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Volume 9 — Number 1

Recording engineer producer

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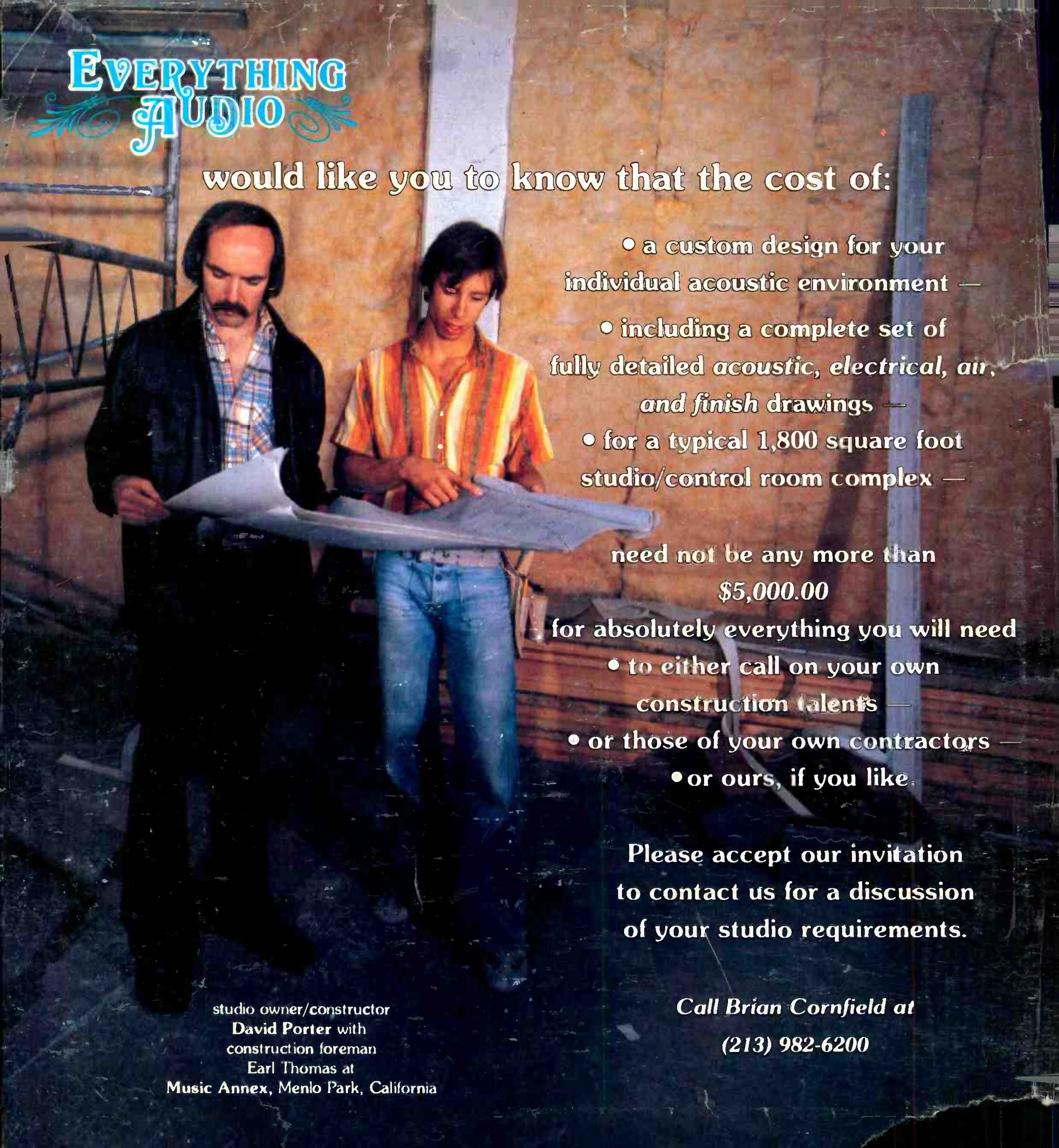

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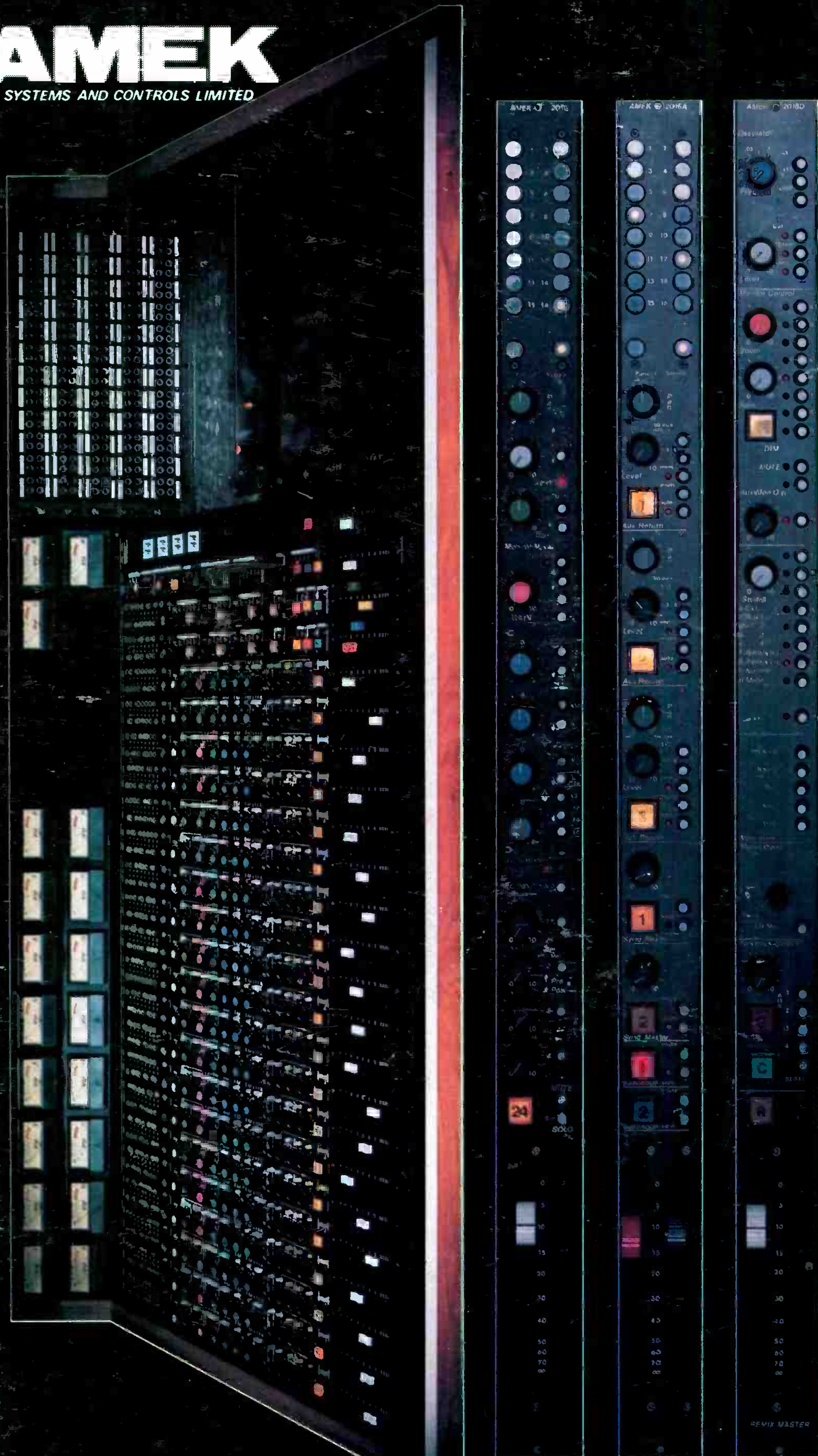

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design and construction aren't found in your local equipment store, where studio design is merely a hobby or sideline. It is impossible to build a truly professional studio from a textbook.

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– the magazine produced to relate . . . Recording ART to Recording SCIENCE to Recording EQUIPMENT.

"RECORDING engineer/producer" is published six times a year by RECORDING & BROADCASTING PUBLICATIONS, 1850 Whitley Avenue, Hollywood, California 90028, and is sent to qualified recipients in the United States. One year (six issues) subscriptions for other than qualified individuals or companies may be purchased at the following rates:

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FEBRUARY 1978
VOLUME 9 — NUMBER 1

— CONTENTS —

Cover Photography:
Dave Harrison
Harrison Systems
Nashville, TN

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SPECTRA SONICS Model 1024-24 Audio Control Console at United Audio Recording, San Antonio, Robert Bruce, General Manager.

Quality: **SPECTRA SONICS** audio control consoles show the care and attention to detail that are the mark of the skilled American craftsman. The internal wiring, module construction, console housing, and the control display reflect the precision and distinctive craftsmanship that is characteristic of **SPECTRA SONICS.**

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Letters & Late News

from: **Robert A. Bloom**
President
Audio Designs and Mfg., Inc.
Roseville, MI

I have just finished reading Mr. Buff's article concerning noise measurements, the article was certainly enlightening. I would strongly suggest that Mr. Buff and the Editors more carefully research their articles before committing them to print, especially when impuning the integrity of U.S. Console Manufacturers as a whole and claiming that we play a game of specsmanship and implying imported console manufacturers do not. If Mr. Buff wishes to levy an attack let him indicate whom he is attacking and back it up with data.

ADM meets most imported console manufacturers in the market place and we are cognizant of their specifications and method of presentation. We find that some imported units take poetic license as well when writing specifications. One prominent European manufacturer specifies an EIN of *greater than –125 dBm* and Mr. Buff states and we agree, that the theoretical limit is –124.8 dBm. So Mr. Buff, we now cannot really compare even the imports specifications. They do not have a standard either. Remember, Mr. Buff, the seed of doubt, when planted in a potential users mind, is a difficult one to remove.

Reply from:
Paul C. Buff
Allison Research
Nashville, TN

Mr. Bloom's response to my article on console noise specifications is a classic example of what I refer to as putting one's foot in one's mouth. In one sentence he points an accusing finger at both the author and the magazine for "impuning the integrity of the U.S. console manufacturers", while in another sentence he agrees with the content of the article.

In the article I did not categorize the entire U.S. console industry, nor did I exclude manufacturers from other countries. I accused only those who I considered as being guilty of perpetuating a myth, either through ignorance or through, as Mr. Bloom states, "poetic license". Ignorance is forgivable, but knowingly mis-stating technical specifications, because everybody else does, is not.

As for the prominent European manufacturer, he and probably other exporters, stands equally accused, but with one understandable reservation. When you do business in a foreign country, you follow the rules of that country, right or wrong. I believe the adage goes, "when in France, do as the French do."

As for myself, as a manufacturer who serves the same customers as the console industry, my allegiance must be to those customers, if I am to expect any serious degree of credibility as a manufacturer.

I believe that the magazine *(R-e/p)*, as well, has demonstrated that its editorial responsibility is fundamentally obligated to act on behalf of its readers regardless of the impact that action might have on certain of its advertisers. Putting aside all these childish games of accusation and buck passing, wouldn't it be a simpler solution for the industry to just clean house and formulate a correct and mutually acceptable format for presentation of technical data to

From the Publisher:

PUBLICATION SCHEDULE

Briefly stated, *R-e/p* has and will attempt to schedule the mailing of each bi-monthly issue on the 26th of the even numbered months: February 26, April 26, June 26, August 26, October 26, and December 26.

This schedule, although not consistent with the usual newstand publication practice of pre-dating issues, is the one which best traces the profile of the audio year. It enables the publisher to best prepare issues for mailing as close as possible, reflecting latest product and news announcements, to the occurrence of both the May and November AES Conventions while maintaining the constant sixty-day interval between publications. Indeed, these target distribution dates have been met within, at worst, a few days to a week.

As *R-e/p* begins its 9th year of publication, schedules notwithstanding, our primary objective will be to continue to both solicit and originate the most authoritative kind of editorial material possible.

Thanks —

M.G.

Are you really serious about a new console?

Model Pacifica 28 Input/16 output

7200M Input/Master Mix Module

We know that buying a large multi-track console is no small decision. For most professionals, it's one of the largest investments you'll make — a decision that you'll have to live with for years.

There are a lot of companies making consoles. Many perform adequately. Others are compromises. Few have all of the features and performance at a reasonable price. So, what are we leading up to? A simple statement of fact that you should consider seriously if you're really interested in an outstanding console system: Quad/Eight has an enviable reputation for quality and reliability. It's something we've worked at for over 10 years. We've also had a reputation for building the industry's most expensive systems too. Now, relax. Our new modular series consoles look expensive. Truth is, they're priced right in the same category as our best competition. In addition to having the best human engineering for operational ease, they're loaded with more features and performance:

- 3 band, 33 overlapping frequency equalization
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- Six auxiliary mixing busses from each input
- Two solo mixes, monitor & positional
- Discrete amplifiers used in the primary signal paths
- High-quality conductive plastic rotary controls
- Penny & Giles Faders
- Color-Coded aluminum knobs
- Individual phantom power switching
- Four fully equalized echo returns
- +28dBm output level
- Noise: —129dBm E.I.N, I.M. Distortion: 0.1% max.

