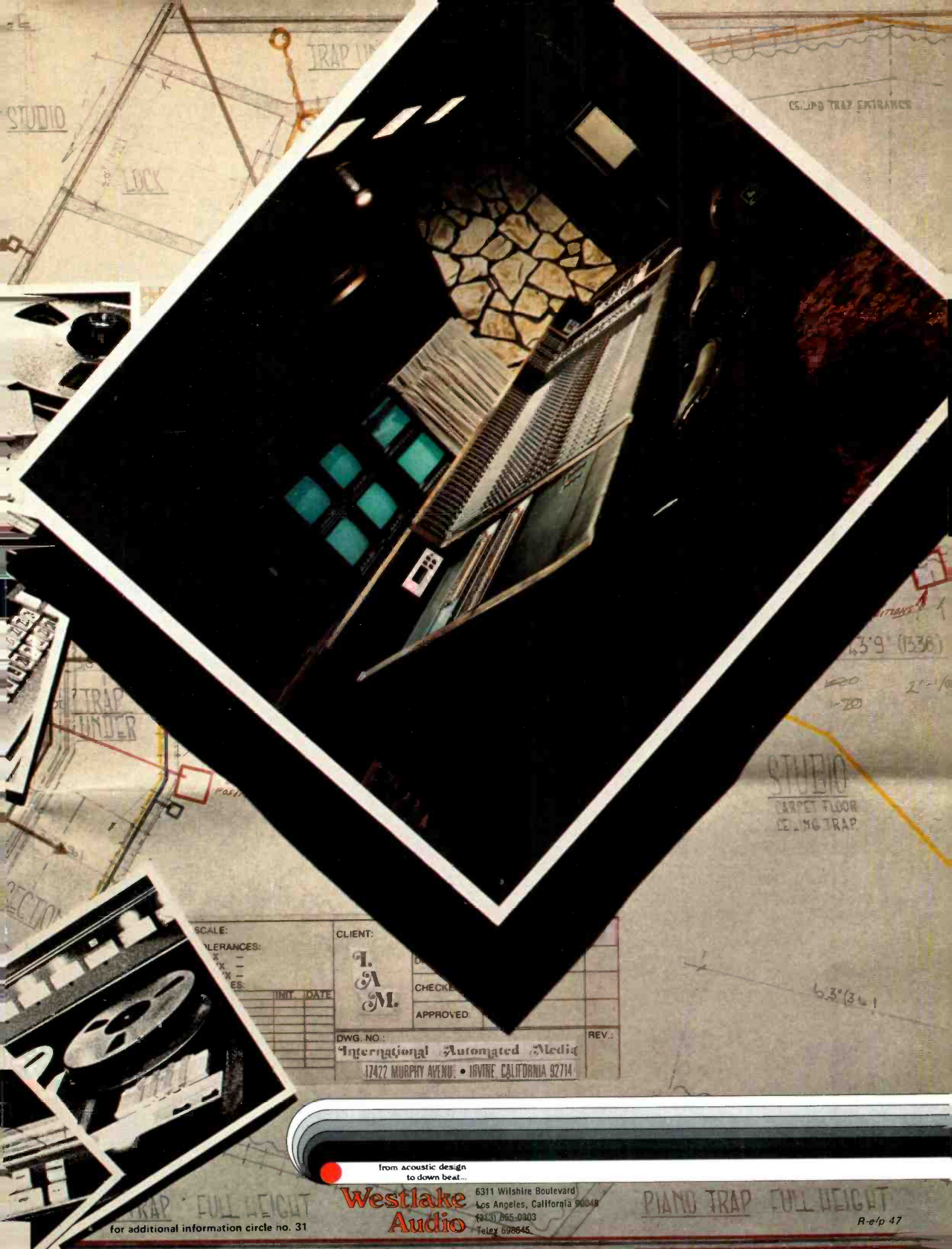
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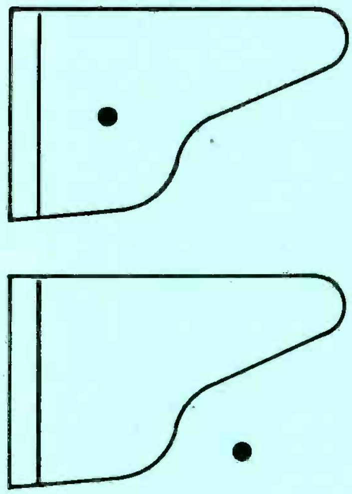

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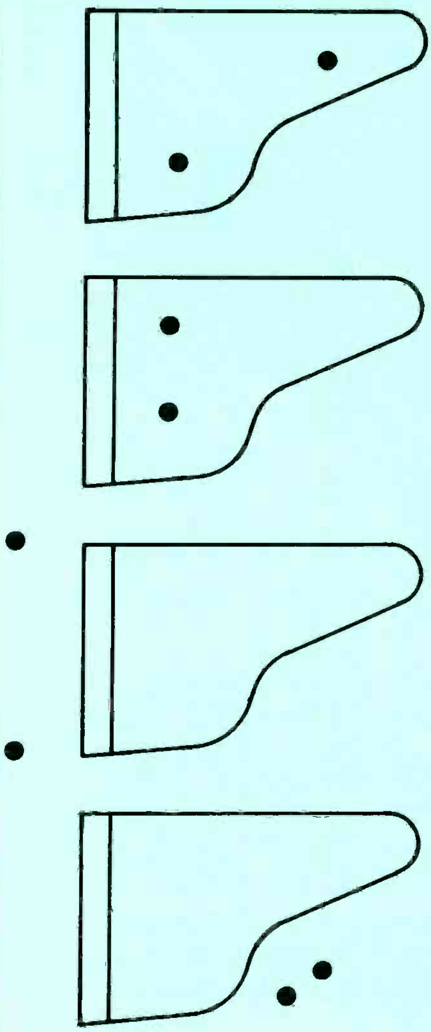

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texture, solo . . . name it. The essential clue to its versatility is of course its huge range, extending 11 octaves from A, 39 notes below middle C to C, 48 notes above middle C. Each hand can play five or even more notes at once (though few players go higher than eight, total) and anything from single notes through fragments and into complex chords. Then, too, the piano can be played several ways although almost everyone plays it from behind the keyboard, depressing the keys with fingers, generating a traditional range of sounds. However, there are people, a good example would be Jim Price and the Doors, who have other ideas. It can be used in such a way as to sound like a cymbalon, by merely going back to the strings and plucking them with one's fingers, which is what Jim Price did for the intro of the Rolling Stones' "Moonlight Mile". Or, you can do, in the words of Bobby Krieger, "inside the piano-strings fooling around", as the Doors did on their avante-garde piece "Horse Laditudes".

These aberrations aside, piano is a large instrument, and in most cases sounds like it. Most piano parts contribute a fair amount of notes, a lot of mass, and a lot of oomph to a song, meaning that it also has the potential to contribute considerable confusion as well. The solid-ness of the instrument plus the fact that he's undoubtedly playing a considerably greater number of notes than anybody else, probably in that same (middle) tonal range as most of the other instruments, makes it easy for the piano to simply wash-out the entire recording to a point where it is virtually impossible to hear what is going on. Additionally, there is the anomaly to contend with of the acoustic piano and the electric guitar, for some reason, can just naturally antagonize each other.

So, controlling the piano as a function of its size relative to the production at hand, as can be seen, is the job of both the arranger and the engineer. Simply stated, the piano can be described as either *little* or *big*. It's size most usually is a direct reflection of its importance to the song; if it is just trimming it would probably be small and would be taken mono for the mix; fairly important and it would be larger (very likely stereo); and if it were the featured instrument it might very well be recorded for large-spread stereo.

Not unlike other instruments, you'd record, a *small* piano would have at least some of the following characteristics: one mike, close-miked, limited, rolled-off bottom/boosted top, mono, little level and echo in the mix. By contrast, a *big* piano would have among the following: multi-miked, close and distance miking, boosted lows, mids and highs, stereo, big spread, lots of level and echo in the mix.

MICROPHONES

Assuming some amount of elevated importance of the piano part in a song, current use of microphones would most often, these days, require two. Each one addressing what might be loosely described as the bass or treble, corresponding to the left and right-hand regions of the keyboard. Exact placement, as can be imagined, must be governed by the individual player; that cluster of variables called *dynamics*. What he is playing, from how soft to how loud. People like Billy Preston and Elton John are very hard, rhythmic players, and as a result have a strong attack, and a reasonably crisp sound to begin with. Another factor governing mike placement is where on the keyboard he's playing. Stevie Wonder and David Paich, say, do mostly block playing in the middle region, whereas others like Nicky Hopkins and Keith Emerson are extremely aerobatic, moving all up and down the keyboard. In any case, dependent on the number of tracks available each mike would almost certainly be assigned to a separate track.

The practice of using one stereo microphone has its devotees, yet the criticism of those who have tried and discarded the single stereo mike is that it gives too even a pan, too little left/right separation, too little mixing flexibility.

There is considerable difference of opinion as to what happens when more than two microphones are used on piano. Bob Margouleff has found three microphones wholly unsatisfactory, lending to serious phase cancellation effects and problems in mastering. Others like Ken Scott regularly use three Neumann condenser mikes, finding it an entirely workable format. When a third mike is added it would more than likely be centered, both in terms of its positioning relative to the two other mikes, and being sent equally to both tracks on the tape. A third mike can, and is most often used to fill-in the hole that often appears in the center of a wide stereo perspective, or for ambience or texture if placed more

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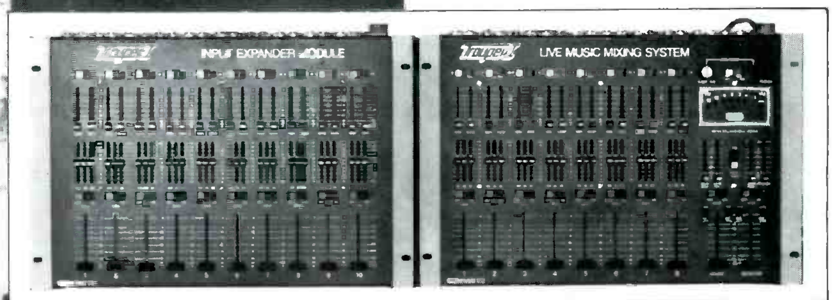

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distantly.

Four mikes are occasionally used, in which case they would more than likely be approached as two fairly symetrical pairs, say, a close left and right augmented by a distant left and right. Tom Perry has used two up above the hammers, and two in the back. He considers that it worked for that particular tune. "It was okay — it wasn't startling."

MICROPHONE PLACEMENT

There are a number of basic regions from which the piano can be miked. It is most often miked from above. One mike generally goes right over the middle. The sound holes are occasionally used. There are usually three but on some models a number of smaller ones as well, function to help the piano sound project. Occasionally they prove effective for keying mike placement. Mikes have been placed around and sometimes in the sound holes (usually one mike over the middle hole in the mono days) but in recent times most people have had little luck with this placement except where the emphasis from this very localized pick-up of selected notes (strings) is the objective. In short, microphone placement in, or close to any of the holes produces uneven level and tone, selecting for the notes and frequencies directly beneath them, perhaps masking what is coming from the rest of the keyboard. Also, the sound in general is said to be constricted. Mack says, "The effect is not marvelous. Still it does work in cases where the player is playing right under the hole or holes being miked, and you want that kind of constricted sound."

The piano is often miked from any number of points around the periphery, lending a looser, funkier, roomier, old-timey kind of sound. The most common location is probably off to the right, near the middle of the curve to capture the sound deflected by the lid.

Mack used peripheral miking for some of the piano overdubs on the new E.L.O. album, with a Neumann KM-84 set to cardioid behind the player, and a tube style U-47 set to omni, three feet beyond the back of the piano, both mikes six feet off the ground.

Rarely is a piano miked from below, and when so, almost always for greater piano isolation.

The normal range of *close-miking* seems to be from 8 to 30 inches from the strings. Closer than 8 inches can produce undesirable components; uneven dynamics (the notes closest to the mike being too loud), uneven perspective (too exaggerated a *hole*), and unwanted content. A piano too close-miked might reveal too much hammer sound as well as curious ringing, mushiness, wooliness.

In conditions of complete or nearly complete isolation, the acoustic piano can be fairly distantly miked. According to Andy Johns the piano on the Rolling Stones' "Moonlight Mile" was all alone in a completely empty room at Stargroves, having a wooden floor and plaster ceiling and walls, and miked with one U-87 four or five feet directly above the piano. The lid was completely off.

John Sandlin (the Allman Brothers, Bonnie Bramlett, Cowboy) often mikes Capricorn's nine foot Steinway (once the resident piano at Carnegie Hall) from 10 feet away on overdubs.

LIMITING

Limiting/compression was used for a time on the piano — particularly in England — but is not used that much any more. An integral part of the "English" piano sound, it lends what Jimmy Miller refers to as a "grindy" sound, as opposed to the "pure" or "open" sound. The Audio & Design and Neve limiters are reported to be excellent on piano, as are the UREI LA-2A's, LA-3A's, and 1176's.

THE VARIOUS AESTHETIC QUALITIES OF THE ACOUSTIC PIANO AS RELATED TO EQUALIZATION

		BOOSTING
0 - 250 Hz.	:	Warmth, roundness, rumble, fullness
250 Hz. - 1.5 kHz.	:	Fullness, roundness, warmth
1.5 kHz. - 3.5 kHz.	:	Hardness, metallic quality, edge, percussiveness
3.5 kHz. - 12 kHz.	:	Sharpness, pointiness, crispness, edge, percussiveness
12 kHz. - and above	:	Ringiness, noise, fuzziness
		CUTTING
0 - 1.5 kHz.	:	Coldness, iciness, flatness, hardness, edge
1.5 kHz. - 3.5 kHz.	:	Hollowness, percussiveness, coldness
3.5 kHz. - and above	:	Drabness, roundness, murkiness

EQUALIZATION

Having such a full tonal range, the piano responds well to equalization. Boosting under 100 Hz. can, as with most instruments, bring up rumble. And likewise, boosting above, say, 12 kHz. can emphasize what are probably unwanted overtones and cymbal leakage. Interestingly, a piano may be made to sound somewhat like a tack piano by cutting in the 1.5 to 3.5 kHz. region, which seems to remove the "fullness" or "body" of the notes, thus emphasizing the attack and leaving that "hollow" sound.

STEREO PIANO

Stereo piano emerged — quite incidentally, though — in the late 50's or early 60's. It was because some consoles back then didn't have a center buss, and so the best way to center the piano was to use two mikes and bring one up left and one right. The Doors did stereo piano on "Yes, the River Knows" (July 1968), but the institution of Stereo Piano didn't really begin until the 16-track era.

Stereo piano nowadays is probably the norm. It is usually medium-spread, and, in instances where piano is the song's main instrument and/or it's a record by a featured artist who plays piano, often given a full spread.

the 3M system:

DIGITAL AUDIO RECORDING

Jack Mullin, one of the pioneers of the recording industry in the U.S. said it just last month: "From this moment, the future has to be digital." Jack was reacting to the demonstration of a commercial mastering system introduced by 3M Company at the New York AES Show. Jack, having been associated with 3M for many years, is naturally enthuiastic about that firm's accomplishments, and we might think his comments were unduly prejudiced.

But every indication is that his comments were not inflated. 3M seems to have pulled it off, introducing a practical 32-track digital mastering system which uses one-inch tape. The system was developed in conjunction with the British Broadcasting Corporation, which also has been deep into digital for many years.

WHY DIGITAL — DIGITAL VERSUS ANALOG

Digital recording offers a complete absence of flutter; timing errors; print-through; amplitude variations caused by tape or minor contact loss, or both; modulation noise caused by high-frequency (scrape) flutter (Figure 1); and erasure problems.

Besides, digital technology has very favorable specifications for a number of characteristics:

— Signal-to-noise ratio greater than 90 dB without noise reduction systems.

— Harmonic and intermodulation distortion less than 0.03 per cent above 100 Hz. at maximum input/output level (+28 dBm), and across the entire passband with levels not exceeding +15 dBm (Figure 2).

— Crosstalk down to 90 dB or more between any two tracks, in the worst case, including the sync mode.

Finally, digital technology offers a bandwidth of 20 kHz. and has excellent track-to-track uniformity.

DIGITAL PRACTICALITY

Digital recording, at least in theory, is not new. The possibilities of noise-free, distortionless audio recording have been apparent for some time. But the big stumbling block has been to develop it on a practical basis.