If you're really serious about a new console and the quality of your work, then do yourself a favor and contact us for full information on a new outstanding line of modular consoles.

*The Coronado, 40 Input/24 Output equipped with Compumix III available in October, 1977.

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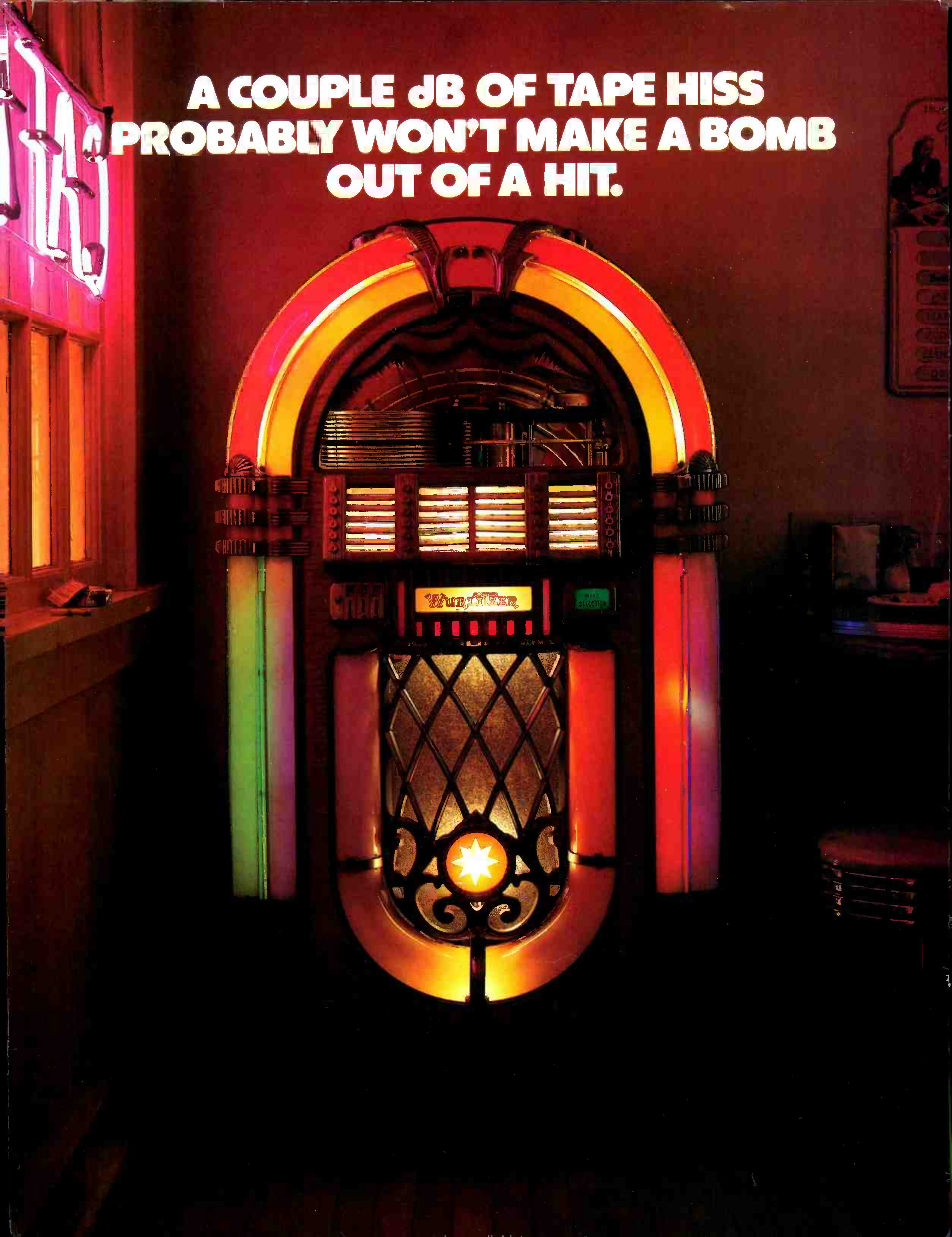

A COUPLE dB OF TAPE HISS PROBABLY WON'T MAKE A BOMB OUT OF A HIT.
WURLITZER

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 aboveboard. 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.

its customers?

Is technical credibility really too much to ask from a technically oriented industry in a democratic society? Is the alternative of an FTC house cleaning (such as was brought about by smugness of portions of the Hi-Fi industry) a more attractive alternative?

By the way, your underlined statement "greater than -125 dBm" should read "an EIN lower than -125 dBm", to be technically correct.

and still more on the subject . . .

from: **John G. McKnight**
Magnetic Reference Laboratory
Mountain View, CA

Paul Buff discussed "Console Noise Specifications" in *R-e/p, December 1977.* Indeed, as he says, there is much confusion on this subject. Buff presents many good ideas, although I disagree with his "definitions", and with a few of his calculations. But the purpose of this paper is to offer some comments, ideas, and references that add to Buff's paper.

I've not spent any space on items I disagreed with, because I basically agree with Paul's conclusions, and because the parts I disagree with ("definitions" and some of the math) are really extraneous to the basic problem. In fact, they disappear completely if my recommendations are followed.

1. FORGET "POWER MATCHING"

Much of the confusion in audio systems in general, and in noise measurement in particular, comes from trying to describe signal and noise *power* characteristics, because (as Buff says) for best noise performance, the source must *not* be loaded, and therefore there is *no* input power whatsoever! Snow [1] and [2] has interesting discussions of this. The present IEEE Standard [3] goes to a great deal of effort to force *all* kinds of systems — equal source- and load-impedance systems, "bridging" (non-loading) systems, and all things in between — into a "power transfer" mold. The result is self-consistent, but has little else to recommend it — it is very difficult to understand, and I feel it is of little theoretical or practical value. (We tried in 1968 . . . 1971 to update this standard, but it was not accepted. I think we should resurrect that draft.) Smith and Wittman [4] also spend a lot of words on this problem. What they calculate about "EIN in dBm" and "EIN in dBV" is technically correct, but ultimately not very illuminating, except in a historical context of the IEEE Standard [3]. If anyone really wants to know where the "extra 6 dB" comes from, see IEEE [3], and Haefner [5].

Forget power transfer and power levels (dBm, Buff's dBme, etc.) completely — they are a useless and confusing fiction, especially in low-level input systems. Consider only signal and noise voltages and voltage levels.

2. IS THE SOURCE REALLY RESISTIVE?

If we are really serious about input noise measurements, we should ask "How good is our basic system model that assumes that the source impedance is a frequency-independent resistance of the rated value?" Snow [1] discusses this a little. The only resistive source that I can think of is the carbon microphone! All the others — dynamic-, ribbon-, capacitor-, and the piezoelectric-microcphones and phonograph pickups, and tape reproducing heads — are fundamentally reactive. Perhaps the pre-amplified capacitor microphone is resistive, but because of the pre-amplification, the console input amplifier gets a higher input signal level, so its noise is much less critical.

Werner [6] discusses the impedance of ribbon microphones, but he gives only the magnitude of the impedance, without the phase angle. Perhaps the microphone manufacturers have this information. In any case, measurement is not difficult — just time consuming.

If the sources turn out to be frequency-variable resistances (especially in the 8- to 16-kHz region), then does a measurement and specification of noise performance relative to a theoretical frequency-independent resistance have any value? I doubt it, but this deserves some thought.

But for now, I'll discuss measurements based on a frequency-independent resistance of the rated value, as though it had real meaning.

3. EQUIVALENT NOISE RESISTANCE

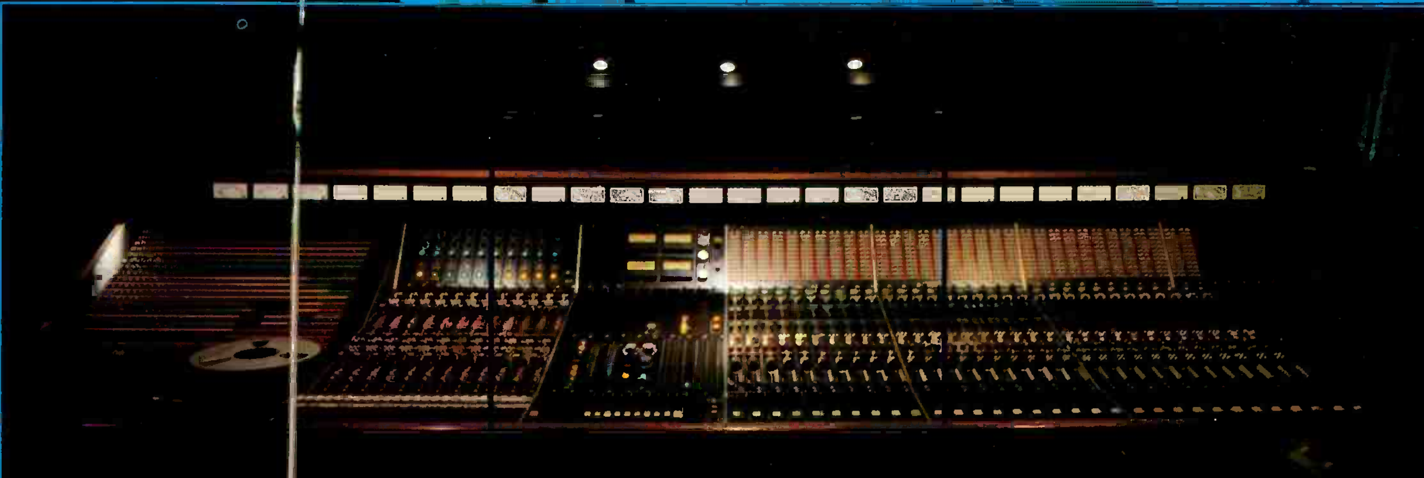
For design purposes, the designer wants to know the equivalent noise resistance of the input stage, in order to optimize its value if he has a fixed source resistance, or to optimize the source resistance if it is adjustable (such as with an input transformer).

This concept is detailed by Snow [1] and Argimbau [7] for vacuum tubes, and by Smith and Wittman [4] and their references, for transistors. Once the equivalent noise resistance R_{eq} and the source resistance R_s are known, the noise index is calculated directly [7] from:

$$N.I. = 10 \log_{10} [(R_s - R_{eq})/R_s].$$

4. MEASURING NOISE INDEX DIRECTLY

I agree completely with Buff's conclusion to measure the Noise Index (20 log noise voltage figure). Buff, and Smith and Wittman [4], mention only the method that Terman and Pettit [8] call the "Brute-Force Method" — measure the gain, the equivalent bandwidth, and the output noise, and calculate the Noise Index. Terman and Pettit also describe a "Noise Generator Method" which would seem to have many advantages in simplification of measurement and reduction of possible errors of measurement. It uses the principle that when two equal noise voltage levels are



The English Performer With An American Back-Up Group

It always takes time to break an English act in the States. Fleetwood Mac, Elton John, Rod Stewart . . . Each took years—and an infusion of American back-up talent—to make it in this market.

Since 1968, Trident's London Recording Studios have been at the centre of the English music and recording industries. From the Beatles' "Hey Jude" to Queen's "Bohemian Rhapsody," Trident's own equipment has been used in more heavily produced hits than you can shake a cricket stick at. We've learned what it takes to manufacture superlative recording consoles. Like extensive monitoring facilities for the producer and engineer's convenience; conductive plastic, Penny & Giles attenuators for reliable low-noise performance; and four-band parametric equalisation on each input of the new TSM consoles.

Ten years have also taught us the crucial difference craftsmanship can make. Trident's TSM, A-Range and Fleximix all have electrical specifications which ensure outstanding channel separation, low distortion and exceptionally quiet performance. We've come to recognize the role of ergonomics: the art of designing consoles with their operators in mind.

Yet until recently you Americans haven't seen many British boards in studios here in the States. Some Yank studio owners have been reluctant to commit themselves to consoles which might require special expertise to service or install.

Very well, then: Trident is pleased to announce their association with Studio Maintenance Service.* These technically minded Yanks appreciate fine Trident craftsmanship. They are a zealous back-up group who do more than stand behind the equipment they sell; they get under and into it when necessary. At any time of the day or night. No savoir-vivre. But a veritable plethora of exceptional, comprehensive service.