A great deal of digital experimentation has taken place — Soundstream, Nippon Columbia, Sony, Mitsubishi are names that come to mind.

The contributions of these developers is not to be minimized. But, it would seem, there are two areas where the 3M/BBC development is extremely significant: the new system has a sampling rate of 50,000 per second, and it has only one track per audio channel.

The sampling rate is important to fidelity. Sampling simply means the amplitude measurement of the electrical output of an analog waveform to derive a digital value. Obviously, the more often you sample the original music — how thin you slice it — the more faithfully you re-create the original. If you

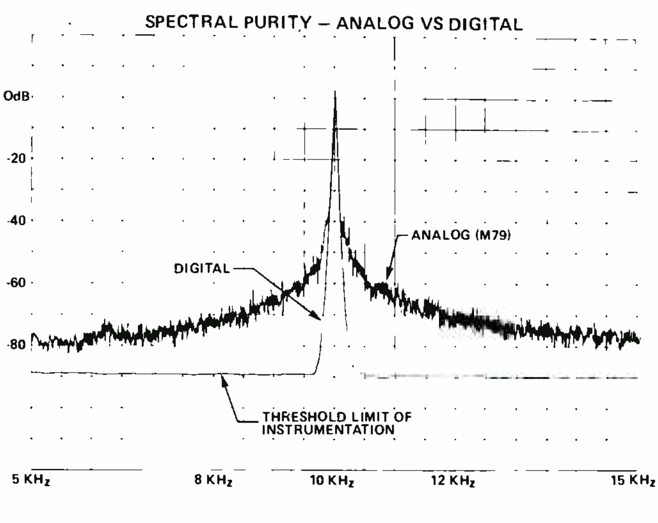


Figure 1: **Modulation Noise caused by high frequency (scrape) flutter (a spectral purity problem).**

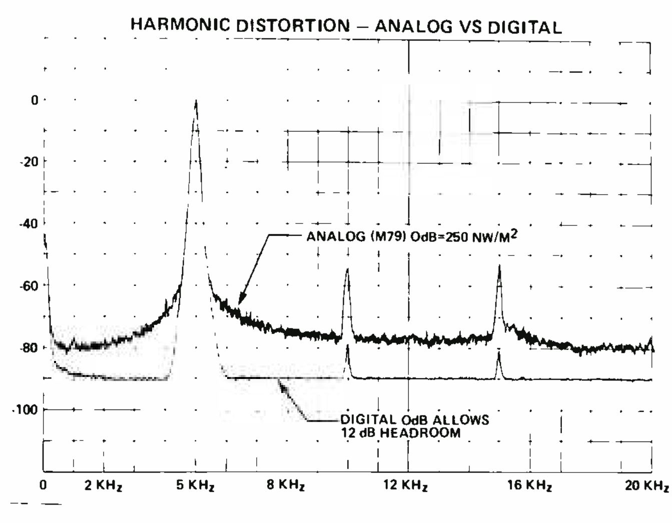
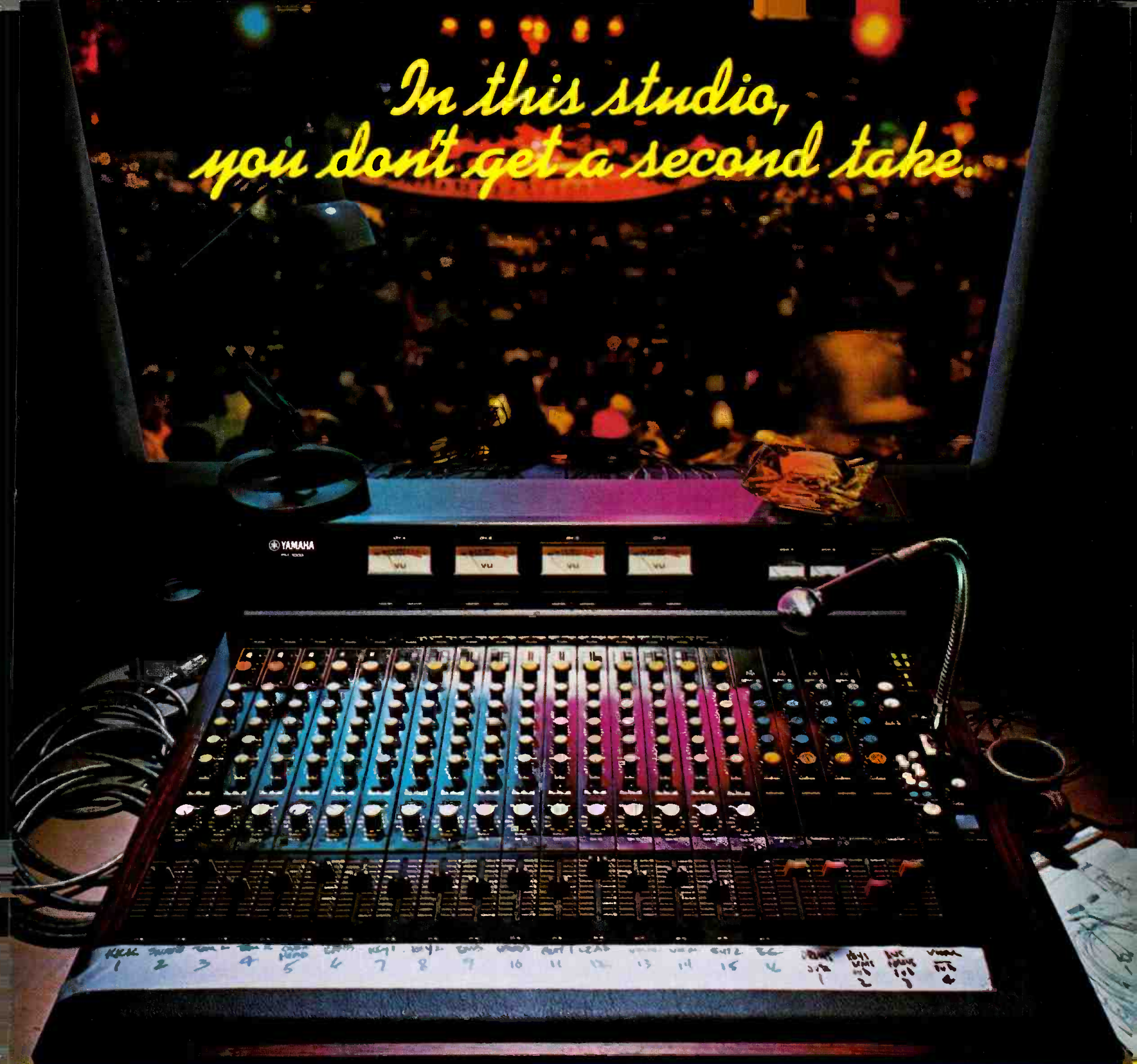


Figure 2: **Harmonic Distortion <0.03% above 100 Hz. at maximum input/output level (+28 dBM) . . . across the entire passband with levels not exceeding +15 dBM. On analog recorders, distortion is typically specified at 3% at a single frequency at maximum record level.**



When you perform in front of a live audience, you put everything on the line.

That's why you're so careful in selecting sound reinforcement equipment. Because once the music starts, you can't afford to have it stop.

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Tough isn't enough. Realizing that every job has different sound requirements, Yamaha also designed the PM-1000 Series for maximum flexibility. With features like an exclusive 4x4 matrix with level controls that allows four independent mono mixes.

There's also the complete complement of controls you'd expect to find on the most sophisticated consoles. Transformer isolated inputs and outputs. Dual echo send busses. An input level attenuator that takes the +4dB line level to −60dB mike level in 11 steps. Plus 5-frequency equalization. To give you plenty of headroom for clean, undistorted sound, the PM-1000 can drive a 600 ohm load to +22½dBm.

Get your band on the wagon. All around the world—night after night, gig after gig—you'll find Yamaha mixing consoles the choice of more and more professionals. People who don't regard professional quality as a luxury, but as a necessity. Your Yamaha pro sound dealer can give you all the reasons why you should join them.

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sample at the rate of 100 times per second, electronically it is much easier, but the fidelity is poor. Each order of magnitude in sampling rate makes a tremendous difference.

Some systems have gone to 30,000 times per second. But one rule of thumb says that the rate should be more than double the highest audio frequency you plan to sample; thus, 30,000 means an audio ceiling of 15 kHz. — and more likely, 12 to 14 kHz. The 50,000 rate used in the 3M/BBC system means a ceiling of 20 kHz. or better — which permits overtones and other intricacies of tone typical of full-concert performances. Certainly, 20 kHz. is beyond the range of most humans.

Then, the practical convenience of only one track per audio channel is good news to studio mastering engineers, because of the ease of mixing and dubbing. Further, the 3M/BBC system puts 32 tracks on one inch of tape — unheard of before! — yet runs at a practical speed: 45 ips.

DATA PROCESSING TO DIGITAL RECORDING

CONCEPT CONVERSION KIT:

Digital technology, including error correction, is the backbone of the data-processing industry. To readers with a digital background, "parity", "cyclic redundancy check" and some other concepts may be no problem. But to help those without such a background, the obviously oversimplilfied "kit" is offered as a "bit" of help.

BINARY NUMBERS

Electronic manipulation of numbers uses the lowest range of numbers: two. Only 0's and 1's are used to make up the numbers in even the most complex digital system. This is because the numbers are easily represented by the binary state of electronic circuits: on or off, positive or negative, magnetically "north" or "south", etc. Larger numbers are simply made up (as are our "normal" numbers) by putting the binary digits ("bits") in order of magnitude:

0 = 0; 1 = 1; 10 = 2; 11 = 3, etc.

A value of any size can be represented by binary numbers if enough are put into place. Each order of magnitude is twice the adjacent one; thus, a 16-bit word as used in the 3M Digital Audio Mastering System can hold any value up to 64,536.

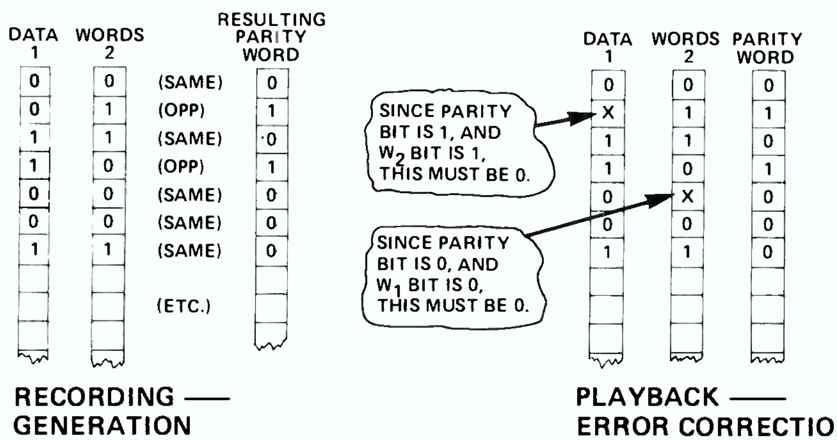
PARITY

This is a technique to ensure correct transmission or storage of bits in electronic systems. It's something like "odd man out" when you match for coffee. The corresponding bits of two binary numbers are compared, and a third — parity — word is created, whose corresponding bit depends upon what the other two are. In the 3M system, if both "data" bits are the same, the parity bit generated is a "0"; if they are dissimilar, the parity bit generated is a "1". Then, later on — at playback, or at the other end of a transmission — the corresponding bits are compared. With the "ground rules" by which they were originally compared and the parity bit generated, it is possible to re-create a bit in a missing or "damaged" word.

DATA PLACEMENT ON TAPE

Since digital notations are simply groupings of numbers, related ones can be on non-adjacent locations on a tape. As long as any digital "word" — a specifically identified digital grouping — can be "labeled" in some manner, it can (within limits) be at any place on a tape and yet routed into the proper "stall" for parity comparison or other treatment. Conversely, words can be routed from temporary storage circuitry to specified positions on tape.

PARITY COMPARISON



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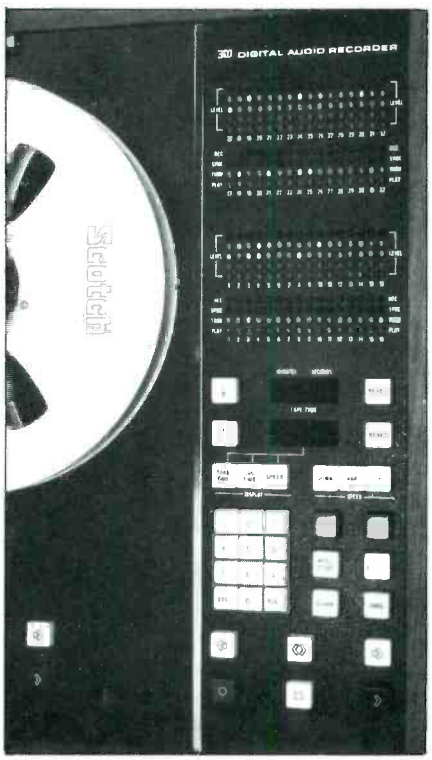
Bringing the technology within everyone's reach.

The New *3M* Digital Audio Mastering System . . . *records 32 audio channels on special 1" tape. This unit and the system's 2/4 track recorder (not shown) are each comparable in size and style to present day professional recorders.*



Close-up view of the operator's panel of the 32 channel, one inch digital recorder showing operational controls including built-in electronic position addressing.

The control panel can be lifted out of the recorder cabinet for use at nearby remote locations.

Other developers have traditionally been plagued by the need to have redundancy of up to four tracks per channel, or excessively high speed — or both — to compensate for drop-outs.

As 3M developers explain, there was close cooperation between 3M equipment designers and 3M magnetic tape specialists, so that one of the major development studies concerned itself with the realistic incidence of drop-outs and their distribution on tape. Armed with the statistics which could tell the probabilities of drop-outs, the equipment designers could undertake to build an error-correction system that is compatible with those probabilities.

3M, a pioneer in magnetic tape, could draw on its experience in audio, video, and scientific instrumentation tape and equipment — both for statistics, and to develop the best tape for the new digital system.

The streamlined error correction system that resulted is what permits the single track, practical-speed mastering recorder that 3M plans to start placing in commercial studios in 1978. What's the secret? Here's a brief description.

THE ERROR CORRECTION SYSTEM

The 3M Digital Audio Mastering System requires only a "50 per cent redundancy", compared to the two-times redundancy typical of some earlier systems. This is what makes the single-track configuration possible. Each 16-bit digital word needs essentially only eight "redundant" bits to ensure its accuracy. Packing 28,000 bits per linear inch then allows the relatively slow speed of 45 ips.

The recording circuitry is block-diagramed in Figure 3A. Analog signals from the microphone are amplified, routed through an anti-alias or low-pass filter to reject spurious frequencies which could beat against the 50 kHz. sampling rate and cause unwanted audio frequencies, then into the A-D converter and the recording encode circuitry. The recording mode is delay-modulation or MFM; the encode circuitry also generates the parity word for each two data words compared and then puts the component words down on the track (the order must be defined, but words generated in sequence by the analog sampling need not be adjacent on tape).

Figure 4 shows further detail of the record encode function. Parity comparison is made of two data words which are not adjacent — thus, a dust particle or other problem causing a drop-out is statistically most likely to interfere with only one of the two words being parity-compared, because the other is physically distant from the first.

The two non-adjacent words are compared and a parity word is generated. The record encode circuitry holds and assembles digitized data into blocks comprising 16 data words, eight parity words, a cyclic-redundancy check (CRC) word and a sync word. The latter two, shorter than 16 bits, mark the border between one block and the next, and insert CRC signals which later can be used to flag an error in the block.

Digital data, as it is generated and organized by the record encode circuitry, is laid on the tape (analog "sine" wave) with no "awareness" that errors may be encountered, but in a manner that permits their detection.

The playback circuitry, block-diagrammed in Figure 3B, is the inverse of the recording circuitry but has the additional burden of dealing with potential errors.

Pre-amp, equalizer and bit-sync circuitry massage the analog bits as they are played off the tape, ensuring that they are in phase and of proper amplitude to be read as a digital non-return-to-zero (NRZ) code.

The time base corrector (TBC) eliminates flutter and wow problems. Words are stored temporarily, and gated out of the TBC under control of a crystal oscillator. At the same time, the comparison between their arrival rate (determined by tape speed) and the clock results in servo signals which ensure that their average arrival rate remains within bounds — there is neither a data lag nor pile-up of any significance.

The error corrector examines each block of 16 data words and eight parity words, looking first at the CRC word. CRC logic compares the played-back block with the recorded block and triggers further error-correction circuitry into action if necessary. The details of the CRC logic are not being divulged, but an extremely oversimplified analogy could be something like this: if you were going to add up a long list of numbers derived from, say, sales tickets, the first check would be to see that the number of counting bits in the block, is able to indicate whether a problem caused by drop-outs has occurred.

If such an error has occurred, a run through of the data words in that block is in order. The comparison allows the reconstruction of "damaged" words (See the "Concept Conversion Kit" accompanying this article) so that an error is not in existence by the time the words reach the D-A converter.

In cases where a pair of drop-outs is so located that *both* data words are clobbered, reconstruction is impossible. Statistics are on the side of the listener, though, because 3M has found this sort of problem occurs very rarely. When it does, the playback system simply mutes for

FIGURE 3A.

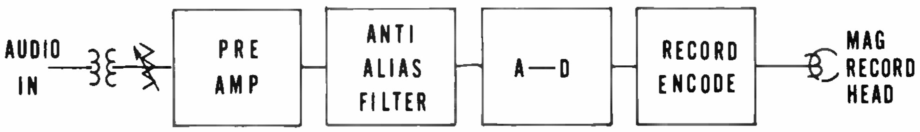
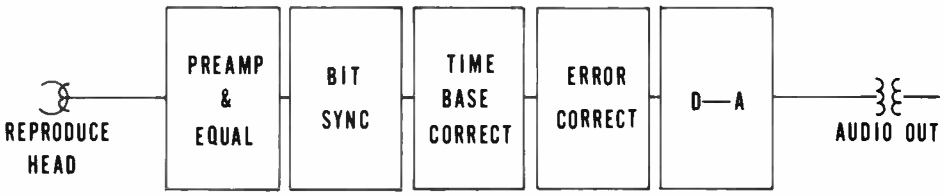


FIGURE 3B.



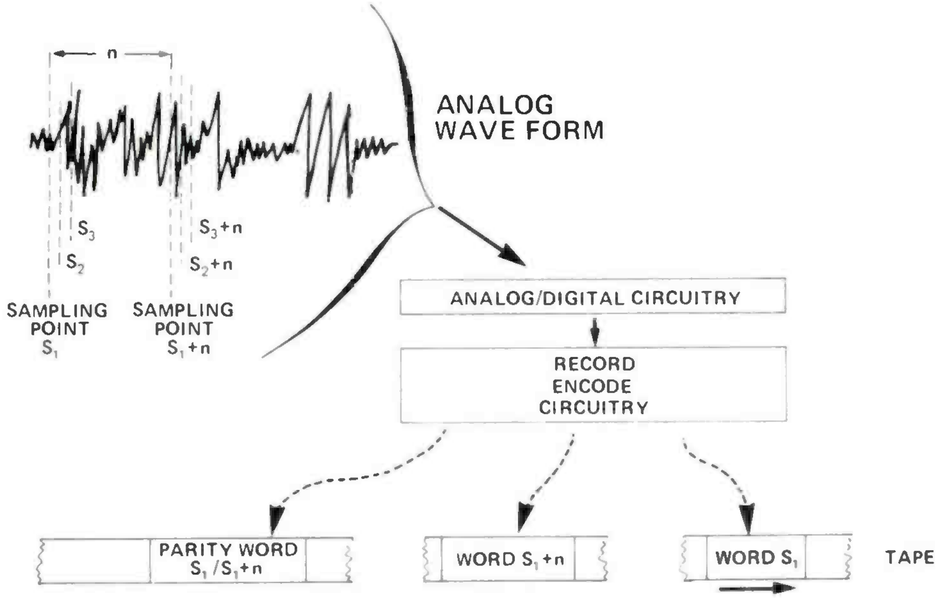


FIGURE 4. SIMPLIFIED ERROR – CORRECTION SYSTEM MAP

an extremely brief interval.

THE RESULTS

The new system works, as the demonstration at the AES proved. The 3M/BBC effort brought about a recorder that looks and acts like those the recording industry is already familiar with. Theoretically, the new machine could simply replace a present analog multi-track recorder (while providing more tracks than were previously available).

Another important advantage claimed for the digital system should be of great interest to professional engineers: simplified maintenance. Tweaking an analog recorder is an arduous process in some shops, requiring time and talent that could better be put to other use. In the digital system, since the signal laid down on the track is less complex, there are only a few adjustments necessary such as the input signal amplitude, which is a simple go/no-go setting, and the signal level sent to the record head.

As Jack Mullin put it, all the "old bugaboos" of S/N, headroom, wow/flutter, distortion and many others are absent with digital. This is especially important in multi-track mixdown, because even with the best analog systems, there is a small loss every time a track is dubbed. Digital systems don't copy — they re-create an original signal according to the ideal specifications

The claim has been made that there have been three epochal milestones since Edison and Berliner began the recording phenomenon a century ago: the plating/pressing technology of making copies (1877), the shift from acoustic to electronic mastering (1925) and the practical utilization of digital technology (1977). It looks as though these claims may be correct.

TOWARD MICROPHONE TRANSPARANCY

by Paul C. Buff

The problem is this: How to amplify the microphone with absolute transparancy?

The term transparancy indicates many things, or to be correct about it, the absence of many things. Like noise, distortion, saturation, phase errors, frequency discrimination and other electronic or magnetic effects which color the sound picture offered by the mike.

To satisfy the requirements for professional use, a microphone amplifier must also maintain a balanced line structure, have excellent common mode rejection and be able to accommodate signals ranging from a whisper, to the line levels produced by an over zealous drummer miked at 2".

Last but not least, the amplifier must accept the mike at its own impedance, and, ideally, should be invisible to the mike, that is to say, the amplifier should present an input impedance which is very high and which is uniform throughout the frequency spectrum.

TRANSFORMERS

We have been indoctrinated to turn to the transformer as a solution to most of these requirements because, after all, it does a pretty good job. Pretty good, that's the key word. Let's see how it really stacks up against today's desired level of performance, let alone tomorrow's.

The first job, in fact the only job a mike transformer does is to *transfer* the electrical *power* of the microphone from one realm of impedance and ground structure to another. I emphasize the words "transfer power" to dispel the ugly rumor that transformers exhibit some magical ability to give "free" (noiseless) gain. Some "engineers" even give them credit for producing signal-to-noise ratios higher than that produced at the microphone itself!

Transformers can, and do, give *voltage* gain (with the accompanying current loss), but *cannot*, and do not amplify. In all cases, passing a signal through a transformer results in a *loss of power*.

That good stuff in the mike is not voltage, it is *power*. A very minute amount of power that hovers a few scant decibels above the unescapable floor of thermal noise. What is thermal noise? It is the effect of the earth's warmth agitating the electrons in any electrical conductor. The

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the quality defies comparison...
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From now on you don't have to pay more money to get Studer quality. The new Studer A80/RC two-channel recorder costs the same as or less than two of the other three popular names.

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absolute maximum signal-to-noise ratio is in the mike element itself, and is limited by the thermal noise produced within the same element. We cannot improve upon this ratio, we can only hope to preserve it. If any of the signal is lost on the way to the amplifier, the S/N ratio is irrevokably degraded and cannot be recovered.

A TYPICAL MIKE TRANSFORMER HAS AN INSERTION LOSS OF AROUND 2 dB

Reduce your optimum signal-to-noise ratio by this amount.

Since a transformer has, in addition to the desired inductance effects, stray capacitance, it must employ terminating resistors on its windings in order to dampen resonant effects produced by the resultant L/C circuits. These resistors, as well as certain core losses and inductive and capacitive reactances appear as a power absorbing shunt circuit across the microphone element. An effective 1,200 ohm termination across a 150 ohm mike serves to absorb 11% of the potential power. Scratch another 1/2 dB signal-to-noise ratio.

So far, we've lost right about 45% of the potential signal power, and we're just getting to the amplifier. Let's see what else has happened along the way.

A TYPICAL MIKE TRANSFORMER PRODUCES FREQUENCY AND PHASE ERRORS

Those stray L/C circuits have, of course, introduced a certain amount of deviation from an ideal phase and frequency curve, as well as causing some degree of ringing and overshoot. They have also caused variations in the loading presented to the mike, thereby disturbing its frequency response.

A paradox exists in practical transformer design wherein an attempt to achieve the high inductance needed for the lower frequencies is always foiled by a corresponding deterioration in high frequency performance, due to increased capacitance. Consequently, the low end suffers the effects of insufficient inductance, which are manifested in phase errors, frequency rolloffs and, due to core effects, a low end frequency response which changes with the applied signal level.

A TYPICAL MIKE TRANSFORMER EXHIBITS IM AND THD DISTORTIONS

In order to minimize losses at low signal levels, the core is made of a magnetically highly sensitive material. A transformer core is a highly non-linear device. As the signal level gets higher, the core characteristics change, and eventually the core saturates. Core saturation inherently occurs earliest at the low frequencies (which just happen to be the ones usually containing the most power in music). It is not uncommon to find unacceptable amounts of IM distortion from signal sources which aren't really that loud, but which contain strong low frequency components. (Ever notice your cymbals buzzing with each bass drum lick?)

MIKE TRANSFORMERS HUM

A transformer is basically a coil of wire around a super-sensitive magnetic core. All the ingredients needed for a magnetic field detector. Mount the device in a nice strong field, such as a control room, and you've got hum. Special winding techniques and triple mu-metal shields help, but there's still a measurable hum component left.

MIKE TRANSFORMERS EXHIBIT RESTRICTED COMMON MODE REJECTION

In a balanced system, unwanted noise signals are induced into the cable in such a manner as to be of equal magnitude and polarity on both signal lines. If this equality can be preserved, the noise signals may be cancelled. The degree of cancellation is commonly referred to as Common Mode Rejection Ratio, or CMRR.

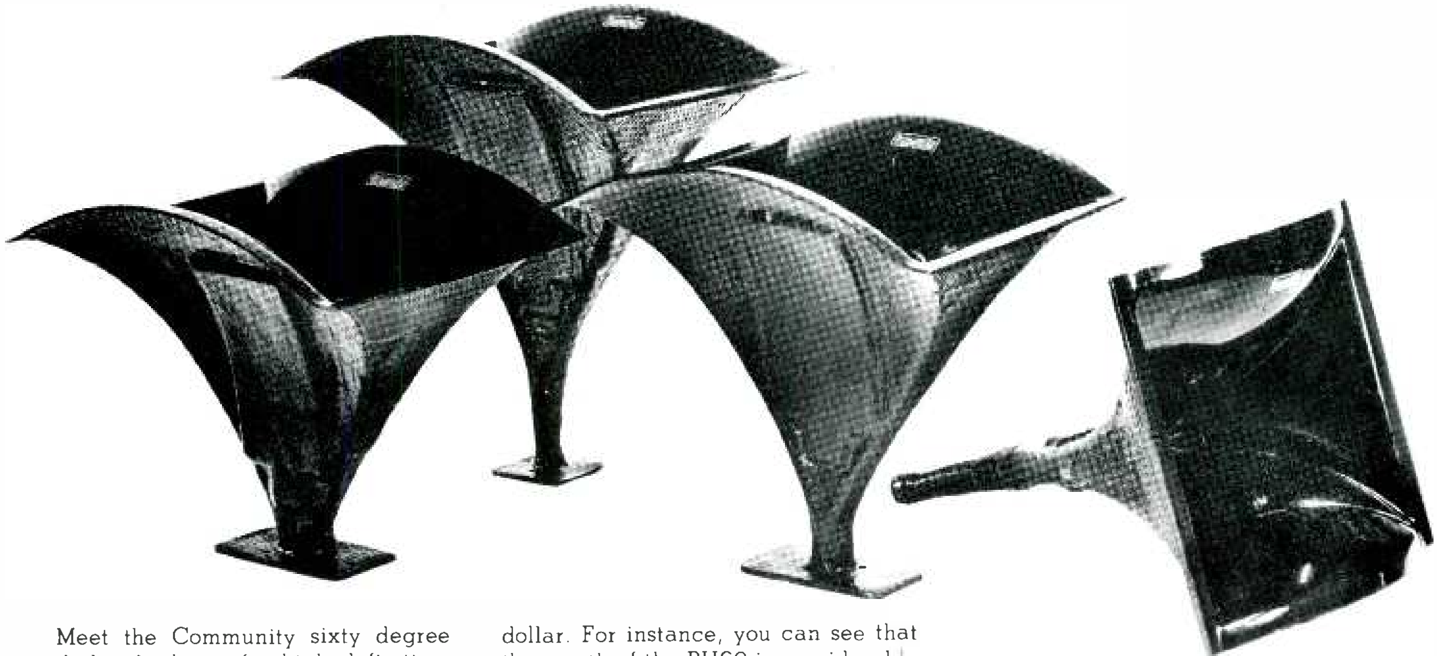
Theoretically, a transformer can completely reject common mode signals. In practice, however, the CMRR of most mike transformers is reduced to around 60 dB to 85 dB, and is frequency dependent, due to inequalities in the winding orientation and to stray capacitances.

It's pretty clear to see that the good old transformer is not really up to the caliber of performance desired by today's pro-audioite.

I.C. MIKE PRE-AMPS

Now that we've examined the mike transformer, let's couple it to a monolithic op-amp representative of types

RH60 RADIAL HORNS



Meet the Community sixty degree radials, *the* horns for high definition, understandable sound. The horn pictured is our RH60, the midrange mainstay of the large system. We've recently added two new sixty degree horns to our line for HF and VHF projection, the SRH60 and the SQ60.

Some people still think that our horns and cabinets look a little strange, perhaps not realizing that at Community shape and construction are determined by the laws of physics, not marketing, packaging or the almighty dollar. For instance, you can see that the mouth of the RH60 is considerably taller than that of comparable sixty degree horns. Why? Well, if a horn is to act as a wave guide at its lower operative frequencies (which it is) it must have a tall mouth to support the larger wave forms generated near crossover. The idea of a thin, wide-mouthed radial may be pleasing in terms of packaging and handling, but it is a convenience that does not pay off in operation. Some conveniences that do pay off in operation are one-piece construction, low resonance, high strength-to-weight ratios and the meticulously executed design that characterizes a Community horn.

Would you like more information? We recently published a catalog which details the performance of all Community products. Already it has been called a must for anyone wishing to design a sound system on a professional level. Please write or call to order.

RESPONSE AND SPL

RH60-A DRIVER: **GAUSS HF4000** BANDWIDTH PINKNOISE: **250Hz-16KHz**

1 Watt @ 1 Meter **107.24** dB-SPL 1 Watt @ 4 Feet **107.28** dB-SPL

Hz	40	50	63	80	100	125	160	200	250	315	400	500	630	800	1K	1.25	1.6	2.0	2.5	3.15	4.0	5.0	6.3	8.0	10.0	12.5	16.0 KHz
									−12	−7	−6	−4	−3	−1	0	+1	−1	−2	−1	−3	−5	−5	−9	−9	−9	−13	−17

SRH60-B DRIVER: **ALTEC 288.16G** BANDWIDTH PINKNOISE: **350HZ-16Hz**

1 Watt @ 1 Meter **108.99** db-SPL 1 Watt @ 4 Feet **105.52** dB-SPL

Hz	40	50	63	80	100	125	160	200	250	315	400	500	630	800	1K	1.25	1.6	2.0	2.5	3.15	4.0	5.0	6.3	8.0	10.0	12.5	16.0 KHz
											−10	−6	−4	−2	0	0	0	0	0	−2	−2	−4	−8	−8	−6	−10	−12

SQ60-C DRIVER: **EMILAR EA. 175.16** BANDWIDTH PINKNOISE: **800Hz-16KHz**

1 Watt @ 1 Meter **103.85** dB-SPL 1 Watt @ 4 Feet **102.14** dB-SPL

Hz	40	50	63	80	100	125	160	200	250	315	400	500	630	800	1K	1.25	1.6	2.0	2.5	3.15	4.0	5.0	6.3	8.0	10.0	12.5	16.0 KHz
														−6	−2	−2	0	0	+2	0	0	0	−2	−2	−5	−12	−16

Community

Community Light & Sound, Inc. 5701 Grays Avenue Philadelphia, PA 19143 (215) 727-0900

used in many of today's consoles.[1]

In order to evaluate the performance, I will conduct a complete and accurate analysis, based upon the op-amps published specifications.[1] For this analysis, I will assume that the gain control is set for the maximum gain (40 dB) (R5 = 300 ohms). It is at this gain that, while the circuit produces the highest output noise, it provides its minimum NOISE FIGURE.