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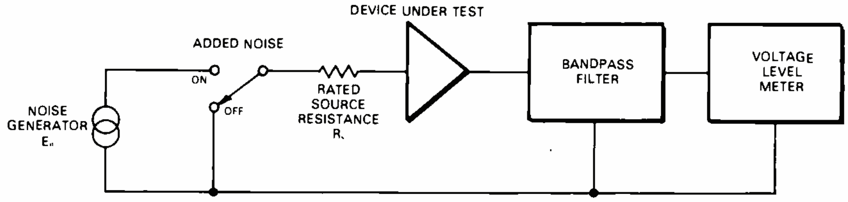

Figure 1: **BASIC CIRCUIT FOR MEASURING NOISE INDEX BY THE "NOISE GENERATOR METHOD."**

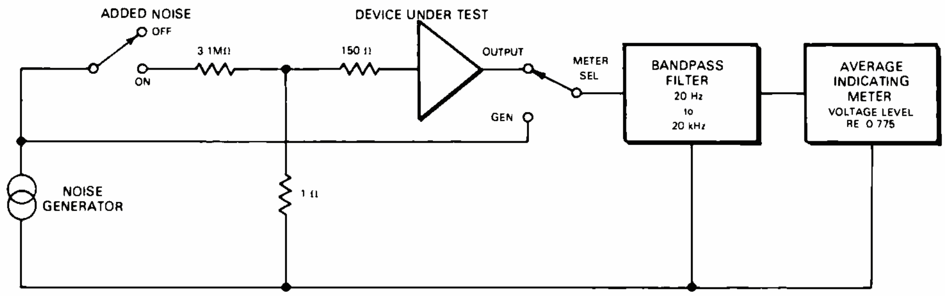

Figure 2: **PRACTICAL SETUP FOR A DIRECT-INDICATING MEASUREMENT OF THE NOISE INDEX BY THE "NOISE GENERATOR METHOD."**

These particular values hold only for a source resistance of 150 Ω, bandwidth of 20 kHz, and an average-indicating meter of voltage level re 0.775 V. (A balanced circuit could be used if needed.

Procedure:

1. Meter Selector to "Output"; Added Noise to "Off". Read Output Noise Level on Voltage Level Meter.
2. Meter Selector on "Output", Added Noise to "On". Adjust Noise Generator so that Output Noise Level increases 3 dB.
3. Meter Selector to "Gen.". Level read on Voltage Level Meter is directly Noise Index of this system.

added, the sum of the levels is 3 dB greater than either level. Thus a known noise voltage is added to an unknown noise voltage; the known is adjusted to make the sum just 3 dB greater than the unknown alone. Then the unknown must be equal to the known. In practice, one would connect a noise voltage E_n with bandwidth B in series with the rated source resistance R_s to the amplifier input, as shown in Figure 1. Measure the output level in bandwidth B with the added noise E_n disconnected. Then connect the added noise voltage, and increase it until the output level in bandwidth B increases 3 dB. Measure the noise generator voltage E_n in bandwidth B, and calculate the Noise Index from:

$$\text{N.I.} = 20 \log_{10} (E_n/E_{ref}),$$

$$\text{where } E_{ref} = \sqrt{(4kTR_sB)}.$$

This method still requires one bandpass filter to set the given bandwidth B of the voltmeter, both when measuring the output noise level and adjusting the added noise level, and when measuring the "known" added noise voltage E_n. But there are several other simplifications over the "brute-force" method: — There is no need to measure the gain, and get embroiled in that mess. The rms value of the noise generator voltage must be measured, but there is no need to correct for the response of an average-reading *output* meter, since it affects both the "with" and the "without" added noise reading equally. And, finally, any source loading losses are automatically included.

In a practical measuring system, rather than *calculating* the Noise Index from the noise voltage ratio, one could easily design an attenutator between the noise generator and the input circuit to take care of all of the miscellaneous mathematics, so that an average-reading voltage level meter could *directly* indicate the Noise Index. The values given in Figure 2, for instance, are calculated for use with an average-reading meter which reads levels referred to 0.775 V; with R_s = 150 ohms, B = 20 kHz, T = 300 K (37 °C). The level reading on the meter will be the Noise Index in dB, directly. How's that for simplicity? No goofs, and no fudge.

I must admit having not run these measurements myself, but it certainly sounds like an idea that the industry should consider. If it really works out in practice, it should be written up for publication, and be made into an industry-wide standard method. If this or some other methods work out in practice, and are acceptable to audio engineers, then I am sure that the Audio Engineering Society's Standards Committee would be anxious to help with the procedures of standardization.

REFERENCES:

[1] W. B. Snow, "Audio-Frequency Input Circuits", J AES Vol. 1, pp 87-94, 1953 January.

[2] W. B. Snow, "Impedance—Matched or Optimum?", J AES, Vol. 5, pp 66-70, 1957 April.

[3] "Gain, Amplification, Loss, Attenutation and Amplitude-Frequency-Response—Methods of Measurements of", ANSI C16.29/IEEE Std 150—1956; also Proc. IRE Vol. 44, pp 668-686, 1956 May.

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[5] S. J. Haefner, "Amplifier-Gain Formulas and Measurements", Proc. IRE Vol. 34, pp 500-506, 1946 July.

[6] R. E. Werner, "On Electrical Loading of Microphones", J AES Vol. 3, pp 194-197, 1955 October.

[7] L. B. Argimbau and R. B. Adler, *Vacuum-Tube Circuits and Transistors*, New York, John Wiley & Sons, 1956. See Chapter XV, Noise.

[8] F. E. Terman and J. M. Pettit, *Electronic Measurements*, New York, McGraw-Hill, 1952. See Chapter 8, Amplifier Measurements; Sec. 8-12 Noise in Amplifiers; Sec. 8-13 Noise Figure in Amplifiers; and Sec. 8-14, Measurement of Noise Figure of Amplifiers.

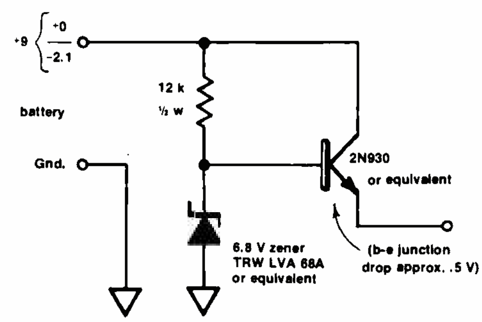
Editor's Note:

Mr. McKnight's comments arrived too late for a response by Mr. Buff in this issue. For additional information on the subject please see Mr. Buff's follow-on article beginning on page 82.

from: **Richard James**
Los Altos Hills, CA

I would like to comment on and add to Lewis Mark's good article about a budget click-track metronome in last October's issue. He mentions the inconsistency of the unit at turn on, suggesting to let it warm up (for about 8 bars; assumed 4/4 and medium tempo). The culprit of that problem is actually the specified 9 volt battery. The drain on even a fresh 9 volt transistor battery is the cause of the start-up problem and also the cause of more subtly and importantly the metronome almost imperceptible slow-down throughout a composition. Thus, the last four bars will probably be a few mm beats slower from the first four bars which could create big problems when editing.

The consistency of the unit may be improved by the addition (slightly more money well spent for musicianship) of the zener diode regulation circuit (current source) pictured below.



The drop in magnitude of the supplied regulated voltage probably will have no effect on the operation of the published circuit other than reducing the output signal

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

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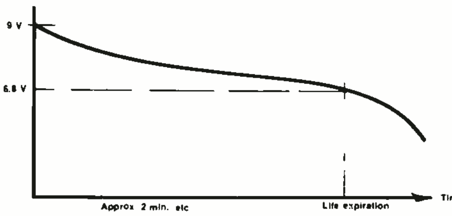
which could be compensated for by simply increasing volume, and having to recalibrate mm speeds if done previously (more likely just reference marks). I have not built and tried Lewis's circuit probably because I already have two different pocket unijunction metronomes revised with the regulator circuit and both work well. The best post metronome/A & Bb tone unit I have seen was produced privately by (Bob Downs in Orange County, 714/835-1033) Development Associates, P. O. Box 1193, Tustin, California 92680, but they would have had to double the price to sell them in music stores. So their production remains reasonably unknown.

Incidentally, the drain on battery life curve is shown below with the effects of zener regulation shown as a broken line.



NEW COMPUTER LATHE INSTALLED BY CBS RECORDS IMPROVES RECORD LENGTH AND SOUND QUALITY

A new computerized disc mastering lathe is now "on line" for general use at Columbia Recording Studios in New York and other cities. The patented CBS $250,000 DISComputer™ automated lathe, more than three years in the making, is said to bring with it dramatic improvements to the "state-of-the-art" in record mastering.

Developed by the staff of the Columbia Technology Center in Stamford, Connecticut, for the express use of CBS Records, the CBS DISComputer™ system integrates the memory and anticipatory abilities of a special set of algorithms developed for programming with the widely-used Scully and Neumann mastering lathes. Programming is based on extensive investigation of record grooves, shapes, parameters, distortions, and recording problems.

One of the most significant benefits of this major breakthrough in lathe design is its ability to anticipate loudness and softness of inputs more accurately, and respond to them more quickly than currently available commercial lathes. Records mastered on the CBS DISComputer™ system have consistently shown dramatic improvements in the signal-to-noise ratio and equally impressive control of distortion and a pickup's possibility of mistracking, it is reported by the company.

The most obvious benefit of the new mastering system is its ability to increase by as much as five minutes the amount of music on each side of an LP, and its delivery of increased recorded levels with enhanced sound quality.

Engineers can also use the CBS DISComputer™ system to re-cut identical masters from the same tape at will.

The CBS DISComputer™ system, developed at a cost in excess of $500,000, is presently available at Columbia Recording Studios in New York and Nashville, with installations already under way in Canada, Japan, and Europe. Further expansion of the system's availability is being made as quickly as possible, according to CBS Records Director of Recording Engineering, Erik A. Porterfield, "and ultimately will be available at Columbia Recording Studios in the U.S. and at CBS Records Studios world-wide."

Headquarters of the Columbia Recording Studios — where the first CBS DISComputer™ automated lathe is housed — are located at 49 East 52nd Street in New York City.

TEAC/TASCAM INTRODUCES UNIQUE FINANCING PLAN FOR BUYERS-TO-BE

For the first time in the history of the industry, a finance plan for those buying TEAC Tascam Series recording equipment has been established by one of the nation's leading lending institutions.

FinanceAmerica, a financial service company of BankAmerica, with headquarters in Allentown, Pennsylvania, has developed a new consumer program for Tascam dealers across the country. "We are providing the means for the dealer to sell through," Ken Sacks, national sales manager for TEAC Tascam Series, said.

"We have been told over-and-over again by dealers that people want to buy more than the dealers can sell," he said. "Historically, the biggest stumbling block in the way of a sale has been the lack of a sound finance program. In the past, a customer literally had to provide collateral, like his home, before he could qualify for a loan. Under the FinanceAmerica program, he fills out the application, and, if he qualifies, he can purchase the equipment on credit."

The application, in addition to obtaining the usual credit information, allows the prospective purchaser to list his agent or personal manager, union card number, and past, present and future bookings. According to FinanceAmerica, "This will allow our branches to verify the information supplied and more accurately gauge the real total income of the potential customer."

A separate section of the application is devoted to the studio operator, who is not a professional musician, but a technician who obtains Tascam equipment for the purpose of recording for musicians for a fee. Because "this individual most likely derives a greater portion of his gross income from this operation," he will have to supply either a W-2 or an audited financial statement.

"Tascam has long realized the need to supply equipment for the so-called semi-pro who work in their homes, creating music any time of the day or night," Sacks continued.

FinanceAmerica requires a down payment of 10 per cent for purchase up to $6,000.00, and 20 per cent over that. Terms range from 36 to 60 months, depending on the amount of the purchase. The rate is a 10 per cent add-on in those states permitted by law, and as close as possible to 10 per cent in the others.

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. . . continued on page 98

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AMPEX

engineer FRED CATERO at San Francisco's Automatt

by James Furman

At the age of 45, Fred Catero is perhaps the most visible proof that the recording studio scene is alive and well in San Francisco. A veteran of 28 years in recording, his career has encompassed every phase of the industry. He began in the "direct-to-disc" transcription days, and when recording on tape became available, he gained a reputation as one of the best tape editors in New York City. Later on, he moved to Columbia Records' New York facilities. Three years later, he was selected to become one of five mixers (out of a staff of 120 engineers). In this highly-sought-after position he gained experience in every category of music: classical, stage, pop, country, jazz, and rock.

During his early years at CBS, Fred met and began a life-long collaboration with a young CBS staff producer, David Rubinson. In 1969, Fred and David both moved to California and worked together at the Fillmore Corporation. Three years

Let's get right into it: Microphones?

Fred Catero: Okay. I've heard engineers say that they can only work with certain mikes. When we worked at Wally Heider's, some engineers I knew said they couldn't record unless they had a particular mike on the snare, to the point where they would actually cancel the session if the mike was being used elsewhere, and they couldn't get it. In all my years, I have tried very hard not to fall into that trap. Because in this day and age, there are too many good microphones, and too many pieces of equipment that you can use in conjunction with microphones, so that you shouldn't tie yourself into any particular mike.

I *do* like Neumann 87's; they're a standby in the industry. I still like the old RCA ribbon microphones — horns sound very good, fat and round. I like the Shure SM-56's, the dynamics — I use them a lot on tom-toms, the drum set-up. Sometimes I use it on guitar amps. It has that mid-range-y quality that seems to record very well when you go from tape to disc.