NOISE FIGURE, or NF, is, in simple terms the amount by which an amplifier degrades the signal-to-noise ratio theoretically possible at its input. A NF of 0 dB indicates a completely noiseless amplifier.

NOISE ANALYSIS OF FIGURE 1

There are a total of 6 sources of noise produced in the circuit, excluding any additional sources such as power supply hum or cable induced noise. The 6 sources are uncorrelated sources of essentially "white noise", and as such may be combined on the basis of additive power as implied by the formula:

$$E_t^2 = E_1^2 + E_2^2 + E_n^2$$

The 6 noise sources mentioned are:

E_1 = The thermal noise of the source resistance (point A). (The theoretical minimum noise of the circuit.)

E_2 = The thermal noise of the resistors at point B.

E_3 = The op-amps characteristic noise voltage at point A.

E_4 = The op-amps characteristic noise voltage at point B.

E_5 = The noise voltage produced by the op-amps characteristic noise current flowing through the impedance at point A.

E_6 = The noise voltage produced by the op-amps characteristic noise current flowing through the resistance at point B.

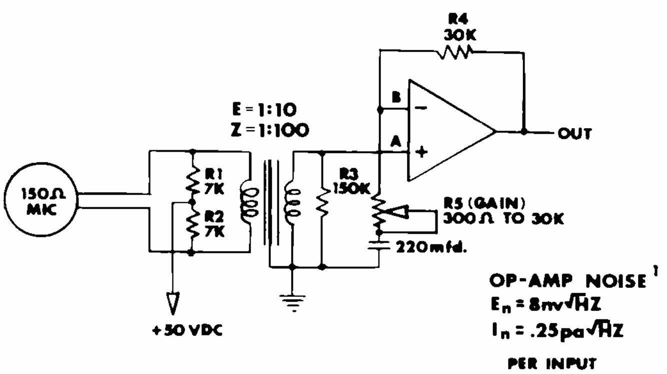


Figure 1: **Typical Transformer Coupled Microphone Pre-amplifier.**

All of the noise sources are, by nature, essentially white noise. The calculations will be done using a 20 Hz. to 20 kHz. bandwidth.

Noise source E_1:

The formula for determining thermal noise is E_t^2 = 4KTRB, wherein:

K = 1.38 x 10^{-23} (Boltzmans Constant)

T = Absolute temperature in Kelvin

R = Resistance in ohms

B = Absolute bandwidth in Hertz

For the circuit of Figure 1, using a typical mike transformer[2], the total effective source resistance, including the reflected impedance of the microphone, the transformer resistance and the shunt dampening resistance, equals approximately 17 Kohms. (The exact impedance is not critical to the calculation.) Assuming a room temperature of 300K (also not critical), and a bandwidth of 20 kHz., the theoretical minimum (thermal noise) voltage $E_1 = 10^{-6}$ vrms.

2.373 x 10^{-6} volts at 17Kohms = 110.27 dBv,
or −124.79 dBme.

Noise source E_2:

Applying the same formula to the parallel combination of R4 and R5 (at pot position R5 = 300 ohms), the thermal noise E_2 = .315 x 10^{-6}vrms.

Noise source E_3:

At the stated 8nv $\sqrt{Hz}$, E_3 = 8nv x $\sqrt{20{,}000}$ = 1.131 x 10^{-6} vrms.

Noise source E_4:

E_4 is equal to E_3 = 1.131 x 10^{-6}vrms.

Noise source E_5:

E_5 is derived by the formula E=IR, wherein I = .25 x 10^{-12} x $\sqrt{20{,}000}$ and R = 17 Kohms. E_5 then = .601 x 10^{-6}vrms.

Noise source E_6:

E_6 is equal to .25 x 10^{-12} x $\sqrt{20{,}000}$ x 300 ohms, or E_6 = 75 x 10^{-12}vrms.

Applying the formula $E_{total}^2 = E_1^2 + E_2^2 + E_3^2 + E_4^2 + E_5^2 + E_6^2$, the total noise present at the amplifier input = 2.941 X 10^{-6}vrms, or −108.41 dBv, or −122.93 dBme.

Since the theoretical minimum noise (thermal noise of the source) is −124.79 dBme, while the actual circuit input noise is −122.93 dBme, the circuit has added 1.86 dB of noise to the signal, and thus has a NOISE FIGURE of 1.86 dB.

ADDING IT UP

If we take the 1.86 dB S/N lost in the amplifier and add the 2.5 dB or so loss in the transformer, we have lost a total of over 4 dB S/N ratio over what is available from the microphone under ideal conditions. Since the accurately stated Theoretical Minimum Noise for a 20 kHz. bandwidth is −124.79 dBm, the Equivalent Input Noise for the transformer coupled mike pre-amp just described is about −120.4 dBme and it may theoretically be improved by 4.4 dB.

Isn't it ironic that the published E.I.N. of many consoles is −129 dBm? Then, of course, we have one manufacturer who claims his equipment is 3½ times quieter than even these figures. I hope that my companion article on interpreting noise specs (in this issue) will shed some light on this subject.

ALTERNATIVES

Obviously, the first order of business is to eliminate the input transformer, and couple the mike directly to the amplifier. This has been suggested and tried in the past, without impressive results. Let's analyze the classic approaches for doing this, and find out why they won't work.

In each of the following examples we will assume the op-

[1] Harris HA911 op-amp

[2] Jensen Transformer

amp used is a high performance discrete transistor device exhibiting a low noise voltage of 3nv $\sqrt{Hz}$., a noise current of .5pa $\sqrt{Hz}$. and having a unity gain bandwidth of 10 mHz.

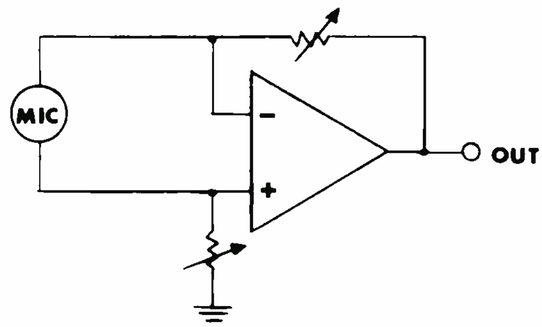


Figure 2: **Single Op-amp Direct Coupled Microphone Pre-amplifier.**

Although the circuit of Figure 2 is theoretically capable of good CMRR, the two input signal lines see unequal impedances. When the mike cable is subjected to external noise fields, the current induced is not common mode (not of equal magnitude), and thus will not be rejected with any precision.

Due to the relatively high noise voltage of even a good discrete op-amp, a poor noise figure of 9.15 dB is obtained.

In order to provide a true balanced input impedance, and the resultant high CMRR, the circuit of Figure 3 has enjoyed wide use in precision instrumentation applications.

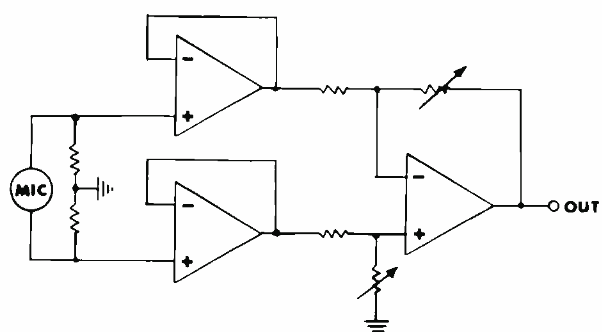


Figure 3: **Instrumentation Amplifier Type Direct Coupled Microphone Pre-amplifier.**

For use as a mike pre-amp, however, the circuit of Figure 3 is dead before you even build it. A total of 8 significant noise sources, plus 6 insignificant ones, yield a total noise figure of around 14 dB.

GAIN, DISTORTION AND FREQUENCY RESPONSE

With either of the above circuits, due to the 10 mHz. bandwidth of the op-amps, the amount of 20 kHz. feedback at 40 dB of gain would come to only 14 dB, thereby making 40 dB about the maximum circuit gain for acceptable high frequency performance. Since 55 dB to 60 dB of gain is required for professional use, additional amplification would be necessary. In order to cover an adjustable gain range of, say +10 dB to +60 dB, a multi-gang switching scheme must be used, since the ratios of at least 3 sets of resistors must be switched to cover this range. It is also important to note that in both of the examples, the resistor ratios must be held in exact balance (.001%) in order to maintain a CMRR of 100 dB. It is then impractical to vary these elements.

A FRESH APPROACH

Using previously known techniques in attempting to eliminate mike input transformers is obviously not going to work. The designer must explore new concepts using, of course, those components which are currently available at reasonable cost.

Suppose we were to pose the following question to the top experts in the field of applied pro-audio: "What would it take, in numbers, to realize a transparant microphone amplifier. That is to say, one whose audible performance would not materially differ from that of a 'wire with gain'."

I believe that after we canned the ones that said, "Response from D.C. to light, no distortion, no noise," the answer would come close to this:

1. "The device must work with professional 150 ohm balanced mikes."
2. "The device must not alter the frequency response, phase characteristics or other timbre related microphone parameters."
3. "The device should be able to accept, without padding, the output of any professional mike, under any condition of studio use."
4. "Related to #2 and #3 above, the device should introduce less than .01% distortion, IM or THD, even at line input levels."
5. "The device should add no more than 1 dB S/N beyond what is potentially available at the mike itself, including losses from loading."
6. "The device should exhibit a frequency range and full power bandwidth beyond 100 kHz, and must not be frequency discriminatory in its mike loading pattern."
7. "The device should exhibit a CMRR in excess of 100 dB, and must maintain a percise equalibrium of balance at its input in order to discourage non-common mode noise induction."
8. "The device should have an adjustable gain range of from under 10 dB to over 60 dB, and must conform to all other requirements while operating within this gain range."

With the requirements thus known, we set about to attempt to meet or exceed them. Our approach was, of necessity, one of scrutinizing the data books for semiconductor devices whose parameters, when configured in previously untried ways, might fill the bill. An intense period of theorizing, prototyping, testing, recalibrating the test equipment and re-testing and, yes indeed, listening to microphones ultimately resulted in the device to be described.

No magic or specmanship is involved. The device does not go "beyond the theoretical limits" (although it comes mighty close in some areas). The measured performance coincides precisely with what theory says the device should do.

INTRODUCING THE TRANS-AMP™

Specifically, the Valley People, Inc., Trans-Amp™ is a differential in/differential out amplifier optimized for extremely low noise and distortion when employed in a balanced format at very low impedances.

Unlike the familiar op-amp configuration, the Trans-Amp™ employs a pair of symetrically opposed feedback loops, which return to internal circuit ports, rather than to the signal input ports. This feedback geometry leaves the

Who makes the ideal real-time acoustics analyzer?

Characteristics of the Ideal Analyzer	B&K Model 2131	GenRad Model 1921	Nicolet Model 444-163
1. Both 1/3 and 1/1 oct. capability	**YES** (but requires rerun of data)	**NO** (⅓ oct. only)	**YES** (from **one** run of data)
2. Built-in narrowband analysis to localize noise sources, detect sharp "notches" and "peaks"	**NO**	**NO**	**YES** (400-element analysis included in same unit with dual averager memories)
3. Wide frequency coverage	42 ⅓-octaves 1.6 - 20 kHz (opt. to 160 kHz)	30 ⅓-octaves (typ. 25 Hz - 20 kHz **or** other 10 octaves from 1.6 Hz to 80 kHz	42 ⅓-octaves **in 5 ranges** from **6.3 Hz - 80 kHz** (opt. L: 8 ranges from .63 Hz)
4. Filter shapes meet international standards	ANSI Class III (1/3) and Class II (1/1)	Same	Same
5. Compares, equalizes, or calculates transmission loss from 2 stored spectra	**NO** (two spectra held but only visual comparison)	**NO** (stores only one spectrum)	**YES** (stores two ⅓ oct. **and** two 400-line spectra, and displays **differences** or **ratios**)
6. Mass storage and recall of past spectra, equalization curves, and test standards	**NO**	**NO**	**YES** (including difference and ratio comparisons with current spectra using Model 144 Digital Tape Recorder)
7. Capability of interfacing with computer/calculator to convert to Sones, Loudness, perceived noise, etc.	**YES**	**YES**	**YES** (simple, inexpensive calculator interface option)
8. Single, portable unit of minimum weight	one rack-mounted unit of 64 lbs.	3 separate units of 113 lbs.	one portable unit of 45 lbs.
9. Average non-stationary data in exponential or peak hold modes (as well as RMS linear mode)	**YES**	**NO**	**YES**
10. Automatic capture, display and narrowband analysis of transients	**NO**	**NO**	**YES**

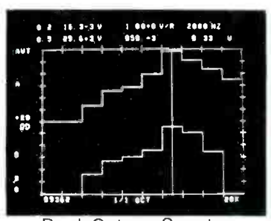
Data on competitive equipment is based on the latest available published specifications.

A glance at the chart will show you. It's not B & K. It's not GenRad. It's Nicolet.

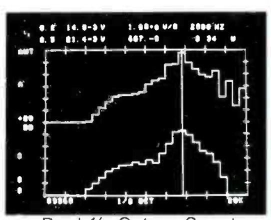
The Nicolet 444-163 all-digital FFT ⅓ octave dual-memory analyzer is the most versatile, the only one with **all** the characteristics the ideal real-time acoustics analyzer must have. Why these characteristics? Because an acoustics analyzer has to **solve problems** like these:

• What is the **sound transmission loss** of this material? • How does my product's **"free field" noise** spectra differ from spec? • What mechanisms in my product are the **major sources of noise?** • **Which machine** (or part of a machine) in my factory needs quieting? • Is the **response and distortion** of my amplifier, speaker, etc., in spec? • What is the maximum **passby noise** of my vehicle, aircraft, etc.? • What is my product noise in terms of **SONES, LOUDNESS** or **PERCEIVED NOISE?**

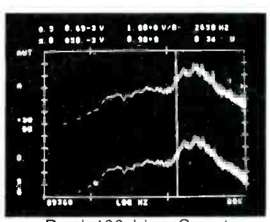
The characteristics we have chosen for the chart are the ones which will help you solve these problems faster, easier, and more completely.



Dual Octave Spectra.



Dual ⅓-Octave Spectra.



Dual 400-Line Spectra.

Nicolet Model 444-163 with "digitally calculated" precise A-weighting.

The final characteristic necessary for the ideal analyzer is reasonable price. The NICOLET Analyzer has that too. Call Joe Deery, Sales Manager, for a demonstration at **your** facility on **your** data.

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for additional information circle no. 46

signal input ports configured as a high impedance balanced bridging input.

A proprietary arrangement of input transistors results in two important situations which affect the noise performance of the device, as follows:

1. The total noise voltage of each input port is reduced to .05nv Hz. (about 1/6 that of typical discrete transistor op-amps, or about 1/20 that of typical monolithic op-amps.)

2. The configuration effectively cancels most of the noise contributed by the transistor noise currents, when fed from a floating input source. (Such as a microphone.)

In terms of tangible results, the total noise figure of the Trans-Amp™ in a typical transformerless mike pre-amp

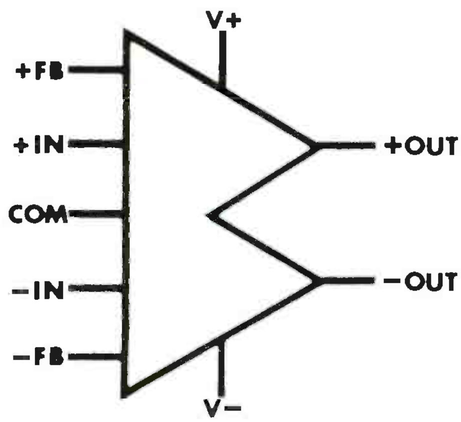

Figure 4: **Symbolic Representation of the TRANS-AMP™.**

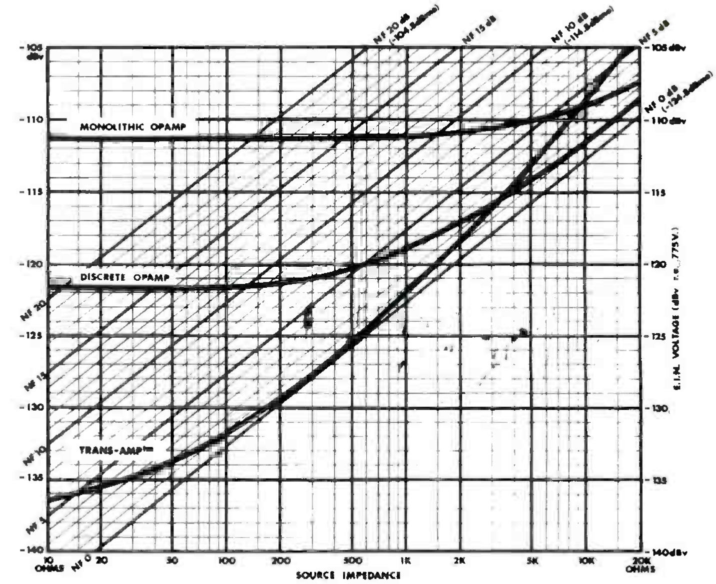

Figure 5: **Graph Depicting Equivalent Input Noise of the TRANS-AMP™, vs. Typical Monolithic and Discrete Transistor Op-amps.** (Broadband Noise from 20 Hz to 20 kHz). **Graph Indicates Total Input Noise In Typical Operating Circuits, in terms of dBv, dBme and Noise Figure (NF).**

configuration is .5 dB, when fed from a 150 ohm source. (See Figure 5.) This means that the signal-to-noise ratio is within ½ dB of what is theoretically possible, with any amplifier. Of further interest is that the Noise Figure remains under 1.8 dB for microphones as low as 50 ohms, or as high as 3.5 Kohms.

GAIN BANDWIDTH

A second major attribute of the Trans-Amp™ configuration is its ability to alter its effective open loop

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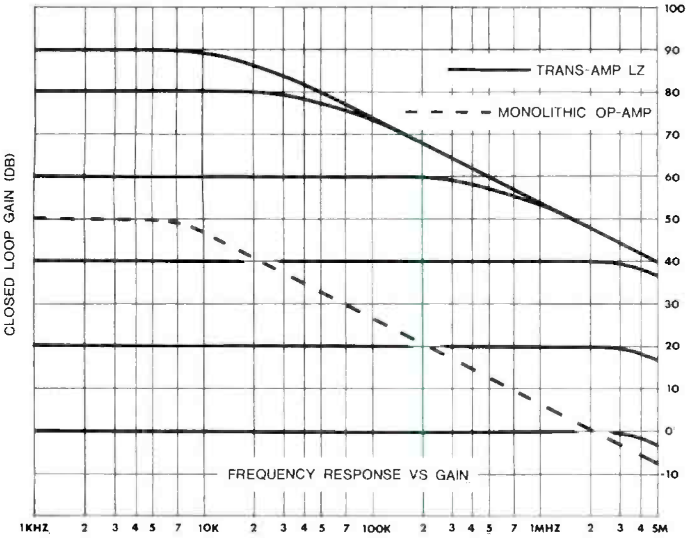


Figure 6: **TRANS-AMP™ Bandwidth Characteristics vs. Typical Op-amp.**

bandwidth as required by the selected closed loop circuit gain. As the closed loop gain is adjusted upward from unity gain, the open loop bandwidth changes from an initial 5 mHz. point, upward to a maximum of .5 gHz (500 mHz.) This action allows stability over an extremely wide range of closed loop gains and results in unusually high bandwidths at high gain settings. (.5 mHz. @ 60 dB gain — See Figure 6.)

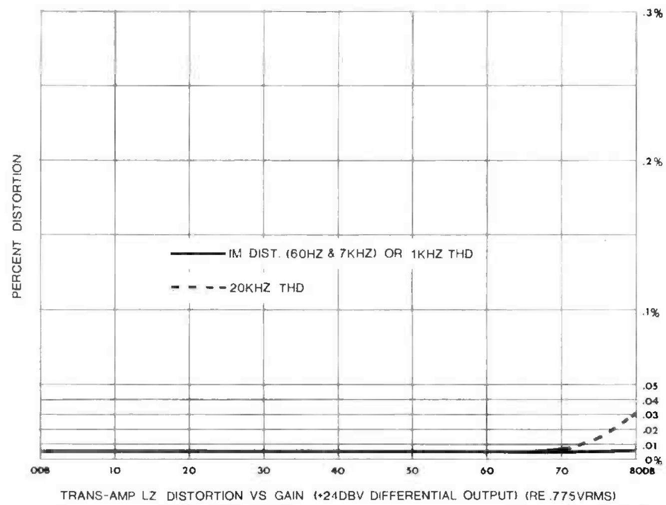
The variable open loop bandwidth phenominon is also an important factor in meeting the requirements for minimum T.I.M. distortion.

GAIN RANGE/INPUT OVERLOAD/DISTORTION

When used in the suggested mike pre-amp circuit of Figure 8, the gain is continuously variable from 6 dB to 70 dB, using a single potentiometer. When set for the minimum gain of 6 dB, a level of +21 dBv may be applied directly to the input without saturation effects, or increased distortion. It is therefore unnecessary to employ pads, as input overload from microphone sources is impossible.

At the maximum gain of 70 dB, no deterioration of bandwidth or distortion parameters is evident. (At 70 dB gain the bandwidth is in excess of 150 kHz.) Throughout the variable gain range the distortion products, either

Figure 7: **TRANS-AMP™ Distortion Characteristics.**



THD or IM, remain below .01% (typically .005%), including 20 kHz. THD.

The Trans-Amp™ exhibits a differential output capability of +27 dBv, with full power bandwidth exceeding 150 kHz. (Differential slew rate of 26v/usec.)

INPUT IMPEDANCE/BALANCE/CMRR

The Trans-Amp™ presents a precisely balanced 100 Kohm differential input impedance to the microphone, while presenting a 25 Kohm impedance to ground for common mode signals. This results in essentially a zero loading factor so that the full potential output of the mike may be realized, both in terms of signal-to-noise ratio as well as frequency response. The configuration also insures that any external noise induced is coherent on both signal lines, and therefore rejectable. The Trans-Amp™ can be trimmed to produce a CMRR well in excess of 100 dB (typically 120 dB).

LOW FREQUENCY CONSIDERATION

The Trans-Amp™ is internally A.C. coupled, thus eliminating the need for external coupling or feedback capacitors, except in phantom powered systems. The effective low frequency unity gain bandwidth is .0003 Hz., allowing response down to 1 Hz. at 70 dB of gain.

PACKAGING/EXTERNAL CIRCUITRY

The Trans-Amp™ is packaged in a 1¼″ x 1¼″ x .550″ epoxy module. It requires bipolar 15 volts @ 8 ma. In most applications, the differential outputs of the Trans-Amp™ must be combined into a conventional op-amp, in order to produce a single ended output and to maximize the CMRR. Although the choice of coupling op-amps will affect the maximum output voltage, slew rates and frequency range, they will have little or no effect on noise characteristics.

An excellent, and inexpensive, choice would be the Signetics NE5534 series, while the Texas Instruments TL-071 series (under $1.00) will suffice in many applications.

OTHER TRANS-AMP™ APPLICATIONS

While the bulk of this text has been devoted to mike pre-amp uses, it is obvious that the Trans-Amp™ has powerful applications in other areas of amplification. One of these is shown schematically in Figure 9. As can be seen in Figure 9, the Trans-Amp™ can be converted to a differential input current summing amplifier, by the simple expedient of grounding the signal input ports and feeding the signal into the feedback ports. In this configuration, all parameters of noise, gain, bandwidth and slew rate will remain the same

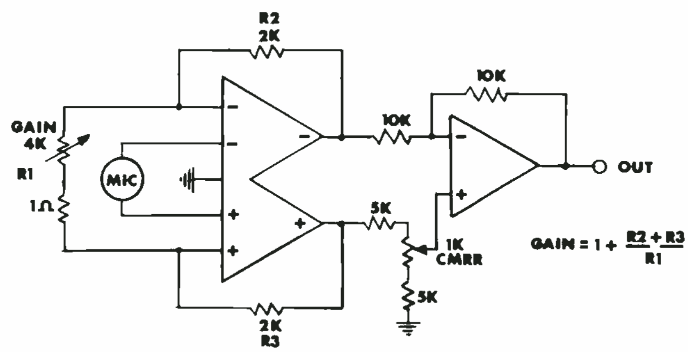


Figure 8: **TRANS-AMP™ In Typical Configuration as a Direct Coupled Microphone Pre-amplifier.**

for additional information circle no. 49

as described for mike pre-amp use. The input impedance will now, however, revert to a virtual ground current summing point. In typical op-amp active combining networks, summing, say 40 inputs @ 10 Kohms, the effective source resistance to the op-amp is in the area of 250 ohms, much too low to match the optimum noise impedance of the op-amp. (Typically around 10 Kohms to 50 Kohms.) The result, of course, is a significant decrease in S/N ratio over what is possible. A second fault lies in the fact that as more and more inputs are combined, the bandwidth of the ACN changes dramatically, while high frequency distortion components increase, due to decreased feedback.

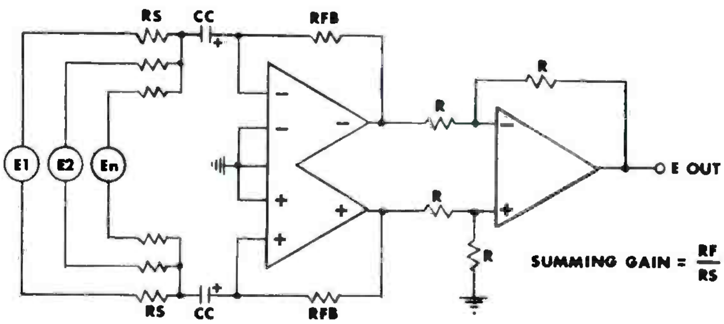


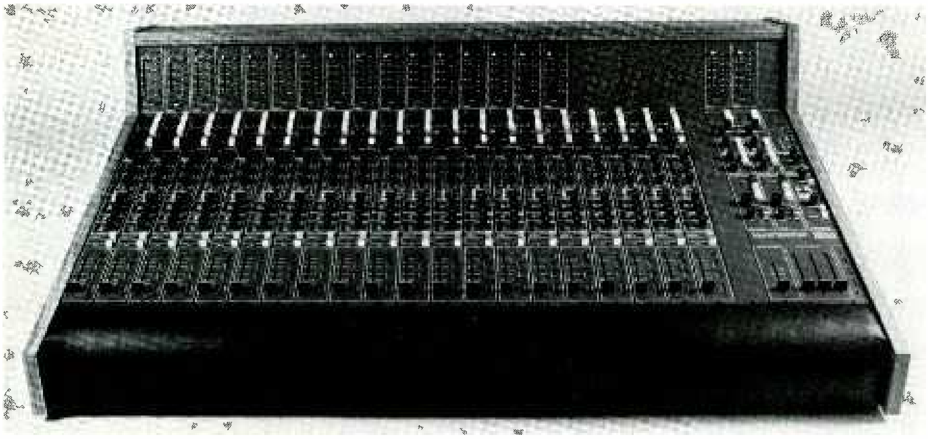
Figure 9: **TRANS-AMP™ In Typical Configuration as a Balanced Differential Active Combining Amplifier. (Current Summing Amplifier)**

In ACN service, the Trans-Amp™, besides offering the noise rejection qualities gained by differential summing, can offer up to 20 dB less buss noise. Because of its gain proportional bandwidth, no change in bandwidth (5 mHz.) or distortion will occur as the number of inputs is increased from one to one hundred.

The Trans-Amp™ is also quite capable, due to its Noise Figure, of combining inputs at gains higher than unity, thereby easing the output requirements of the driving sources.

More obvious applications for the device may be found in tape head amplification, phono cartridge pre-amps, line input stages and a host of instrumentation, medical, geophysical and aerospace applications.

Valley People, Inc., offers an evaluation kit consisting of the Trans-Amp™, together with all required external circuitry required to configure the circuits described in this text, for $35.00. Delivery of sample quantities is two weeks, production quantities available in 4 to 6 weeks, ARO. Pricing in O.E.M. quantities is under $20.00.



SOUND WORKSHOP INTRODUCES MODULAR AUTOMATED CONSOLE

Calling it a new philosophy of console design, Sound Workshop has announced its new Series 1600 Automated Mixing Console. Featuring a fully modular mainframe, the Series 1600 may be purchased in configurations from 12 x 8 up to 36 x 32. All configurations may be expanded up to full capability by adding mainframe expander sections.

The console is available with the complete automation package installed, or the automation may be added at any time. The automation retro-fit is accomplished in 2 steps. The first is the addition of VCA Automation Control Cards to each input module, permitting VCA Input Sub-grouping. The second step is the addition of the Sound Wörkshop Automation Processor, allowing full level and mute automation. Sound Workshop's automation system is compatible with MCI's current automation system.

The console is not designed around the standard I/O module, but rather has a separate input module and output module which interfaces electrically and mechanically to form one unit. In addition, both the equalizer and the send assign matrix are separate internal subassemblies, allowing ease of service, as well as a choice of equalizers. Two equalizers are presently available; one is a 3 band, peak/dip type with 4 frequencies per band; the other is a 3 band parametric with a 20:1 frequency sweep and 4 "Q" positions per band.

Console interface is simplified by the unique design of the modular patch bay. All jacks associated with a given Input/Output Channel are mounted on a separate removable PC board. All connections are by heavy-duty Molex connectors. The patch will be available separately for use with all consoles.

The Sound Workshop Series 1600 is fully balanced on all inputs and on all Track outputs. State-of-the-art circuitry is used throughout yielding superior sonic qualities and specifications. Frequency response: +0, –½ dB (20 Hz. - 20 kHz., up to +26 dBm out). Maximum output: +26 dBm. THD (Mike-in to Track-out): less than .07% (+26 dBm, 20 kHz.) typically less than .03%. IM Distortion: less than .02%, typically less than .008%. Mike Equivalent Input Noise: –129 dBm, DC-20 kHz., UNWTD.

The Series 1600 ranges in price from $10,000 to over $60,000 depending on configuration and options. Initial orders are being shipped in February.

**SOUND WORKSHOP
1040 NORTHERN BOULEVARD
ROSLYN, NY 11576
516/621-6710**

for additional information circle no. 52

AMPEX SEARCH TO CUE ACCESSORY INTRODUCED

A new multi-point search-to-cue and tape timing accessory that provides up to 20-cue storage capability for a variety of mixdown and overdubbing operations has been introduced by Ampex Corporation for use with Ampex ATR-100 and MM-1200 audio recorders.

A 10-button keyboard panel and a cue store control allow access to a

for additional information circle no. 51

Modern sound reinforcement is reaching a level of sophistication that demands only the most critically engineered, high performance equipment available. That's why so many professionals are buying Peavey.

The Peavey line of precision sound reinforcement equipment is the result of years of research and development by a group of the industry's most knowledgeable sound engineers, designers, and acoustic consultants.

Creating professional, state-of-the-art products at reasonable prices is what Peavey is famous for. Musicians the world over have come to expect value and performance from equipment bearing the Peavey name. We're proud of that reputation. Our new line of sound reinforcement gear reflects that pride.

See your Peavey Dealer. He'll show you why the Pros are buying Peavey's value and performance. We think you'll come to the same conclusions.

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digital memory array that is a key feature. The keyboard is used to store up to 20 cues in memory and to recall them as desired.

Three separate time displays show tape time, memory time, and keyboard entry time. (The time displays are alternately available through the main LED readout.) An additional display shows current cue reference. The main time display automatically returns to tape time upon reaching the desired memory time or cue.

The keyboard includes a separate "0/10" button that shifts cue access from the first to the second group of ten memories. This one-key access to memories 11 through 20 is especially helpful during multiple cueing on-the-fly.

The operator can abort the search-to-cue mode with either the stop control on the recorder or with the tape time control.

Zero reference can be set anywhere on the tape, as well as at the beginning of the first selection. The zero point determines the location of all subsequently introduced cues. A "Dual-Side-of-Zero" search permits access to any of up to 20 previously set cues which may be located before or after the zero reference; this is accomplished at full shuttle speed (fast forward or rewind).

The approach-to-cue is controlled in a proportional trajectory in the search mode, eliminating overshoot and aiding in gentler tape handling by the recorder. An added convenience feature allows the operator to pre-select either "play" or "stop" upon arrival at the recalled cue.