The idea is I get just the frequencies I want, and not all the extraneous shit. I use condenser mikes on anything that has really broad frequency response, where I want to get a really crystal clear pickup. But there are a lot of instruments, especially electric ones, that don't even have the range of even the cheapest mikes. So, why should I go out and put up a $500 Neumann that's going to give me all kinds of extra frequencies that are not necessary, that are not even generated by the instruments. I try to make it a rule to use mikes that fall within the range of the instrument. A tom-tom is most effective at a particular set of frequencies. A dynamic mike works very well. It's a better choice than a condenser mike — for close miking. I'm speaking of close miking because that's how we have to record nowadays. For far miking, I would generally use a very good Neumann. I love Neumann mikes because they have variable pattern, and they have the pad and the roll-off. You can do most anything with a Neumann.

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Under what circumstances would you place a mike far from the sound source?
Fred Catero: When I want a natural sound. Let's say I have three trumpets in the studio, and they were overdubbing. I would prefer to have the three trumpets as a section play into one mike, with the mike ten feet away, than to put a separate mike on each trumpet and try and balance it in the booth. The sound of the instrument is most beautiful and effective far out away from the instrument, than a mike shoved right in the bell of the instrument. The same way with a drum set, (like timpani), or acoustic guitar. An acoutic guitar miked five feet away, in a very quiet studio, gives a magnificent sound, if the studio is not dead. Our studio is a very beautiful sounding room, and whenever I can, I like to take advantage of that.

What about when you're doing basics, and leakage is a problem?
Fred Catero: Then I have to mike close, and that's when I use dynamics. Dynamics are limited in what they can do, and those limitations are very advantageous when you're close miking. They don't have the transient response that a condenser has. They don't have the frequency response. When you're miking very close, you have tremendous dynamic range happening. With a condenser mike . . . it does cause distortion, because it will go beyond the headroom of the board. A dynamic mike, just because of the way it works, doesn't have that much range. It's almost like a limiter. A condenser mike also has low frequency capability. It can pick up air-conditioning rumble, it can pick up harmonics that are way below useable audible range. That all translates to electrical energy at the board . . . It means I don't have as much room for the useable frequencies. If I use a dynamic that rolls off at 60 Hz, I have more useable energy for the same amount of voltage, which means it will sound louder, and I can record hotter.

EQUALIZATION

Here's another point. I try to do as much EQ'ing as I feel necessary as I record. Now, I know that there are engineers who put up mikes and record everything on the tape flat, and then, in the mix, they start EQ'ing. This is one of the things about making recordings that are very quiet and clean. You put on as much of the EQ as you think you're going to need when you record, because if you wait until mixdown, you are EQ-ing all the hiss as well. If a particular vocal track sounds best boosted 5 dB at 8,000 cycles, if you don't do it when you record you're also boosting the background noise at 8,000 cycles. Now, I don't say go hog wild. If you have a doubt, split the difference. If you think the voice needs some edge on it, but you're not sure how much, don't go all the way — put on a little, because at least you're that much ahead in making a quiet recording.

Then if you know in advance that you will need to cut the highs on a track, should you deliberately wait and do it in the mixdown?
FC: I would wait and do it in the mix, unless the highs that are coming through are causing distortion on the tape.

NOISE REDUCTION

I've mentioned before, I dislike using Dolbies. I dislike using any noise reduction device, because I have had so much trouble with the restoration of the signal back through the Dolbies. If you use the same system and set-up technique, Dolbies are fantastic. The trouble is, we have made albums using Dolbies, and have mixed down the the Dolbies, and it sounds fantastic in the studio. We take the tape to L.A. to have it mastered, and they have their own Dolby set-up. Now, unless those Dolbies are exactly matched, it can screw up the whole mix. Sure, the tape will be quiet, but the EQ will make it sound nothing like what you had. We've been burned so many times that way that I just refuse to use Dolbies. Dbx, I understand, is a lot more reliable from unit-to-unit, but I just don't like to use them for various esthetic reasons. I'd rather make every effort to get it properly recorded on the tape the way it should be, than to be conservative in my levels and in what I do, and then rely on electrical equipment to make a noisy tape quiet.

Suppose you were recording an instrument like a solo piano, where there were some really quiet places where noise might be much more apparent than it ever would be in rock music. Would you consider using noise reduction equipment in a case like that?
FC: Yes, I probably would, if it was one of those things that really required super quiet.

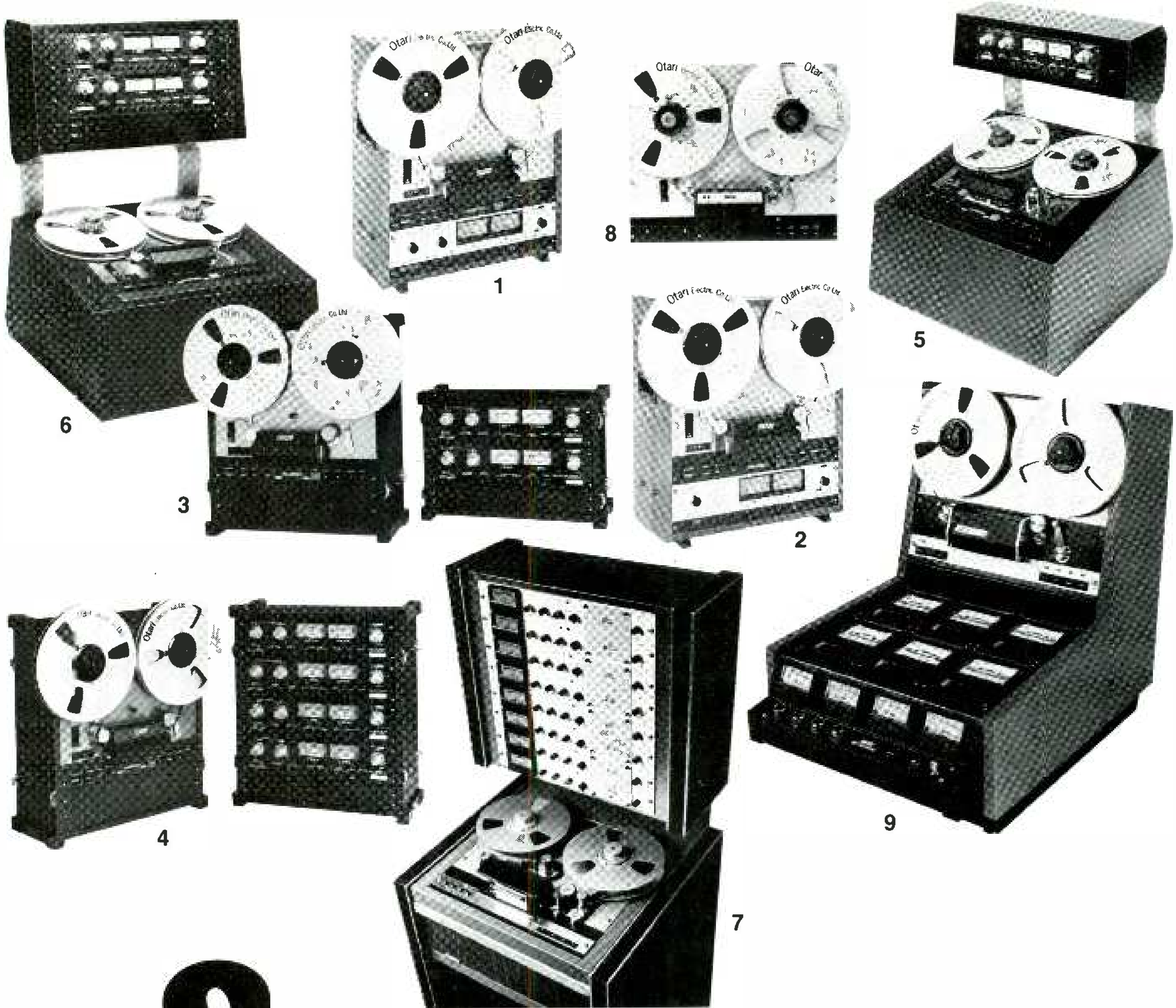
Let me say something else about making quiet recordings. First of all, use the right microphone. That is, a microphone that gives you the most information in the frequency range you need. Number two, is setting up your equipment, that is, the board, the tape machine, and everything else, to the optimum operating position. There is a point on the board where there is more hiss than you have to have, because you're not driving the module hard enough, so you have to run your fader higher up. Then, there's another point on the other end where you're hitting it too hard, so you don't have a lot of noise but you're in danger of having distortion. If you observe the point at which you're at the lowest noise level that the board will operate at, and at the highest level before distortion, you're already ahead. Then, the next step is to calibrate your tape machine in such a manner that you can record +3 peaks with a minor percent of distortion — at the highest allowable level that the tape can handle without causing excessive print-through or distortion.

There's one additional thing to observe that they don't tell you in almost any manual: On the new tapes you're allowed to record 3 dB hotter than the old standard operating level, with peaks of +3, which means 6 dB above the standard zero. After that, they say you can't record anymore because it just distorts. Basically, this is true if you're recording full frequency range information. But what they don't point out is that there are certain frequencies that the tape can record even higher than that. There are certain things that can be recorded higher, and the distortion resulting is an advantage, rather than a disadvantage. For instance, electric guitars. I can record them maybe 8 dB higher than standard. True, they may be a little more distorted than they sound naturally, but, not only do

Fred Catero
— *continued* —

later, the two men left the Bill Graham aegis to start their own production and recording firms. This culminated late in 1976 in the establishment of the Automatt (a division of David Rubinson and Friends, Inc.), a state-of-the-art, recording studio on San Francisco's Folsom Street. The Automatt's furnishings and decor are modest, almost Spartan, but no expense has been spared to provide the best and most sophisticated equipment possible. The centerpiece is an automated Harrison 4032 console, custom modified for the Automatt, and interfaced with the Allison memory-plus system which allows an engineer to build up to a final mix bit-by-bit, rather than all at once, since the board's memory circuits remember and automatically re-execute any previous parts of the mix that have been done correctly. Other equipment includes MCI 24 and 2 track recorders, and a Scully 8 track, all of which may be sync'ed together, as well as a host of patchable outboard gear.

Over the years, Fred Catero has received four engineering awards and ten Gold Records. He is President of the San Francisco Chapter of the National Academy of Recording Arts and Sciences (NARAS) and National Vice-President. Among the huge number of artists that Fred has worked with are: Janis Joplin, Chicago, Simon and Garfunkel, The Pointer Sisters, Laura Nyro, Aaron Copland, Barbra Streisand, Linda Ronstadt, and Robert Goulet.



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Fred Catero — Technical Director Michael Larner

they sound better, (by that I mean more distorted and violent), but because they're so loudly recorded it allows me to bring that track way down when I mix, which gets rid of a lot of hiss. Certain synthesizers, too. So, taking that into account, you don't have to record everything at a zero level. You can get away with letting certain things peak.

Then, too, when you mix down, there is no reason to have the tracks that are not activated going through the circuit. For instance, if the vocal doesn't come in until the second half of the tune, there's no point in leaving that track open in the board (until the vocal begins) — all it's doing is adding hiss. So, when you mix, you must learn all the cues of who plays where, what track is active where, and, when they're not working, remember to bring the fader down or kill the module so that those tracks are not operating when they're not playing. *That* keeps it quieter.

It's a lot easier to remember all those extra cues if you have an automated board like the Automatt's.

FC: Or, if you have a musical sense, which is very important for engineers. That's like an auto mechanic who's never driven a car. I don't say that an engineer has to have a master's degree in arranging or musicology, but he should have the feel. He should be able to tell when four measures go by; he should be able to recognize a guitar solo when it happens. But some engineers don't. Some engineers are so hung up on audio that they have no concept of what music is all about.

LIMITING

Do you ever use limiters?

FC: Yes. That's another thing that will help make quiet recordings. For instance, if you have an instrument that has tremendous peaks in it. By limiting it slightly, it allows you to record it a little louder on the tape. Even in mixdown. If you don't limit it, then you ultimately have to record it at a lower level to avoid distortion, to accommodate the flash peaks. This will make the overall useable energy on the tape at a lower level than need be, because the flashes are just momentary. So if you put a limiter in, and compress it slightly, you don't alter the dynamic range appreciably, the ear can't really tell, but it allows you to record louder on the tape. Sometimes, rarely, I use limiters to really squash the sound, but that's just to get a special effect, rather than to give me more level.

There's one more thing to observe, and that's phase. All engineers should be conscious of it. A lot of poeple reading this will say, "I know that, that's nothing new". But you'd be surprised at how many control boards don't have phase reverse switches on them which indicates to me, that until recently, most engineers didn't consider phase a problem. Otherwise, they would have demanded the switches be put in the boards. Phase problems occur the moment you use two microphones on any given instrument. You can't help it. It has nothing to do with electrical phase. It has to do with the acoustical phase that happens when two mikes are picking up a given sound from two different locations. As a rule of thumb, if you're recording with more than one mike, balance it up by listening to it monaurally. Start with one microphone, bring in the next microphone and reverse that phase of it. Set it for the sound you want. No matter what phase it's in, some frequencies are going to be cancelled, and some amplified. The point is, which ones are the ones you want? If you want a rich, fat sound, you will find that if the mike is in phase with the other mike, it will be a fuller sound. If it's out of phase acoustically, you'll have a tendency to cancel a lot of that richness in the lower range. In some cases that might be advantageous, if, for instance, a drum set is too tubby, sometimes throwing one of the overhead mikes out of phase will get rid of a lot of that low stuff.

Here's an example where it's so important that things be in phase. Two mikes on an acoustic guitar. You want it spread in stereo. You want one of those big Simon and Garfunkel guitars that's spread right across the speakers. If the mikes are acoustically out-of-phase, then, in stereo, it will sound all right. It won't sound perfect, because the sound will sound like it's coming from inside your head, but you'll hear it. But if each mike is picking up almost the same information, but out-of-phase, when it's played on the radio, you won't hear the guitar, because it'll cancel when it goes monaural. That's why you have to check it monaurally.

Fred, when you go into an unfamiliar studio, do you bring in any of your own equipment?
FC: I'll never bring in a microphone.

A Tape to orient yourself to the monitors?
FC: It helps to have something you're familiar with. Unless you're really cocksure of yourself, I advise young engineers or engineers who aren't that sure, to bring a tape that they're familiar with to their session. They should play it for themselves, not in front of everybody. I've seen engineers who bring the best tape they've got, play it real loud, stick out their chest, and it's like bragging. "Look what I've done." But there's a danger there. Suppose the people they're recording with aren't as good! It's almost like an open put-down. You're coming in flashing your big bucks in the ghetto, and people don't like that.

I would suggest he go in early, before the musicians show up, put on the tape and really listen. Don't sit there and dance to the music and hype yourself. Don't let the guy who works there do it all. Walk in the studio. Listen to the room. Look at the walls, and the floor, Check out the screens and the instruments. See what you got. See what kind of complement of mikes they've got. Usually, as long as the mikes are working and aren't ruined, and you're familiar with the laws of sound, you'll get a pretty decent recording. But, it helps to know what the speakers sound like.

A couple of the many Herbie Hancock albums you've recorded have been sound tracks for movies (The Spook Who Sat By The Door, *and* Deathwish). *Do you EQ differently, knowing a mix will be heard by film audiences in theaters?*
FC: I keep in mind the frequency response of film, but I don't record the initial thing that way. With sound track albums, some that you buy sound excellent — beautiful frequency response, they're just great. Others sound like they were recorded through a telephone. The point is, I usually try to do the best job I can, from a sound standpoint, no matter whether it's for film or a record. But when it's transferred from tape to film, then you have to be sure that the limitations of the film aren't going to obliterate some key components of the sound that you're looking for. It's a good thing you mentioned this. Another thing an

engineer must always do, is realize that ultimately, he's not making a tape. He's making a record. And records have certain limitations, certain characteristics. Home phonographs and pickups have certain limitations, certain characteristics. Home phonographs and pickups especially have limitations that tape does not. He must realize that ultimately it will wind up on somebody's shitty phonograph with a half-worn needle, with speakers that are really not working too well . . . and be sure that what he records will be heard on these phonographs. This is the same concept one has to be aware of in recording for a movie. The film has a much shorter bandwidth. So, if you have a triangle, make sure it's a triangle that has low frequency components low enough that when it's transferred to film, that the natural cutoff of the system does not obliterate the fact that a triangle was hit. Those are the considerations that *I* take into account. The rest are taken into account by the producer, the musicians, the arranger, and the scorer.

MONITORING

When you mix, do you listen on big studio monitors and also on little car radio type speakers as well?

FC: When I mix, I try to mix at a comfortably low level — *not* comfortably loud.

Like the level that the average record buyer listens at?

FC: Right. Not the level that a rock'n'roll enthusiast might listen at. I listen at a normal level, which is somewhere around 70 dB. I mix on large speakers usually, which have a fairly decent frequency response. We have Altec LC1A's. The big reds. They sound very nice. I listen at a normal volume, and then, when I'm finished, I play it back again on the small speakers. We use Auratones, but you can use any halfway decent small speaker. Then I do play it loud, to hype myself, and to make sure I don't hear any dropouts, or hiss changes, and that my fade was good. I never, or very rarely, mix super-loud. At 100 or 105 dB, the ear cannot notice a 2 dB change as readily as if I was listening at 70 dB. The other theory being that if at a soft level, I can hear everything, then, when I play it loud, I *know* everything is there.

At loud levels, the highs and lows will appear to be there . . .

FC: Whether they're there or not. Not only that, I don't get as tired. The ears don't fatigue so fast . . . I have to consider that I'm going to be in that studio many years after these guys have come and gone. I make my living that way. The point is that if I have to play it loud, because some musicians have lost their hearing because of live concerts, and they don't seem to care, I walk out of the room. I'll get the tape all set, and I'll say, "When you're ready, just press this button." It doesn't offend them. They understand that by hearing well, I'm going to give them a good product.

I know that you've been experimenting with the Sennheiser Vocoder for possible use on an upcoming Herbie Hancock album. Have you been able to achieve any interesting effects with it?

FC: Yes. It's a fantastic device. For anyone who's not familiar with it and its potential, let me say that as a person who has been fortunate enough to try it (because they cost $15,000) that it's a fantastic tool. it allows the modulation of one source by another source. You can take a synthesizer, speak into a mike, and put it through the box while you play the synthesizer, and all the vowels and consonants that are spoken are transmitted to the synthesizer and cause the synthesizer to appear to be talking with your voice. It doesn't combine your voice with the synthesizer. It actually makes the synthesizer like your voice. All the character of your voice, all the color, all the inflection, all the vowels. The clever thing is that it can make *anything* do that. You could take the sound of a waterfall. By speaking into a microphone you could cause that waterfall to appear to be talking. It doesn't sound like your voice, but it has the same intonation. But it's still the waterfall. It doesn't have to be voice, either. It could be a drum. It could give a piano a snare sound. It would sound like a snare with a tone to it. Who knows what its possibilities will be, especially for a person like Herbie Hancock, who can't sing, but would like to, and he has so many things in his mind that he could do vocally. What he does is, he says the words to a song, plays it on the synthesizer, and his voice comes out but the synthesizer is the notes. So it doesn't matter if he sings in tune or not, because whatever note he plays, that's the note his voice has.

THE AUTOMATT

Why did you decide to open a studio like this in San Francisco? Is there enough business to support this kind of operation?

FC: Now I am speaking for David (Rubinson) because it's David's studio and we were at Wally Heider's, as everybody knows, in San Francisco at least, and I love it there, and I think it's a fantastic operation and everything else. The only problem was that we couldn't get the studio when we wanted. OK? They're obviously in business to book the studio to whoever brings in the top dollar. They will not give up a studio or make a studio available to us exclusively, they have to keep circulating.

Why is it necessary to have exclusive use of the studio?

FC: Because we have so much product, we have so much work, you see. Our point is we have maybe eight albums a year that we do, at least, and all these side things like promo spots, singles that are associated with the album, TV sound tracks for the groups we record. So we are in that studio I would say an average of five days a week but with some holidays and time off and whatever. We couldn't always get the studio we wanted because of the fact there are a lot of people waiting and if they forsake them for us, they risk losing that potential business. Plus there were certain things there that they didn't have that we wanted. Because as we take on bigger and bigger and more complex situations, and as the industry advances technologically, we would like to keep abreast of that and Wally Heider is being run by a big corporation — they are run by Filmways, not by Heider himself. I know Wally, I love him, and I could speak to him on a personal level and say, "Hey, Wally, you know we would like to have a Harmonizer," or, "Hey, we'd like to have automation," or, "How's about getting the studio re-acousticized by putting hardwood floors in." Now with the big company you have to do a whole number with a big corporation and then the Board sits down and they decide well is it really worth it? And so we found that we were getting at sort of odd ends with getting the kind of service we needed plus the studio availability time that when the opportunity came for David to open his own facility he jumped at it. The idea was also that we could work at our leisure when we wanted — not when we could get the studio — when we want.

Publically, the Automatt is for rental but right now in-house productions are . . .

FC: Keeping it busy almost day and night. But it is conceivable that if people like to work let's say 10 to 3 in the morning they could certainly rent it. The thing is that we don't want anybody to be handling the equipment because it's very sophisticated and very delicate in its own way. There are a lot of studios that open their doors to any engineer, anybody who claims to be an engineer, as long as they pay the bill at the studio rate . . . we really would prefer it to be like Warner Brothers — if you go to Warner Brothers in L.A. the charge is a minimum of $225 an hour for their facilities, and it's sort of a deterrent for the little guy who says, "Hey, I'm the roadie for this group and we'd like to do some demos and let's go in and do a demo and I'll engineer your record." You know, even if you have somebody watching over them, the studio takes the risk of super damage. And I've worked in the studios that had this open door policy, and invariably they're in really bad repair, and it has nothing do with the maintenance department — it's just that things go wrong that are intermittent or that are not absolute failures but fatigues in the electronics. And so all of a sudden you notice the board doesn't have the headroom it should. You notice that the limiters don't limit properly because there's been some damage to them. But not enough for anybody to spot as a failure unless they're really hot; so no trouble reports have been put in or there's so many other things to be fixed that the maintenance department has never gotten around to fixing module 13, which has no headroom anymore because the engineer that was in there was doing something wrong, he put the full line level into the mike input or something.

We don't want that to happen here. We'd like the place to stay in as cherry condition as possible and anybody who is going to pay the kind of money we're charging — we're not charging high because we want to charge high, we have a lot to offer here. But hopefully that when you're ready to pay that kind of money you will at least see to it that you get a qualified engineer to come in.

With equipment of this complexity it would take an awfully long time to familiarize somebody — even somebody who is a good engineer to come to a board like this cold.

FC: Well, there are some, but I'd say by the end of the year there will be quite a few engineers who will understand boards like this, as they're selling like hotcakes it seems. It's the first successful board with automation.

You're completely satisfied with its capabilities.

FC: I am satisfied in the state it's in, but like anything else the moment you create a proto-type, it's obsolete. Already we have ideas of other things we want done and certainly Michael is very hot on this kind of thing. And already it's obsolete, so in that way I'm not satisfied, but I'm more satisfied than I'd be with anything else I've had my hands on.

When the other studio in town (Different Fur) *got their automated board didn't it destroy your exclusivity?*

FC: Nobody is going to destroy exclusivities. Because nobody has the personnel we have. Nobody has the knowledge and background we have and I don't say that out of conceit, I say that out of sheer experience. It's like anything else — it's a lot like drummers. You get these drummers — these young kids who idolize a particular drummer so they go through hook or crook and buy the most expensive set of drums — just like their ideal drummer. And then they'll wonder why in the studio they don't sound like whoever the hell it is they idolize. They say after all I have the same drum, I play the same sticks, I've been practicing his licks, what do you mean, I can't sound like that? So it must be the want it at their disposal should they feel they might like to toy around with it. And this is what makes it so difficult to own a studio. Time was when equipment would cost $300-$400 a unit — whatever it is — a little limiter or equalizer, you could pay $400-$500, you could have one. Now each new device is costing in the thousands and in order for them to be competitive, studios have to stock all this *chazzerai*, this garbage, this equipment on the chance that somebody, the Grateful Dead, Santana or Sly·Stone, might book the studio, after all they're the only ones in town with a flanger or a phaser or whatever the hell it is that is currently the craze. And this makes it very difficult.

So I am sure studios are doing this and

form without all the fancy stuff that's been put in — it's a very basic board. It's just a volume control, a meter to read your levels, a key to turn it on and off and an assign button. And something to send to the cue system with no EQ, something to return the echo with if you want, but basically it's a very simple board and it's only when you start calling on it to do all this craziness that it could get confusing.

Don't get confused that we got the automation because it makes mixing easier, but what the advantage of it is, is that two days from now or a month from now the artist calls us and says, "I'm going to appear on the Johnny Carson Show and we have to have a copy of the track. The track only. Just like in the album. You know, with the crazy effects we put on and the horse galloping across from left to right and the synthesizer part that we took five hours to do, can we get that just like it is on the record, only without voices?" Before we used to have to come in and remember all the cues and get out the album and play it and very carefully A-B it and try to get it right. Now we just come in, get the programmer set, de-activate the vocal tracks so that the programmer doesn't sense those, it only senses everything else. We get a perfect mix of the track.

These kinds of things are what we like because usually the creative process is a one-time thing. I mean, you do a mix, just like a musician that plays a solo and he knows it, the band knows it, the producer knows it. It takes him a long time to get to that creative moment. He knows that he had it once, it's difficult to get it again. If anything, you may get a different great solo. But you'll never get the same one. It's the same with mixing. Mixing has it's creative aspects nowadays. A lot of trickery and a lot of manipulating and creative decisions are made. It's terrible for the guy to call you up at three in the morning and say, "Hey, man, we've been playing the tape all night and, ya' know, it's out of sight, only we forgot that guitar lick that Richie played and he won't let us put the record out unless that guitar lick is really up and I don't think it's really high enough. I know when he hears it he's going to bitch. So we're going to have to go in tomorrow and mix it again." Invariably it has happened in the past, you come in and you don't give your all because you know the mix was right and you know they're just kidding themselves, but to make them happy, you're going to have to do it, so you don't put out as much because you already did your best. Automation avoids all that.

ENGINEERING vs. PRODUCING

Do you ever do any producing, or do you just consider yourself an engineer?