The multi-point search-to-cue is compatible with all Ampex ATR-100 and MM—1200 series recorders. It is priced at less than $2,000. Deliveries will begin in December.

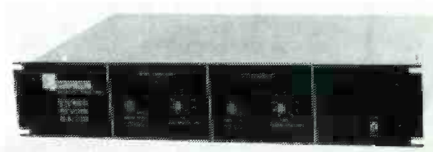
AMPEX CORPORATION
401 BROADWAY
REDWOOD CITY, CA 94063

for additional information circle no. 54

QUANTUM QA-201 STEREO REVERB

The QA-201 is a professional, moderately priced stereo reverberation chamber. It utilizes two Accutronics reverb units which provides two completely independent channels. Each chamber has a volume control and high frequency tone control. A compressor/limiter circuit is included which permits optimum drive to each chamber. A unique overload indicator senses three different points in the circuit to detect possible overload.

The QA-201 accepts input levels



from –20 dBm to +18 dBm. Each input uses a unique balanced circuit with a minimum of 60 dB common mode rejection. Both XLR connectors and 3 circuit phone jack connectors are provided on the rear panel for interface with any balanced or unbalanced external equipment.

QUANTUM AUDIO LABS, INC.
1905 RIVERSIDE DRIVE
GLENDALE, CA 91201
213/841-0970

for additional information circle no. 56

CUSTOM AUDIO ELECTRONICS XPC-16

XPC Series audio mixing consoles from Custom Audio Electronics feature Totally Modular™ construction, i.e. no mainframe; no motherboards.

for additional information circle no. 55

RECORDING engineer/producer

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☐ Producer ☐ Sound Contractor

☐ Other ______________

4) Which of the following describes the functions you regularly perform?

☐ Owner/Manager ☐ Producer
☐ Mixer ☐ Disc Cutter
☐ Maintenance Engineer ☐ Remote Recording
☐ Video/Movie Recording ☐ Sound Reinforcement
☐ Production Quantity Tape Duplicating

☐ Other ______________

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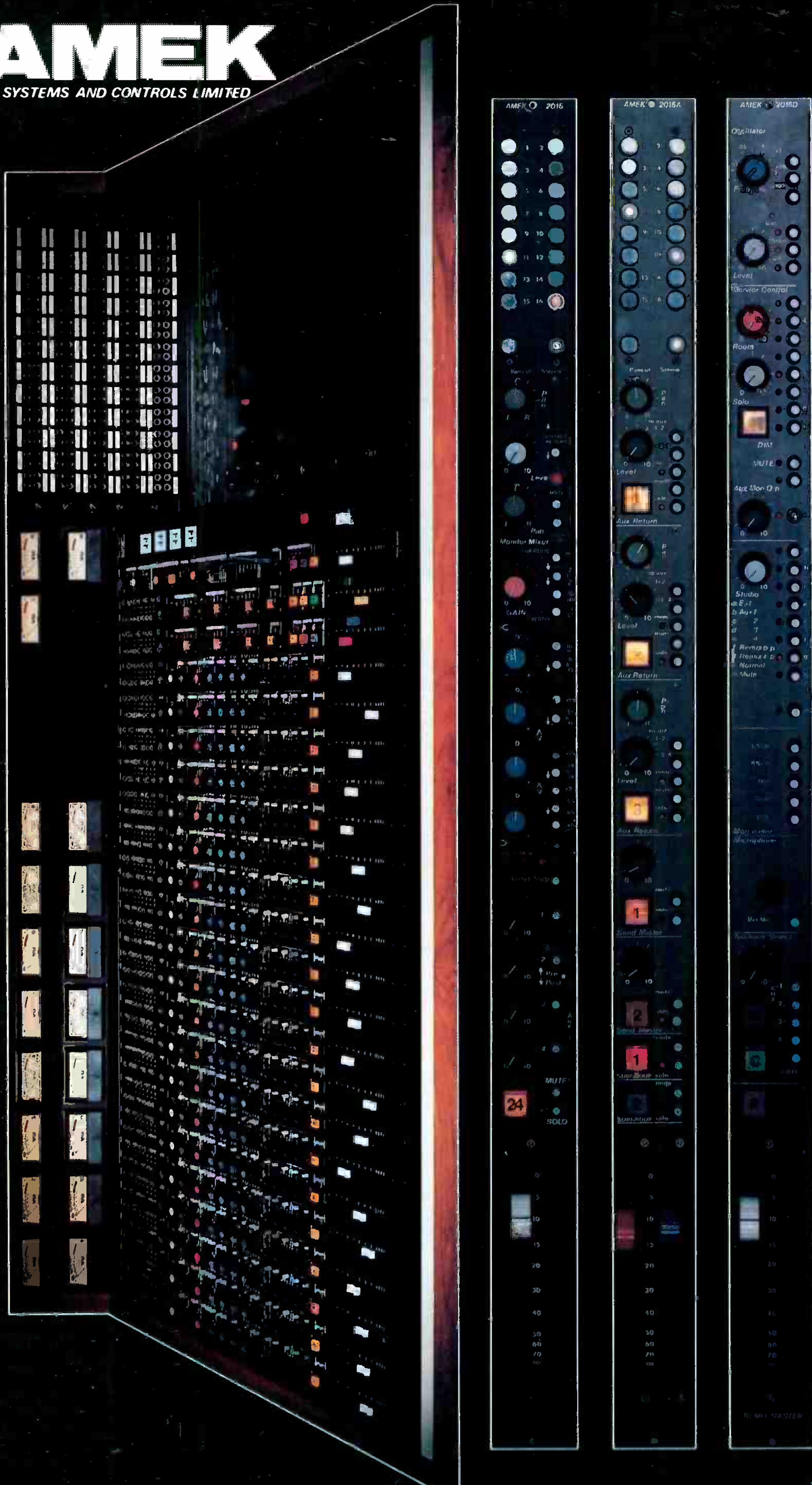
* Please Note: This address is for Reader Service only.
Address other correspondence to P.O. Box 2449, Hollywood, CA 90028.

AMEK

SYSTEMS AND CONTROLS LIMITED



WHERE AMEK HAS BEEN:
Village Way Studio, England
Regents Park Recording Company, England
Berry Street Studios, England
Utopia Studios, England
Slater Sound Studio, Australia
Castle Sound, England
Wembley Conference Center, England

WHERE AMEK IS GOING:
Sierra Pacific Recording Studios, California
Don Costa Productions, California
Music Annex Recording Studios, California
Sun Sound Studios, Connecticut
ABC Records, California
Everything Audio, California

WHAT AMEK HAS:
Main frames in sizes from 20-28 inputs • 16 buss outputs • 19-27 VU meters • 8 direct buss assign • 4 addressable group controls • Penny & Giles faders • 4 auxiliary sends • Stereo solo-in-place in both input and monitor sections • 4 band equalizer with 2 additional filters • One microphone and 2 line positions per input channel • Pan control between live busses • Stereo panning on each module • Stereo panning on monitor section • Phantom power supply • All metal cabinet construction • Frame stand with 19" rack included • 100 position tip, ring, sleeve patch bay with expansion provided • Test oscillator • Talkback • Dim and complete control room and studio control.

WHAT AMEK COSTS:
$17,000 to $25,000

WHAT AMEK SHIPS FROM:
STOCK

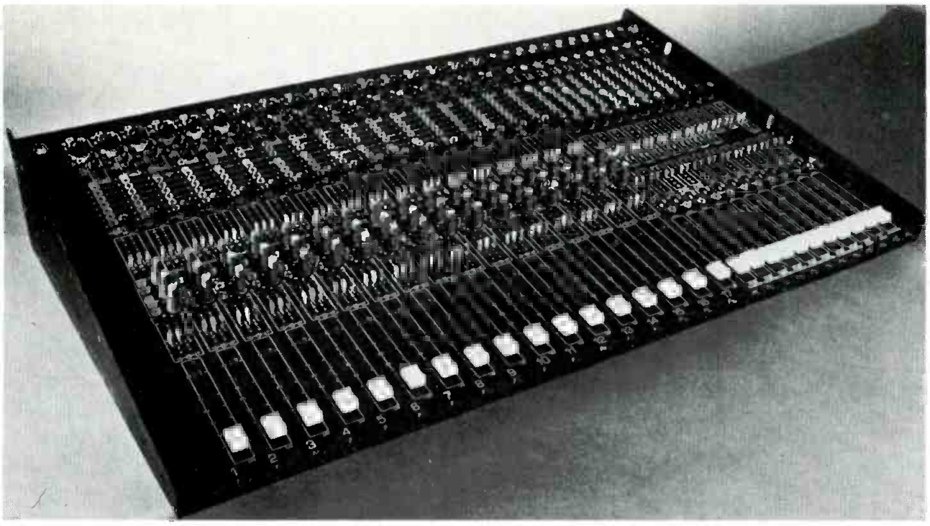
Distributed throughout the United States by

EVERYTHING AUDIO

Accept our invitation to contact us and discuss your studio needs.

7037 Laurel Canyon Blvd., North Hollywood, California 91605 / 982-6200

for additional information circle no. 57



Thoroughly applicable to studio use as well, the XPC Series was originally conceived as a reinforcement mixer for touring. It is ruggedly constructed, built entirely of ⅛" aluminum extrusion.

Standard facilities include full stereo assignable submastering, eight selectable band point equalization, switchable channel breaks, two effects sends (selectable post-pre-amp, post-equalization, post-fader), solo, pan, phase reverse, pad, variable gain, three-light LED meter and integral rear patchbay. Any number up to fifty input modules may be used with a single power supply. Additional modules include submasters, A/B/Solo masters, an eight-mix master module and a communications module.

CUSTOM AUDIO ELECTRONICS
2828 STOMMEL ROAD
YPSILANTI, MI 48197
313/482-6568

for additional information circle no. 58

NEW CONSOLES FROM RUSLANG

Ruslang Corporation recently introduced two new tape transport consoles, the RL600 and RL700. Both are constructed with the same care as the Scully consoles which Ruslang has manufactured for the past 13 years.

The RL600 is designed to accept the new style of tape decks 19" x up to 21" which have their electronics integrated with the transport. The RL700 will accommodate tape transports 19" x up to 24½", such as the Ampex 300 Series. The RL500, introduced last year, handles 19" x 15¾" decks.

All models incorporate the newest design features, including front panel access in both horizontal and vertical positions, plus a rear shelf for power supplies. These compact consoles, constructed of quality materials, are both stylish and sturdy.

Additionally, Ruslang is offering an improved electronics console which can be added on to any Scully 280 tape transport console.

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

for additional information circle no. 60

QSC ELECTRONIC CROSSOVER/POWER AMP COMBINATION

Substantially improved is this unique product combining both an electronic crossover and a high frequency power amp in a single cost-effective package. The new model, designated the Electronic Crossover 1.1, has both new crossover and power amp sections.

The crossover section now has an exclusive QSC circuit with a fully active high-pass filter and constant phase complimentary low-pass deriver. This has a 12 dB per octave slope and a Bessel filter response rather than the common Butterworth curves. The advantage of this circuit is said to be

Engineer as Artist

Sound engineering is as much a part of creative music today as the performance itself, and is changing the scope of the industry. Audio technology is presenting a new range of creativity. It is the audio engineer who applies imagination to this technology and expands the boundaries of creative sound.

The MXR Digital Delay gives the audio engineer a tool for creative application that is unparalleled in versatility, precision and ease of operation. The MXR Digital Delay is designed for a wide variety of applications including; amplified musical instruments, vocals, PA and recording mixes. The basic unit delays a sound between 0.08 milliseconds and 320 milliseconds, fully variable while retaining the dynamic range of the program source. The delay range is expandable to 1280 milliseconds in increments of 320 milliseconds by means of up to three additional plug-in memory boards. These boards are available from MXR and may be installed by the user.

Effects that can be obtained with fixed time delays include echo, vocal doubling and hard reverberation. The MXR Digital Delay contains sweep circuitry which allows additional effects such as flanging, vibrato, pitch bending and frequency modulation. The MXR Digital Delay is also capable of repeat hold (infinite non deteriorating regeneration).

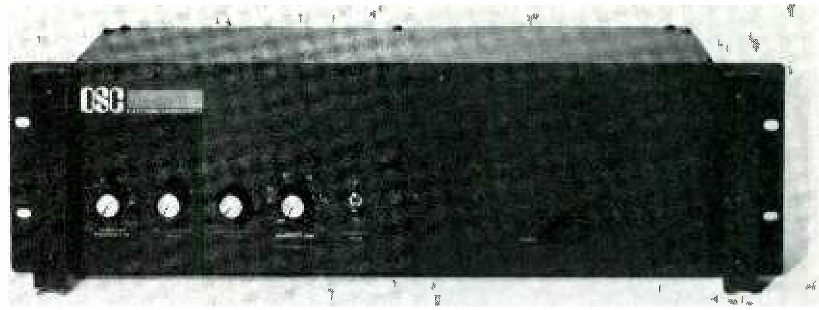
Rack mountable for sound studio installation, it is also available with an optional road case for onstage use or location recording mixes.

MXR's Digital Delay can lead the way to new possibilities in creative sound at a price considerably lower than any comparable delay.

For more information see your MXR dealer.
MXR Innovations, 247 N. Goodman St., Rochester, New York 14607, (716) 442-5320.
Distributed in Canada by Yorkville Sound Ltd., 80 Midwest Road, Scarborough, Ontario.

MXR Professional Products Group

for additional information circle no. 61



optimum phase response and perfect high-low frequency matching without the amplitude and phase errors so common in other types. The result is a clarity that is unmatched by other types of electronic crossovers. Clear and accurate reproduction of the high frequencies has been assured with the new high speed (slew rate of 8v/usec.) power amplifier circuit. Totally redesigned and with a larger power supply the new power amp now delivers 70 watts rms into 4 Ohms with .25% THD and .25% IM distortion. Although intended primarily as a high frequency power amp it can be used full range and has a frequency response of 20 - 20 kHz. ±1 dB.

The controls on the Electronic Crossover 1.1, all fully calibrated, include: Crossover Frequency (250-6K Hz.), High and Low Frequency Gains (out +10 dB), Power Amplifier Gain, and High/Low Phase switch (0 - 180 degrees). The phase switch allows the high frequencies to be reversed, if desired, to improve system performance. There is a complete set of crossover inputs and outputs as well as line level inputs to the power amp that allows it to be used independently. A DC blocking capacitor is installed on one of the outputs to prevent horn driver damage from any accidental low frequency signal. The front panel, finished in black line-grained aluminum, has custom rack handles and a refined multi-color graphic treatment.

The suggested list price is $278.00.

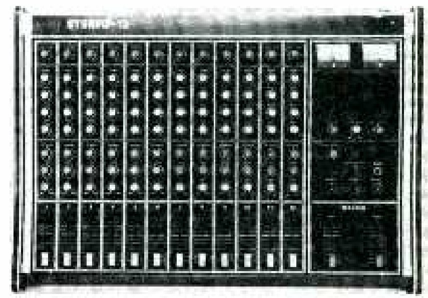
QSC AUDIO PRODUCTS
1936 PLACENTIA AVENUE
COSTA MESA, CA 92627
714/645-2540

for additional information circle no. 63

HH INTRODUCES NEW STEREO MIXER

Audio Marketing, Ltd., HH's exclusive U.S. distributor, is offering a new HH stereo mixer with an optional echo effects module.

The unit's low impedance and balanced inputs are switchable to line level. And continuously variable gain controls match input levels perfectly. The equalizer has four frequency bands while foldback and echo send are separately controllable from each input.



Foldback, echo and all main outputs are fully balanced at +4 dBm into 600 Ohms. Maximum output level is +25 dBm. A switchable headphone monitor and facilities for the unique CCD Echo Effects Module complete this mixer.