FC: I consider myself primarily an engineer. I have done production; I usually do co-producing, because I don't consider myself a producer. A producer is a whole different animal, and it requires a different kind of personality and priorities. But I have co-produced records. Whenever a group basically has what they want and need but they don't know how to transfer it properly and to coordinate everything to make it a record, they don't really need a producer because they sort of produce themselves, but they can't. What they need is the liaison between what they know they want but don't know how to get and somebody who is going to help them get it, and maybe make suggestions based on experience. That's what a co-producer is. I would not tackle, for instance, producing a group who needs rehearsing and who needs material picked, which is what a real producer does. He sometimes goes so far as to hire other musicians to play in place of the band members, because they're not good enough.

Was that the case on the Santana Abraxas *album, and Cold Blood's* Sisyphus *album?*

FC: Yes, that was co-production. Cold Blood has a producer, and if they listed me as co-producer it was only because I contributed so much toward the concept and how to get the sounds. For instance, they know their song, but they have no idea of how it's going to sound on a record or how it should sound on a record. Then, if I get involved, and I like the stuff, and I feel that I can contribute something, then I begin to make suggestions. Why don't we this, or that, or I don't think you should do this, or instead of piano you should play harpsichord, or we ought to take the verse out. Now I'm starting to co-produce, because I'm actually making creative contributions to the ultimate product. If an engineer calls himself a creative person, so is every musician. That doesn't make him a producer. The fact that you do your job well, and get involved, does not mean that you deserve production credit. Production credit, in my mind, is when you actually create a concept, develop a concept, do more than just engineer a record.

Surely you've worked with producers whom you felt made important and perhaps indespensible contributions, and some where the engineer could have handled everything the producer did . . . maybe the producer was even in the way.

FC: Exactly. I've worked with producers who actually knew less about producing a record than any one of the musicians out there, and myself. He just sits there, reads a newspaper, gets high, tells the band to play, has them come in. They listen to the stuff, he asks what they think, and they go, "Well, maybe we ought to do another one", and the guy says, "OK, go ahead". Then, when it's all over, he says, "OK, next tune", and really contributes very little other than doing the paperwork — he writes down the time of the tunes, the names, fills out the musician's union forms, and pays the bill. But aside from that, he's not a producer . . . if he's going to take all that money that producers get, not only counting front money, but royalties as well, points, he should do more than sit there and say, "Well, what do you guys think?" I can do that, I do it all the time, but I don't expect to get producer credit for something like that.

Obviously, because you work with David Rubinson so much, you must think very highly of him.

FC: He's an excellent producer. He knows what he wants. He knows exactly the concept, he knows exactly what he expects that record to sound like, and he knows how to communicate with the musician and the engineer.

Do you have any other favorite producers?

FC: I had a few. It's been a long time since I've worked with any other heavy producers than David, because David has so much work. He keeps me busy, so I don't have very many opportunities to work with outside producers very much. I did like Jimmy Guercio, when I worked with him (Blood, Sweat, and Tears, and Chicago). I worked with Al Kooper, who was kind of strange, but he was talented as a producer.

At a Patti Labelle overdub session, string arranger AL BENT, DAVID RUBINSON, FRED CATERO.

When you're recording basic tracks, do you use a lot of baffling to keep everything separated?

FC: Everything you record is dependent on the studio you record it in, the kind of material, the instrumentation, and what the ultimate concept of the record is going to be. If you have a studio that has a basically bad sound, (by that I mean the room), understand that most studios today don't sound good because of all the screens, baffles, carpeting, and padding. In the past, studios used to be larger rooms than they are today, and they used to have a sound of their own, like a concert hall does. When you got leakage, it actually enhanced the sound. It was something you would look forward to. Musicians would like to work in a particular studio because their instruments sounded very good in the room, and if you got some leakage, it only helped the sound — it didn't hurt it. But today, because everybody's so hung up on isolation, studios are being designed deader and deader, so that the instruments sound as though they're being played in a mattress-walled room, or an anechoic chamber. The room itself offers no coloration to the instruments, so the instruments that usually rely on the room to help them resonate, don't. So, if you get leakage, it makes the instrument sound worse than if you didn't have it, so you're forced to put baffles up. Also, you put baffles up so you can kill the guitars and re-do the piano, or whatever it is. But it's very difficult getting good sounds in that situation.

In our studio, the Automatt, we're very lucky that the room that we use has hard floors and a high ceiling. It's semi-damped. It's not very live, it doesn't have an echo to it, but it's not dead either. It's a good sounding room even though it's small, because of the high ceiling. Most instruments, even electric ones, sound very good in it. So I try to use a little bit of leakage whenever possible, to make it sound like it's not going direct. So I try to establish ahead of time what are we going to keep, and what are we maybe not going to keep. If we're not going to keep the guitar, then I have to isolate the guitar from everything else. But if I know that the guitar is going to be used, then I'm not so critical about the guitar leaking into some of the other mikes, because it sounds better — it has a bigger sound. I don't automatically use screens.

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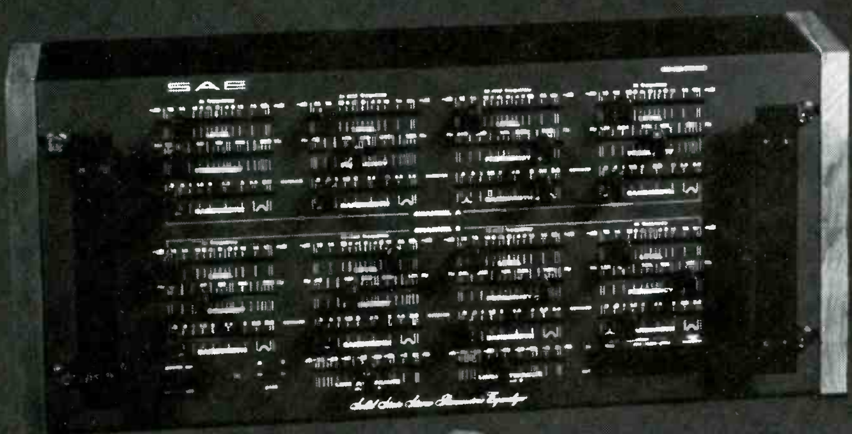

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MASTERING TAPE COMPATIBILITY

by JOHN ROBINSON
3M COMPANY

Demands from recordists, producers, artists, listeners for ever higher quality audio reproduction have spurred tape manufacturers to develop mastering tapes with heretofore unequalled recording capabilities. However, these developments have not come without introducing new concerns to tape users. One such concern, the incompatibility of different mastering tapes, is a subject worthy of attention. First, however, some background on audio tape development.

While studio engineers, mixers and producers may scoff at the sound quality of the audio cassette, its move into the high fidelity arena helped launch a dramatic period of magnetic tape development. The slow speed (1-$\frac{7}{8}$ ips) and the narrow trackwidth (20 mils) of the cassette system meant that to achieve reasonable quality, substantial improvements in tape formulations had to be made. From the laboratories came a multitude of new tapes: high energy, chromium dioxide, ferrichrome, cobalt-encapsulated, and others. Many of the technological developments in ferric oxides helped lead the way to the development of some of the new studio mastering tapes.

—The Author—

JOHN ROBINSON joined 3M's Magnetic Audio/Video Technical Service lab in 1972. He earned his E.E. degree from Southern University in Baton Rouge, Louisiana. In addition to his engineering background, John is an accomplished musician.

More signal level-output, less biased or background tape noise, less print (layer-to-layer signal transfer) and less harmonic distortion (harmonic of the fundamental frequency expressed in percentage of difference between original and produced sound) are all desirable or even essential for improved mastering tapes.

To achieve these, and other improvements, the tape manufacturers must deal with some very basic elements in the design of magnetic tapes. Two such elements are *coercivity* and *remanence*. There has been a general trend toward higher coercivity (magnetically harder particles) and increased remanence (thicker oxide coatings) in recording tapes.

COERCIVITY

The harder the magnetic particles the more difficult it is to magnetize and demagnetize them. As tape moves past the recording head, the head's magnetizing force penetrates the oxide coating to align the particles in a representative pattern of the applied signal. Once past the head there should be no more force acting on the particles. However, higher frequencies, which have more closely spaced patterns than lower frequencies, cannot be moved past the head fast enough to clear the weaker trailing edge of the magnetizing force. Higher coercive oxides, because they are not so easily magnetized, resist more strongly demagnetization once past the head. For reference purposes, Scotch #111 tape has a coercivity of 270 oersteds; Scotch #206, 320 oersteds; and Scotch #250, 380 oersteds.

As mentioned earlier, increased remanence is achieved with heavier oxide coatings (more magnetic material per unit area). Remanence, measured in lines per quarter inch, is the induced flux remaining in a given tape $\frac{1}{4}''$ wide after a longitudinally applied field is reduced in intensity from 1,000 oersteds to zero using a 60 Hz dynamic B-H vs H hysteresis loop tracer. For example, Scotch #206 has a remanence of .93 lines/qtr inch; Scotch #250, 1.25 lines/qtr inch. With this increased coating weight comes greater *undistorted output*, less *harmonic distortion*. Greater output level is possible because of the heavier concentration of magnetic material.

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remanence are required to achieve maximum signal-to-noise ratio without sacrificing other essential tape performance features.

OXIDE AND MANUFACTURING

All oxides have their own properties, such as coercivity, noise and print. But the inherent properties of oxides are not all that determine how a tape will perform. Of equal importance are the various processes in manufacturing — from the milling of raw oxides to produce the ideal particle to the final surface treating to produce the ideal tape surface.

The improvements in output, biased tape noise, lower harmonic distortion, and in other performance parameters have exposed some equally important potential tape problems that relate directly to the manufacturer's processes. Some of the important problems are *low frequency modulation noise, transient response* and *intermodulation distortion*. While each of these phenomenon could be the subject of a separate paper, a brief explanation of each will aid the reader's understanding of this paper.

LOW FREQUENCY MODULATED NOISE is generated (modulated) only in the presence of a signal and is proportional to signal amplitude. Its cause is attributed to non-uniformity of the oxide coating. Primarily audible when generated by lower frequencies, modulation noise is a fuzziness in the sound reproduction.

TRANSIENT RESPONSE is a function of the tape's saturation or overload characteristics. It is the ability of a tape to record strong, harmonically rich signals, which may exceed the normal operating level, without excessive audible distortion. Percussion instruments, such as the drum, produce transient or fleeting signals.

INTERMODULATION DISTORTION exists when more than one frequency is recorded on tape. It is composed of the sum and differences of those frequencies and is much more displeasing to the ear than harmonic distortion.

SUBJECTIVE TERMINOLOGY

Further challenging to tape manufacturers is the subjective nature of recordists' ears. Recordists often attempt in technical terms to explain why a particular tape "sounds" good. But the technical data sheets and the subjective sound quality of tapes often appear unrelated. It is fact that two different mastering tapes can have nearly identical technical specifications yet exhibit substantial differences in sound quality. While there is a body of terms used to describe "good" and "bad" sound reproductions, many or even most terms used in studios by engineers, producers and artists are purely subjective. One man's "clean" is another's "transparent"; one's "mellow", another's "rich". And the list could go on-and-on.

The point is, because these are all terms that refer to a subjective quality and because two engineers seldom use the same term to describe any one single quality, it is difficult to use technical data to express the advantages of one tape over any other. In many cases there simply are no technical terms that accurately define the subjective quality.

The objective of all recording is, of course, to reproduce sound as exactly as possible. The type and quality of tape used are obviously important contributions to the end recorded product.

Designing and manufacturing the ideal tape are difficult charters. Such a tape must embody, simply, all those elements considered important by recordists with as few compromises as possible.

THE PROBLEM OF COMPATIBILITY

All tape suppliers attempt within the confines of their technical and manufacturing capabilities to achieve the ideal balance of features that tape users demand. These efforts have led to the development of a number of mastering tapes.

Ideally, of course, each tape should be set up on a machine before any session to achieve the optimum performance from the tape to be used during that session. Before each outside client arrives for a session, the studio engineers must optimize for that client's tape. With the great variety of tapes that exist, this practice can be quite time consuming. Consider the unfortunate engineer who is confronted with a client for an overdub session with several different types of tape on one reel. Imagine having to reoptimize each track of each machine between every cut! While that extreme condition will not recur often, it could in fact happen. How unlike the days when Scotch #111 was a standard reference. But that's progress.