AUDIO MARKETING, LTD.
142 HAMILTON AVENUE
STAMFORD, CT 06902
203/359-2312

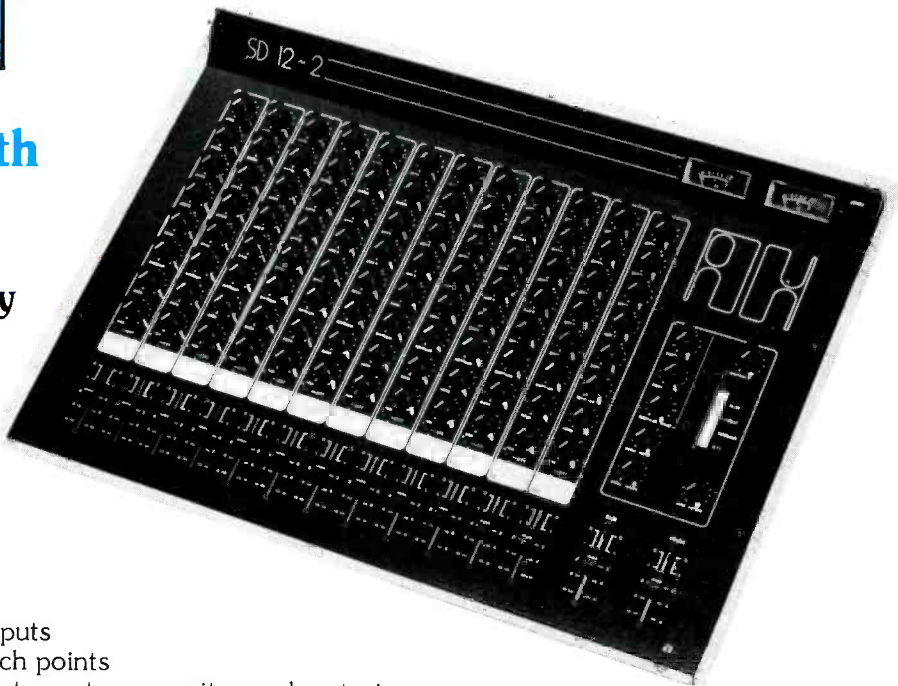
for additional information circle no. 64

AMANITA MOLDED PROTECTIVE ENCLOSURES

Amanita's new line of protective loudspeaker and standard 19" equipment enclosures are designed to withstand the ultimate in heavy road use. Construction is of lightweight, rugged, low-density polyethelene with low resonance characteristics and available in five colors. The unique design of a special shock-mounted speaker panel and 19" rack mount frame protect even the most delicate of components from direct contact with damaging outside impact. Each of the enclosures in the line, including

Allen & Heath SD 12-2

No other mixer delivers so many features for so little money . . .



Pan pots
Input metering
Stereo echo return
Built-in power supply
12 Mic and line inputs
4 band EQ on each input
600 ohm line level on outputs
12 direct outputs and patch points
Headphone monitor with stereo tape monitor and metering
Foldback (stage monitor), echo send, and PFL (solo) on each input

Allen & Heath S6-2

A complete broadcast production console and an incredible disco console . . .

2 stereo RIAA phono inputs with EQ
Stereo main and monitor output
80db signal to noise/.05 THD
Input and output patch points
2 stereo tape inputs with EQ
Automatic voice-over circuit
Gain control on each input
TTL logic machine starts
2 Mic inputs with EQ
Broadcast cue

audiomarketing ltd.

a subsidiary of audiotechniques, inc.

142 HAMILTON AVENUE, STAMFORD, CT 06902 USA TELEPHONE: 203 359 2315

for additional information circle no. 65

monitors, combines the functions of both wooden enclosures and foam-lined carrying cases with the integrated unit being more compact than a separate enclosure and case.

Some of the units incorporate Electro-Voice transducers and a variety of high-end components, including tweeter-protective circuitry, designed especially for the requirements of each type of enclosure. The components are matched for efficiency, wide dispersion, and the greatest sound pressure level before feedback. Acousitcally transparent grill foams complete the units. Handles and clasps are recessed and stacking ribs are provided to facilitate handling and transit. There is no case to store, saving appreciable space on the job.

AMANITA SOUND, INC.
40 MAINE AVENUE
EASTHAMPTON, MA
413/527-6910

for additional information circle no. 66

SHURE ANNOUNCES NEW PROBLEM-SOLVING MONITOR SPEAKER

Called the Model 702, the new monitor is said to be the result of on-stage performance analyses by Shure engineers to determine exactly the performance characteristics a monitor speaker must have if it is to completely satisfy the needs of musical groups and individual performers.

It was determined that one of the primary design features of such a speaker is wide-angle high frequency dispersion. In the Model 702, this feature is provided by a new concept in tweeter configuration that disperses the sound in a broad pattern, which allows entertainers to freely move about on stage, without sound loss or distortion.

Another important feature of the Model 702 is its shaped frequency response, with boosted midrange and controlled bass roll-off. This feature enables the Model 702 to cut through intense ambient sounds on stage.

The Model 702 is also designed to be extremely efficient and can be used with virtually any amplifier capable of delivering up to 50 watts to 16-Ohm load. Sensitivity is 97 dB SPL at 4 feet (1.2 m) with only one Watt input.

It may be used in either of two positions for added flexibility: at 60° for distant throw — and at 30° for close throw. The 702's volume control is recessed to prevent breakage and it is also equipped with parallel phone jacks for interconnecting 702's using only one cable from the amp.

The unit measures 395 mm high, 533 mm wide, and 264 mm deep (15-9/16" x 21" x 10-3/8"). The weight of each speaker is just 9.1 kg (20 lbs.).

Available as accessories are a 15 m (50 ft.) cable with phone plugs and a slip-over cover.

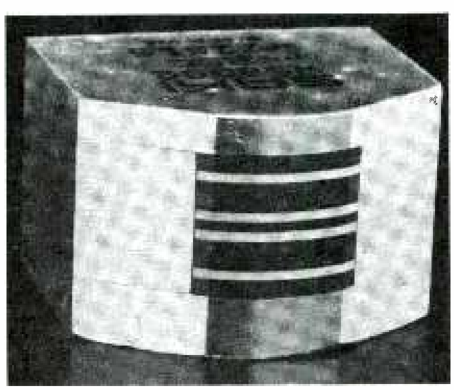
User net price for each Model 702 speaker system is $238.00.

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

for additional information circle no. 67

NEW FERRITE REPLACEMENT HEAD FOR AMPEX RECORDERS DEVELOPED BY SAKI MAGNETICS

Saki Magnetics, of Santa Monica, California, introduces the first hot pressed glass bonded ferrite head for Ampex AG-440 and 350 type recorders available in all track formats. The new



Saki head is manufactured of hot pressed ferrite with glass bonded gaps. The head is compatible with existing electronics. It is said by Saki that this new ferrite will outwear a standard metal head by 10 to 15 times life.

SAKI MAGNETICS
1649 12TH STREET
SANTA MONICA, CA 90404
213/451-8611

for additional information circle no. 68

"PHASE-CHECK" ANNOUNCED BY AMERICAN CONCERT SOUND

The "Phase-Check" allows the user to detect any incorrect phase conditions in a matter of seconds. The device consists of a separate pulser, which connects to the system's input and a phase indicator which designates either plus or minus phase with a red or

green LED. The unit has a built-in microphone for checking speakers, plus an auxiliary input for checking crossover outputs, amplifiers, transformers, microphones, etc.

Price: $150.00.

AMERICAN CONCERT SOUND
731 WEST 23RD STREET
AUSTIN, TX 78705
512/472-2437

for additional information circle no. 69

AUDIO MARKETING PREVIEWS NEW H-H AMP

Delivering 500 watts per channel, and occupying only 3½" of rack space, the S500-D is said to be ideal for PA applications

The S500-D's high damping factor helps to extend speaker life by reducing over-shoot and doubling. It is said to be 50% more efficient than conventional amps while forced cooled dissipators keep it cool even with 2½ Ohm loads.

A modular output section which can be replaced in a matter of minutes is another advantage. Price is under $1,000.

AUDIO MARKETING, LTD.
142 HAMILTON STREET
STAMFORD, CT 06902
203/359-2312

for additional information circle no. 70

NEW TAPCO POWER AMPS FEATURE POWERLOCK™

TAPCO announces the addition of dual-channel power amplifiers, featuring PowerLock™, to their line of mixers and equalizers for pro sound

use. Both basic amplifier models may be operated as two independent channels, or as a single-channel unit in bridged configuration. Model CP120 is rated at 61 watts per channel, 122 watts bridged. Model CP500, rated at 255 watts per channel, is capable of 510 watts output in the bridge mode. Model CP500M is rated the same as the CP500, but has a readout package that includes true peak-reading meters, a blown fuse indicator, and a thermal protection indicator. Both basic amps are designed for 4-Ohm operation for greater multiple-speaker hookup capability.

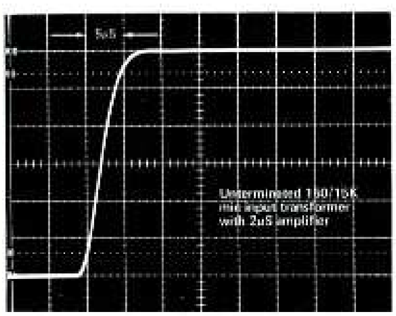
PowerLock™, an integral part of the TAPCO design, senses any large input signals that could cause prolonged clipping. PowerLock™ then controls the output level during the period of time that would otherwise be perceived as clipping distortion. Normal music transients, less than 1 ms in duration, are allowed to pass unaltered. The result is complete absence of distortion and greater average power output, with input signals up to 30 dB above the amp's normal +4 dBm sensitivity.

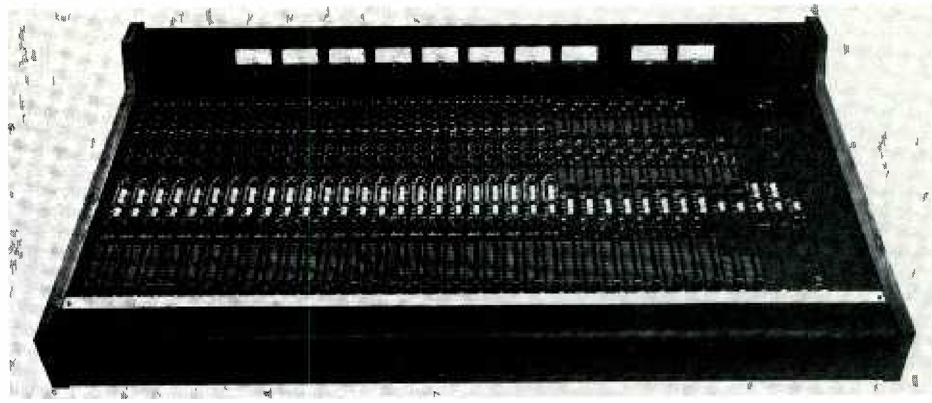
The amplifiers provide three PowerLock™ threshold settings — full, half, and quarter power — for absolute speaker protection against damage from overload. Control activity is indicated by peak-stretching LED's on the front panel. The LED drive circuitry is so designed that instantaneous peaks can be noted with extreme accuracy.

The amps feature double stacked ¼" phone jack inputs, all-steel monocoque construction for professional, on-the-road use, and high reliability components throughout. Transformer-coupled balanced XLR inputs are available as an option at additional cost. The CP120 has a stereo headphone jack on the front panel for monitoring purposes.

All of the new TAPCO PowerLock™ amplifiers have relay load coupling for completely silent operation, as well as gross fault shutdown and overtemp thermal cutoff. Positive protection against DC at the output terminals is also provided.

With PowerLock™ switched in, TAPCO amplifiers sound extremely clean, with clear articulation and sharp definition at extreme sound reinforcement power levels. These





amplifiers are especially suited to bi-amplification, where a CP500 could be used for the bass drivers, and a CP120 used for the high frequency drivers.

Pro net prices are: CP120, $339; CP500, $649; and CP500M, $779. Balanced input option prices will be announced in approximately 90 days.

TAPCO
3810 148TH AVENUE, N.E.
REDMOND, WA 98052
206/883-3510

for additional information circle no. 74

TEAC/TASCAM MODEL 15 CONSOLE

Bill Cawlfield, director of product development at TEAC, in commenting on the Model 15 said, "it materialized out of a need for more outputs for Tascam's 80-8 and 90-16 recorder/reproducers and because feedback from the field zeroed in on such a unit.

"The Model 15 is completely new. It maintains the broadness of the Tascam line and is the first big board designed specifically for mass production so that the unit — again in line with general Tascam marketing philosophy — is available at an affordable price."

Cawlfield said the sonic quality of the new mixer is faster in terms of transient response, due to the all-new electronics. "You can do many more things with the 15," he explained. "The sophisticated echo circuit, for example, can send the reverb signal to print, or the studio, or to the control room." He pointed out that the Model 15 also has a great flexibility, a necessary quality for the technically expanding PA market.

Features of the new unit include: switchable six-bank equalizer; new knob controls that allow pre- and post-fading for both cue and echo mix; two 8 x 2 sub-mixes that can be used separately or cascaded and from which either bus or tape can be monitored, and can be used for the control room and/or studio; new feather-light 100-mm sliding pot controls; and all plug-in modules for easy removal.

The power supply is a separate unit, isolated from the mixer by up to six feet of cord, thus further reducing the possibility of hum.

According to Ken Sacks, national sales manager of the TEAC/Tascam product line, "the Model 15 has a nationally advertised value of less than $9,000 (24 in, eight out) and $7,000 (16 in, eight out)."

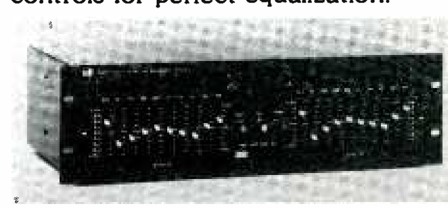
TEAC CORP. OF AMERICA
7733 TELEGRAPH ROAD
MONTEBELLO, CA 90640
213/726-0303

for additional information circle no. 75

AUDIO DEVELOPMENTS TYPE 1500 AUTOMATIC GRAPHIC EQUALIZER

The Type 1500 Dual-Channel Automatic Graphic Equalizer is an accurate control of acoustics, elimination of feedback and adjustment for optimum acoustic response.

Included is a precision pink noise alignment source, Red and Green LED's on each of twenty ±12 dB range controls for perfect equalization.

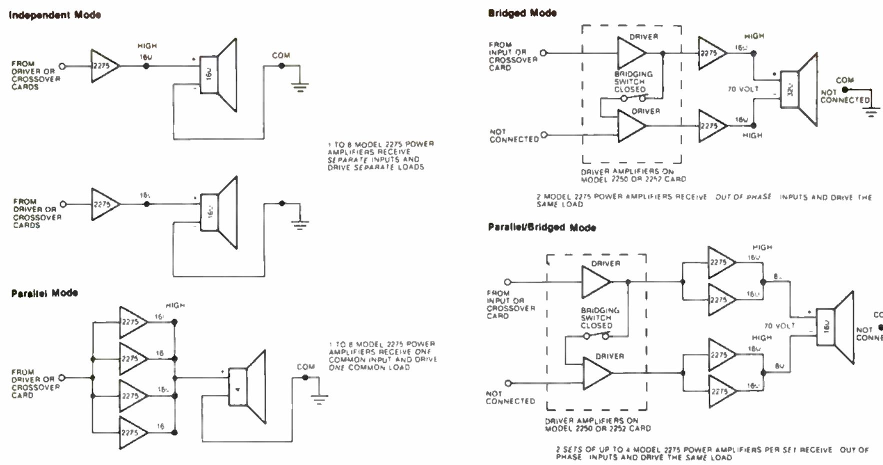


Ultra-low noise and low distortion double-pole Butterworth active bandpass IC octave filters to ANSI standards are used along with digital comparator circuits. The filters used are said to be a generation ahead of older "gyrator" types. Rack mounting and 2 year limited warranty are standard.

Professional price: $795.00.

AUDIO DEVELOPMENTS INTERNATIONAL
530 RAMONA STREET
PALO ALTO, CA 94301
415/321-3035

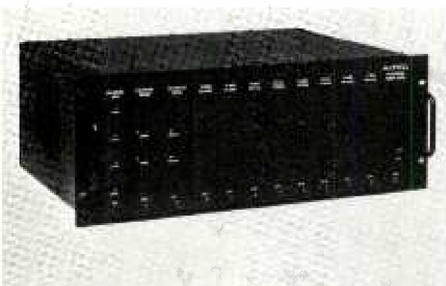
for additional information circle no. 76



Operational Modes for the Model 2275 Power Amplifiers

ALTEC LANSING INTRODUCES INCREMENTAL POWER

According to Paul B. Spranger, Vice President-Engineering, this unique system consists of a 7" rack-mount card cage which contains up to eight 75-watt power amplifiers, an electronic crossover, a balanced or unbalanced input card, and special driver amplifiers with matrix switching for console-like signal processing.