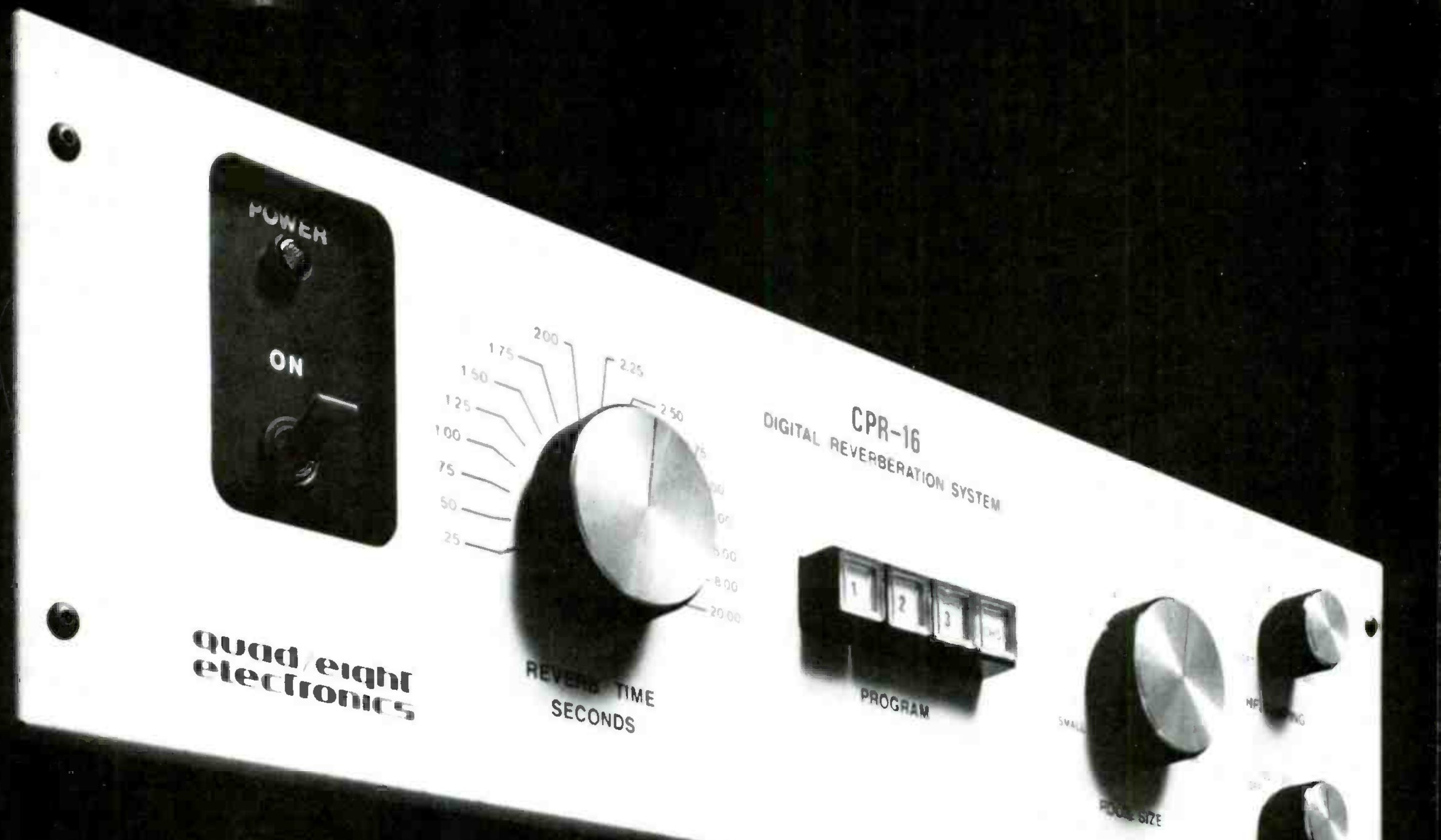
The discussion of tape compatibility continues, however, and is not likely to diminish. What with standard oxides, high output, low noise, low print and various combinations thereof, there is little wonder that confusion over tape types and their compatibility exists.

Let's examine more closely the considerations the studio engineer must make concerning tape/recorder set up and then discuss a suggested solution to the compatibility problem.

Total compatibility would exist if there were no need to change either bias, equalization or level when switching from one tape to another.

Bias and equalization are the two most essential elements in tape compatibility. Both have tremendous effect on other tape properties such as distortion, sensitivity and print level — all of which differ greatly from tape to tape.

The proper bias condition of a tape is a compromise point. Some tape properties improve as the bias current level is increased, others become worse. Optimum bias is a point where the losses equal the



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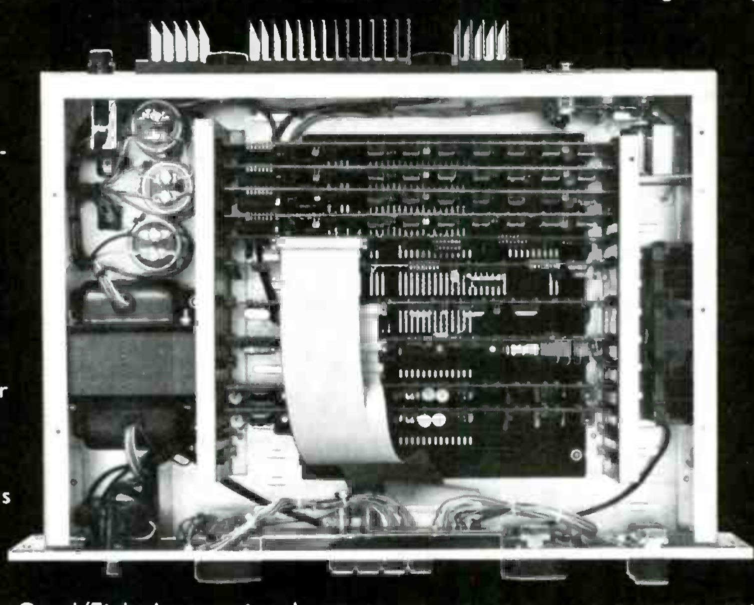

The CPR contains two individual reverb programs and one "open" program provision for future custom sound processing effects. Program 1 is analogous to a live acoustical chamber, and Program 2 creates a simulation of an artificial plate. Program 3 switching facility is pre-wired for a plug-in programming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

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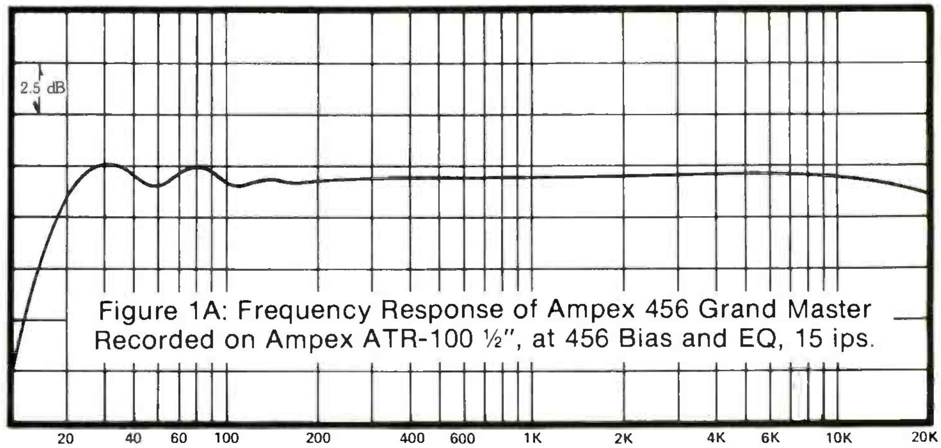


Figure 1A: Frequency Response of Ampex 456 Grand Master Recorded on Ampex ATR-100 ½", at 456 Bias and EQ, 15 ips.

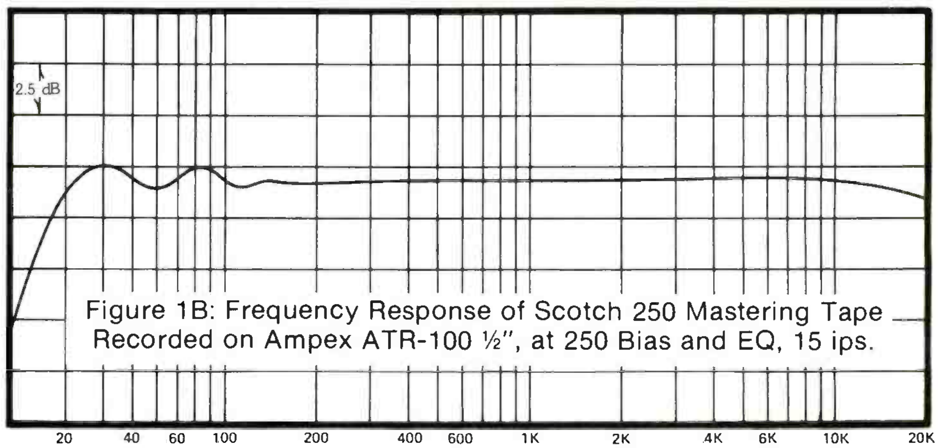


Figure 1B: Frequency Response of Scotch 250 Mastering Tape Recorded on Ampex ATR-100 ½", at 250 Bias and EQ, 15 ips.

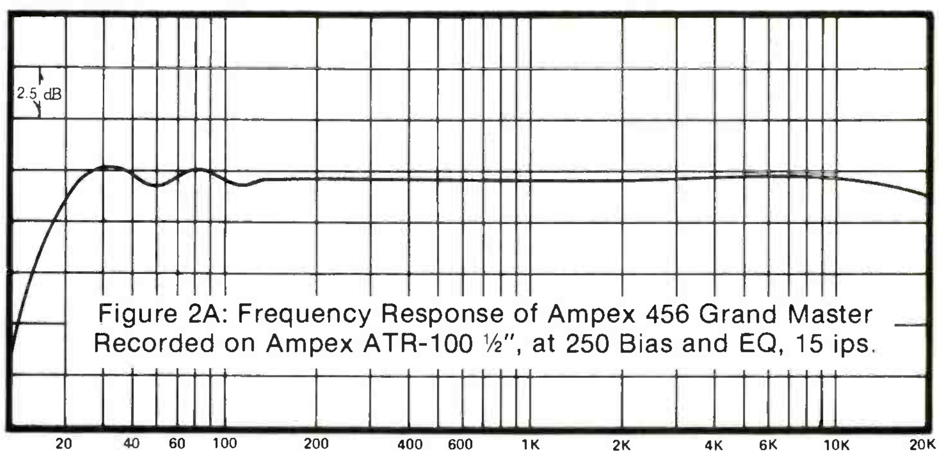


Figure 2A: Frequency Response of Ampex 456 Grand Master Recorded on Ampex ATR-100 ½", at 250 Bias and EQ, 15 ips.

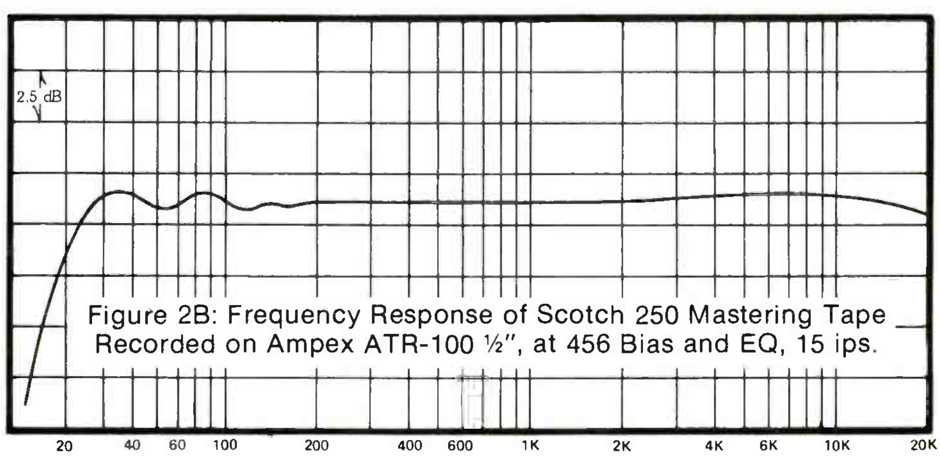


Figure 2B: Frequency Response of Scotch 250 Mastering Tape Recorded on Ampex ATR-100 ½", at 456 Bias and EQ, 15 ips.

gains, where what is sacrificed is offset by what is improved. As mentioned earlier, bias and equalization affect other aspects of tape performance. In addition, bias has considerable effect on equalization. As the bias current is raised from, say, zero, the sensitivity of the tape rises rapidly until a point where maximum output is reached, after which bias erasure becomes predominate at high frequencies and output begins to drop. Additional increases in bias will cause bias erasure to affect lower and lower frequencies until even the lowest frequencies will begin to lose output.

The ability to equalize — increase or decrease isolated segments of the frequency spectrum — can be used to counter the bias caused losses. The counter effect of bias and equalization can be employed to gain acceptable results from all tapes without setting up for each one individually.

The two most commonly used mastering tapes are Ampex' Grand Master 456 and 3M's Scotch 250. Both are capable of being operated at +6 dB (370 nW/m) operating level. By specification and practice their optimum requirements are quite different. Figure 1 shows the frequency response of the tapes recorded on a ½" Ampex ATR-100 machine at 15 ips. Figure 2A shows the frequency response of Ampex 456 recorded at Scotch 250 EQ and bias settings. Figure 2B shows the frequency response of Scotch 250 recorded at Ampex 456 EQ and bias.

The biasing method used for both tapes was to increase the bias level while recording a 1.5 mil wavelength signal until the reproduced signal drops to 2.5 dB below maximum. The same biasing technique should yield very similar results on all other audio tape machines now in use. Biasing at other speeds can be done using the same method, keeping in mind that a modulating signal of 5 kHz is necessary to provide the 1.5 mil wavelength at 7.5 ips and 20 kHz at 30 ips.