"The eight power amplifiers contained in the card cage can be combined in increments of 75 watts to meet almost any audio application," Spranger said, adding, "it can be used in parallel mode to drive high-power, low-impedance loads, or in bridged mode to drive balanced 70-Volt lines among others."

The system goes beyond the preliminary stages of plugging in modules and switching switches to explain system design with Incremental Power. Each of the 5 design examples pictured illustrates one possible configuration for the Incremental Power System. It can be seen that the strength of Incremental Power is its flexibility. In one configuration or another, Incremental Power can be applied, it is claimed, to just about any type of sound system.

The switches on each of the modules perform exactly the same types of functions as the point-to-point wiring between conventional components in a traditional multi-component system. Individually, the Model 2275 Power Amplifiers work very much like any conventional 75-Watt/16-Ohm power amplifier. When combined in parallel, bridged, or parallel/bridged modes, the Model 2275's behave very much like conventional amplifiers. The user simply treats each *set* of paralleled, bridged or paralleled/bridged 2275's as if that *set* were a single amplifier.

ALTEC CORPORATION
1515 S. MANCHESTER
ANAHEIM, CA 92803
714/774-2900

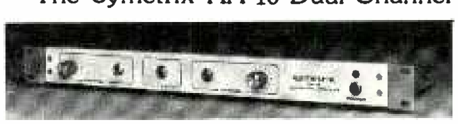
for additional information circle no. 77

SYMETRIX HA-10 DUAL CHANNEL HEADPHONE AMPLIFIER

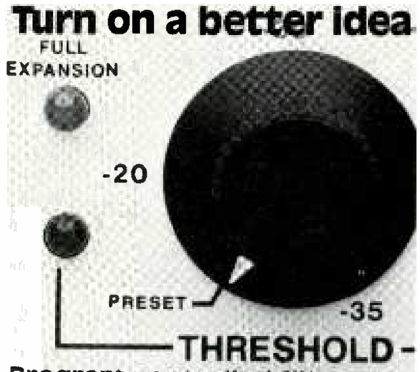
The HA-10 Headphone Amplifier is a device specifically dsigned as a compact, reliable, low-cost amplifier for powering headphones in recording and broadcast applications. The rack-mountable, two channel amplifier has on it's front panel separate (monaural) outputs for each channel and a combined 1 + 2 (stereo) output. The HA-10 delivers better than 10 Watts RMS per channel into 4 Ohms and will power headphones of any impedance to maximum listening levels.

The rear panel contains a total of 10 jacks including for each channel: 2 mixing inputs, 2 parallel inputs, and 1 mono output intended for powering headphones or small high efficiency monitor speakers. Although the inputs are designed to accept line level signals, the HA-10 has enough gain to produce near full output from low level sources such as electric guitars or microphones.

The Symetrix HA-10 Dual Channel



Turn on a better idea



Program-controlled filter suppresses residual background noise in audio reproduction systems. Selectable low frequency, high frequency, or wideband suppression modes.

Inovonics' DYNEX Noise Suppressor

Model 241 – $280

Inovonics Inc.
503-B Vandell Way
Campbell, CA 95008
(408) 374-8300

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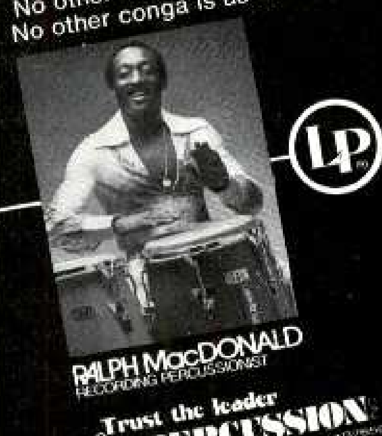

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Continued from Page 15 —

Late News

Digital Standards . . .

reports on the advantages and disadvantages of each frequency will be prepared by committee members for further discussion.

Analog-to-digital conversion was discussed briefly. The general feeling was that a 16-bit system is desirable, though not presently used in consumer systems; and that the record written on storage media should have a 16-bit slot. Pre- and post-emphasis were discussed; they are now used for consumer systems, and this practice is under consideration for commercial systems, as well. Pre- and post-emphasis are not used for studio systems, and the consensus was that this practice should be continued.

The next committee meeting is scheduled for February 1 and 2, 1978 at the Sheraton Hotel in Atlanta, Georgia. The meeting will be held in conjunction with the SMPTE Television Conference. Interested persons are welcome to attend. Details regarding the upcoming meetings may be obtained from:

JOHM G. McKNIGHT
MAGNETIC REFERENCE LABORATORY
229 POLARIS AVENUE
MOUNTAIN VIEW, CA 94043
(415) 965-8187

MOBILE FIDELTIY SOUND LAB LABEL FOUNDED BY MILLER

In reaction to what is claimed to be the serious degeneration of quality of mass-produced disc releases, executive producer, Brad Miller has announced the formation of the new label.

According to Miller: "... the label was originated as a matter of self-preservation, and the initial four releases contain three classic *Mystic Moods Orchestra* recordings. The trademark of the label will be, "Original Master Recording(s)", with all master lacquers cut at half-speed, and both plating and pressings obtained from plants overseas.

"The venture is intended to be a *haven* for quality minded producers and engineers, who need to reach the audiophile market with their product. Distribution will be primarily through audio dealers and via direct mail; we will be interested in *entertainment* values of a given original master tape; for which we will guarantee a superb transfer-to-disc and subsequent pressing.

"May I personally invite producer/engineer inquiries, whether or not a given artist is exclusively contracted for, in the hope that our low volume, high quality, premium retail price would be thought of similarly to that of an open reel tape licensee. (There is a market for open reel tapes, but most record companies would rather not fool around with it, so they choose to license the format to a third party.) That is how the Mobil Fidelity Sound Lab label intends to serve the audiophile marketplace.

MOBILE FIDELITY
P.O. BOX 2157
OLYMPIC VALLEY, CA 95730
(916) 583-2433

LeKASHMAN REJOINS ELECTRO-VOICE AS V.P. MARKETING

After an absence of six years, Larry LeKashman has rejoined Electro-Voice as vice-president, marketing, a position he held during the formative years of the company, according to an announcement by Robert Pabst, president of E-V.

Since leaving E-V he has served in executive responsibilities in two of the largest distributor organizations in the United States. Commenting on his return, he indicated a desire to participate in the basic audio business

LeKashman *Pabst*

at the manufacturing level, noting that "E-V has significantly strengthened its engineering and manufacturing operations under Bob Pabst".

NAUTILUS RECORDINGS RELEASES FIRST FULLY *dbx* ENCODED LIVE SESSION IN TAPE FORMAT

First In Line, The first contemporary music, limited edition tape recording, using full *dbx* encoding, has been released.

Nautilus Recordings, a division of Orion Marketing, Ltd., produced and distributes the tapes, stating that with such encoding a recording can reproduce totally noise-free, full dynamic range music when decoded in playback through any *dbx* 150 series, 187 or 216 tape noise reduction unit.

"Consumers are clamoring for better

Consen Bevan Schulein Smith

software," said Larry Blakely, marketing manager of *dbx*. We're delighted to see someone taking this first step."

The music was recorded simultaneously with Nautilus' first direct-to-disc recording in a live session. Only 250 second generation tapes will be made, dubbed one-to-one in real time onto Ampex 456 Grand Master tape. The original master was recorded on a Studer two-track recorder at 15 ips, and was encoded with a *dbx* 187.

Doug Gilmore, who has produced and written with such stars as Burt Bachrach, Delaney and Bonnie, Jerry Reed and John Denver, produced the recording with musicians who boast such prior associations as Boz Scraggs, Blood, Sweat and Tears, The Beach Boys, Barbara Streisand, among many others.

JBL NAMES CONSEN PRO-AUDIO FIELD SALES MANAGER; ANNOUNCES SOUND REINFORCEMENT WORKSHOPS

As announced by Peter Horsman, Division Manager for JBL, *Ewald Consen* will supervise and coordinate the activities of JBL regional managers in the U.S. He will also play a major role in product development and field evaluations for future JBL professional products.

Mr. Consen started with JBL as Field Service Engineer in 1970 and most recently held the position of Senior Regional Manager. Prior to joining JBL, he operated his own sound contracting firm.

Upon assuming his new duties, Consen announced a series of workshops on the design and installation of sound reinforcement systems to be held in selected U.S. and Canadian cities.

Consen describes the three-day long courses as a "unified and rational approach to the installation and layout of sound reinforcement systems." Several topic areas to be covered are practical acoustics (i.e., sound in free space, sound in enclosed space, wave length-dependent phenomena, etc.) and how to predict the gain of a sound reinforcement system (i.e., acoustic feedback and potential system gain, criteria for optimum sound system geometry, etc.). Several hours will be set aside for discussion of specific jobs or problems encountered by attendees.

The workshops will begin in February 1978, with one workshop to be held in a different city each month thereafter. Attendance will be limited to forty persons. The cities and months proposed are: February, Los Angeles; March, Houston; April, Atlanta; June, Chicago; July, Montreal; August, Kansas City; September, Vancouver. (A May workshop has been deferred because of the AES Convention in Los Angeles.) Exact dates and places will be determined and will be published in the next few months.

For further information on this course, interested readers should contact Mr. Consen at JBL, (213) 893-8411.

SHURE PROMOTES THREE IN ENGINEERING DEPARTMENT

Shure Brothers, Inc., Evanston, Illinois, has announced the promotion of William R. Bevan, Robert B. Schulein, and A. Douglas Smith to new positions within the company's Engineering Department.

Bevan has been appointed Chief Development Engineer-Electronics with responsibility for planning and conducting electronic product research and development programs. He previously held the position of Manager, Electronics Department.

Schulein has been named Chief Development Engineer-Acoustics, with responsibility for research and development activities for such acoustical products as microphones, loudspeakers, and loudspeaker systems. He was formerly Manager, Electracoustical Systems.

Smith has been named Manager, Technical Planning, and becomes responsible for project planning, project management, and development of application information. His previous position was Manager, Electronics Development.

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The Harrison 864 AUTO-SET is a process control microcomputer designed specifically for the Audio Industry. The design philosophy of AUTO-SET has been to complement the mixer and producer. Control and operation can be as simple as turning on the power and pressing one button. However, the true power of the system can be better appreciated after a short period of learning and use.

Data manipulation is an important feature of the 864 AUTO-SET. In the automation program, up to 4 separate mixes may be stored on each channel of the multitrack machine. Each console channel can be individually set to read one of the mixes from the multitrack. This allows important mix decisions to be made after an update or rewrite, not before.

The choice of one of the 4 mixes on each of 63 console channels is called a situation. Up to 10 situations can be stored in semiconductor memory in the AUTO-SET. These situations can then be called directly in sequence through the jump command, the crossfade advance or the time advance.

The Preset program in the 864 AUTO-SET is designed for live performance, television production and recording, where no recorder channels are available for data. Up to 630 separate snapshot mixes or static mix scenes can be stored on a 3M 100 Series data cartridge. Mixes may be named in blocks with a 6 character label for easy identification. Virtually unlimited data manipulation is possible with a 16 deep FIFO mix stack, 10 general purpose mix registers, cross fade X and Y registers and a background 10 wide tape register.

Level Sho is an important display feature of AUTO-SET. The setting of the 63 console channels or the return automation or preset leven can be displayed on the screen allowing level matching and confirmation.

The AUTO-SET employs a multiprocessor approach for optimized execution time and expanded features. Complete data management is included as part of the system, minimizing mixing time and effort. Many software packages will be available with applications in multitrack recording, live performance, television production and master control.

AUTO·SET

Part of the NO COMPROMISE philosophy at

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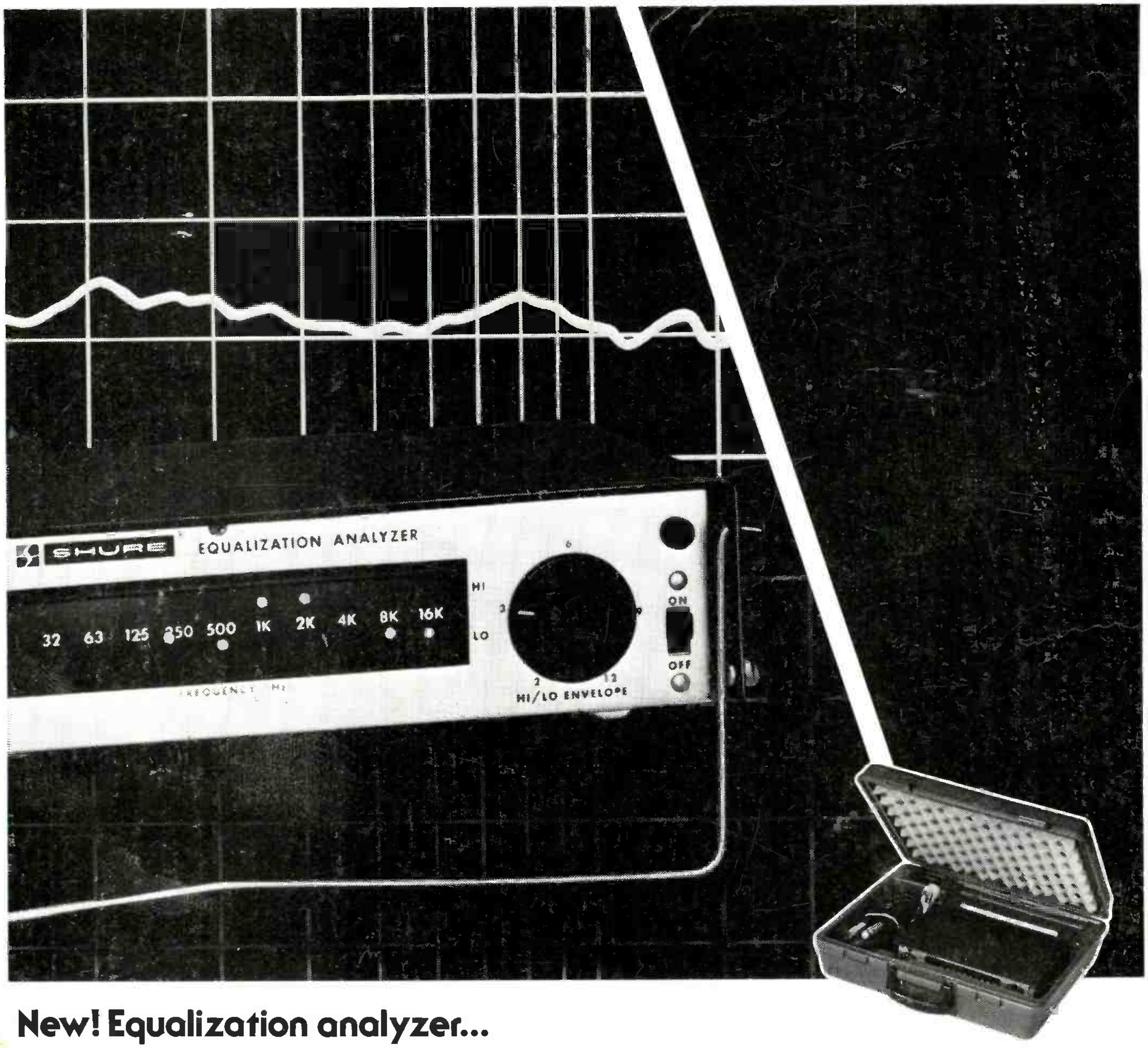

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The portable, 11-pound system (which includes the analyzer, special microphone, accessories, and carrying case) puts an equal-energy-per-octave "pink noise" test signal into your sound system. You place the microphone in the listening area and simply adjust the filters of an octave equalizer (such as the Shure SR107 or M610) until the M615 display indicates that each of 10 octaves are properly balanced. You can achieve accuracy within ± 1 dB, without having to "play it by ear."

Send for complete descriptive brochure AL558.

Shure Brothers Inc.
222 Hartrey Ave.
Evanston, IL 60204
In Canada:
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TECHNICORNER

The M615 Analyzer's display contains 20 LEDs that indicate frequency response level in each of 10 octave bands from 32 Hz to 16,000 Hz.
A rotary hi/lo envelope control adjusts the HI LED threshold relative to the LO LED threshold. At minimum setting, the resulting frequency response is correct within ± 1 dB. Includes input and microphone preamplifier overload LEDs. A front panel switch selects either flat or "house curve" equalization.
The ES615 Omnidirectional Analyzer Microphone (also available separately) is designed specifically for equalization analyzer systems.

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