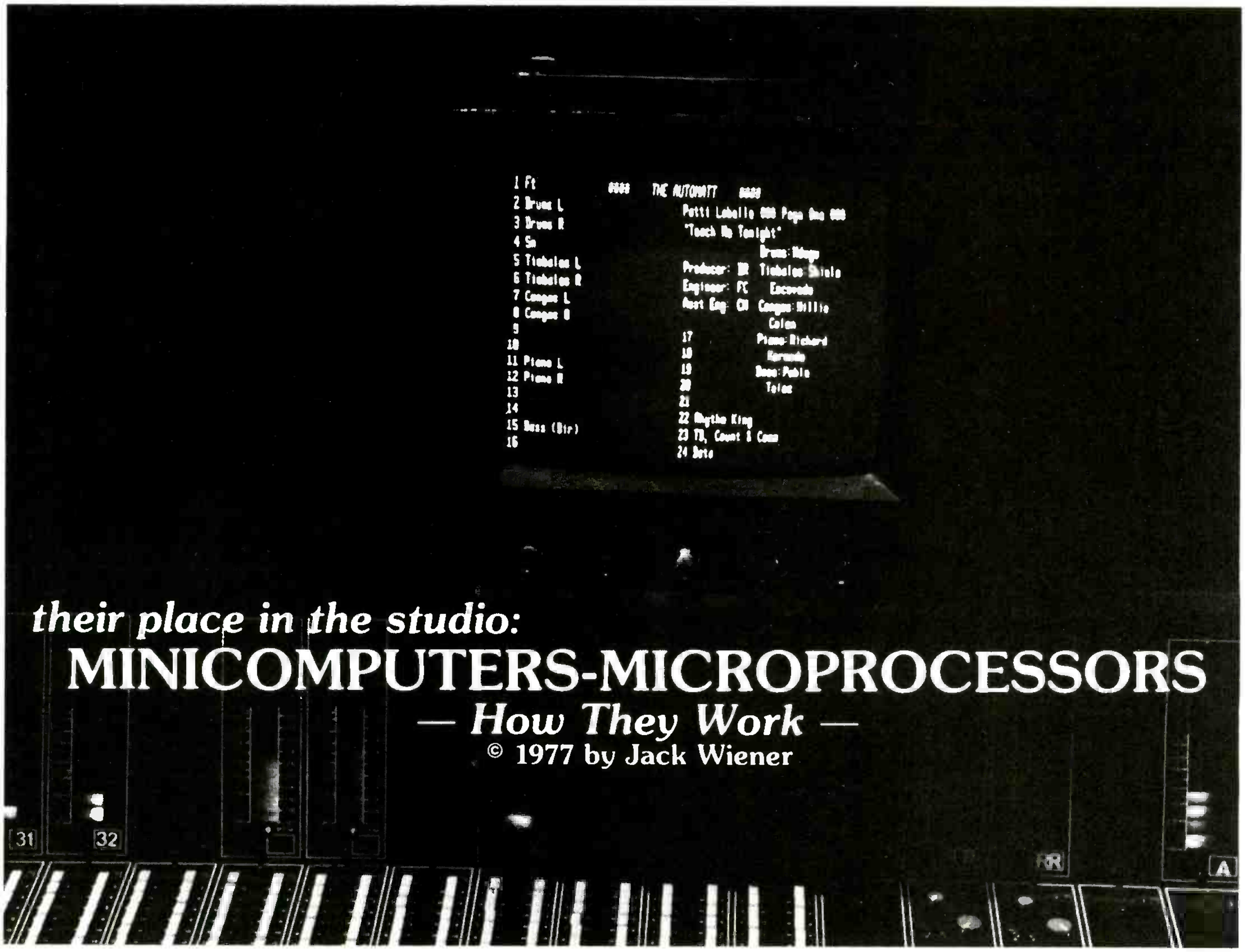
A table (Figure 3) illustrates the effect of bias on other electro-magnetic properties which include maximum output at low frequency, third order harmonic level at reference output, biased tape noise level. By use of the response curves and the table a compromise bias current can be derived which will make the two tapes interchangeable.

Figure 3: EFFECT OF BIAS ON ELECTRO-MAGNETIC PROPERTIES

250 BIAS & EQ		*456 BIAS & EQ*		
250	**456**	**250**	**456**	
0 dB	**+.3 dB**	**-1.3 dB**	**0 dB**	**1K Sensitivity**
0 dB	**+.2 dB**	**-.7 dB**	**0 dB**	**10K Sensitivity**
0 dB	**-.5 dB**	**-1.3 dB**	**0 dB**	**15K Sensitivity**
- - - - - Less Than .5% - - - - -				**% Harmonic Distortion @ 250 nWb/m**
- - - - - Less Than 1% - - - - -				**% Harmonic Distortion @ 370 nWb/m**
+10.6 dB	**+10.7 dB**	**+8.2 dB**	**+11.5 dB**	**Output at 3% Third Harmonic Distortion**
-63.2 dB	**-61.2 dB**	**-62.9 dB**	**-61.0 dB**	**NAB Weighted Noise**

CONCLUSION

Once again, optimum performance can only be achieved by exact alignment of equipment to each particular tape. In situations where, because of time limitations or other factors, it is impossible to optimize for each tape, a compromise bias setting will produce satisfactory results. Though the curves and the table account for conditions at only two specific bias and EQ settings, a condition somewhere between the two would even further minimize the incompatibility.



their place in the studio:
MINICOMPUTERS-MICROPROCESSORS
— How They Work —
© 1977 by Jack Wiener

The recording industry as we know it is about to undergo a dramatic change. Within the next few years much of the equipment with which we have grown familiar may well become obsolete.

The primary cause of this change will be a tiny integrated circuit called the Microprocessor. These devices have been with us for several years, but only in dedicated applications. Individual pieces of equipment have been designed around them to be built and sold at incredible prices. When the Microprocessor is combined with its counterpart Memory Chip they form what has to be called the original *magic black box* . . . a Microcomputer. When the memory is loaded with the proper *program* it can, for example, at one instant look like a multi-machine SMPTE interlock and the next instant turn into a 100+ dB per octave band pass filter. Moments later it can control the cueing of tape machines or handle the business billing.

These sophisticated capabilities are made more meaningfull when we add the newest element, which is also the oldest requirement in the world . . . *cost*. These capabilities can now be acquired for a price many of us can afford. In the last few months we have seen the retail cost of these exotic microprocessor chips, even in single quantities, drop to less than $18.00, and they are available in thousands of stores throughout the country. We can expect these already low prices to drop still lower as chip manufacturers compete for their share of the burgeoning home entertainment and hobby markets.

Sensing the growing fertility of this new market, computer *progrmmers* have started *writing* for these small machines in the hope that their programs will become hits and enjoy multiple sales even though they will be sold at a low price.

Recording Engineer/Producer trys to stay abreast of everything happening in our field. To that end I have been asked to prepare several articles on Microcomputers, and how they can enter audio's day-to-day operations.

This author has seen many changes and evolutions in both recording equipment and techniques. From vacuum tubes to solid state, from ribbon microphones to condenser microphones, from full track overdubbing to multi-track and multi-machine interlocks. Each change in its time seemed impressive and possibly a threat to certain talents that we had acquired and were quite proud of. But this latest evolution, the Microcomputer, is destined to find its way into every single corner of the recording studio and sound re-inforcement industry. This small unit, capable of being built on a single circuit board, will inevitably move-in and take over functions which many of us consider as sacred to our recording talents. There will be some who try to resist the change to Microcomputers. I am not a Crusader, it is not my intention to lure them into the fold. But perhaps a few words of personal experience might be in order.

As one of those people who resisted the change to multi-track recording, feeling that it took the personal touch away from the mix, I couldn't see spending vast sums of money for a multi-track machine when the

---

—The Author—

J.S.(Jack)Wiener is a native of Chicago, where he began his recording career in the mid-fifties mixing and/or mastering top groups for labels such as Mercury, Chess, Atlantic, Imperial, Sun, Playboy and others. He was president of Sheldon Recording Studios (later to become Chess Records' studios).

His recent interest in Motion Picture and TV Audio led him through positions such as Sound Department Head of Reid Ray Films to the formation of his own A/V industry consultant firm, now specializing in computer automation of these industries.

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job could be done just as well on full-track recorders. Now we see our industry, one generation later, matured, totally adjusted to the multi-track concept. There has been a tremendous change in procedures. The final balance is no longer the prime concern on a tracking session. Now it is a matter of capturing every element and making sure it is on a track. But perhaps this is as it should be. With the more complex productions we are mixing today perhaps no one person is capable of sitting at a console and catching every single cue exactly the way the artist wants it, and establishing every balance just the way the producer thinks it should be.

In reality, the change to multi-track recording didn't really take anything away from the mixer, it simply re-adjusted the order of doing things. It postponed the decision on the final balance until all the information was recorded. Undoubtedly in today's environment it is a better way of doing things.

If you have ever missed a cue or brought in the horns a couple of bars late, blowing an otherwise perfect take, you know that there must be room for a computerized assist in doing your job well. And computerized mixing is possibly only the narrowest portion of the involvement you can expect from these microprocessors.

But, how do all these things happen? What goes on inside that magic little box to make it such a versatile machine? That's the subject that we will deal with in this article. We won't get involved with the why's and wherefore's of computer design or high level language or even binary language.

The term *computer* generally relates to the big IBM style machines, built up of racks and racks of equipment. Making up that giant computer are Memory Banks, Card Readers, Magnetic Disc files, Tape Punches and Readers along with other Input-Output devices, and a thing called the *Central Processing Unit*. The CPU itself may fill an entire rack with discrete components or may be a series of circuit boards built up of ICs.

MICROCOMPUTERS

The Microcomputer, a miniaturized version of the big computer, gets its name from the Micro-Processor around which it is built. This is a rather recent development in the art of integrated circuit construction. The Microprocessor, on a single chip less ¼ square inch, performs the same functions as the central processing unit (CPU) in the big computer.

Just for the record, to clear up one other term that's been floating around, we have the Mini-Computer, which fits in between the Micro and the Maxi or full computer. This unit is generally based on a set of processor chips. Each of these three levels of machinery, and all their variations currently being sold offer different degrees of versatility. We will be concerned with only one type machine, the *Microcomputer.*

WHAT MAKES A COMPUTER WORK?

A computer responds to a *program*, which is a series of instruction steps.

The instruction steps are stored in *memory*, which is another vital part of the system. It is helpful to picture memory as a tall stack of storage areas. Each area holding a numeric value. Our program travels through the memory system practically from start to finish until it reads an instruction to perform one of its tricks. Each instruction tells the processor to do one and only one thing at a time.

The task of manipulating data occurs in this processor section. The memory of a computer stores the results of the continuing chain of computations induced by the successive program steps. Again, computers process data one step at a time. This is an important key. One and only one thing is happening at any given time inside the computer.

The processor *controls* what is happening. It is itself directed by either the input, (which in many cases is a typewriter style keyboard, or a tape recording,) or it may be controlled by an internal program.

THE PROGRAMMER

Programs are written by someone with a combined knowledge of the intended application and the abilities of a given computer system. The program is a step-by-step series of *instructions*. Each one telling the processor what type of action to take, one step at a time. Finally an output instruction tells you that the machine has the result you are interested in. The output may go to a *printer* (computerese for a remote controlled unit similar to a typewriter), or to a CRT monitor (Cathode Ray Tube display unit. Very much like a TV set), or even to some mass storage device like a tape recorder. In some cases the computer output will control the actions of machines . . . send them forward, reverse, stop, or play. The important thing to remember is that everything happens *one step at a time.*

Let's look at some of the instructions of one of the most popular of these low priced chips, the 8080 Microprocessor. INR A is the *mnemonic* for Increment the A register. (Mnemonic, pronounced ni-mon-ik. Ni, as in nickel; mon, as in monitor; and ik, as in lick.) A mnemonic is an abbreviated word description of the action that will be taken by the processor, in response to the instruction, which is received as an electrical control pattern. Visualize the processor as having 78 work stations. Each instruction code directs one and only one work station to perform its given task.

REGISTERS

The chip has seven useable registers. Each can be thought of as a holding area for any real number from zero to 255 decimal. The INR A instruction takes the value in the A register, adds 1 to it, and returns the new total back to A.

The CPI instruction (compare immediate) is a kind of subtraction . . . but it does not change the value in A. It is called a *two byte* instruction because it occupies two adjacent locations in memory. The first byte (*byte* is another word for the information held in one memory location) is the coded CPI instruction to the microprocessor, which copies the value from the next memory location and subtracts this value from a value equal to A. Although it does not save the result of the arithmetic it performs a valuable service. If the result of the computation is zero a flag called the *zero flag* is *set*.

With the aid of cartoon drawings lets go through a little how-to-do-it session to get a handle on how these microcomputers work. We will use the following instructions:

INR A — Increment the A register.

CPI N — Compare A with N (N being any number between 0-255).

JNZ XX — Jump on not zero. This conditional jump instruction directs us to

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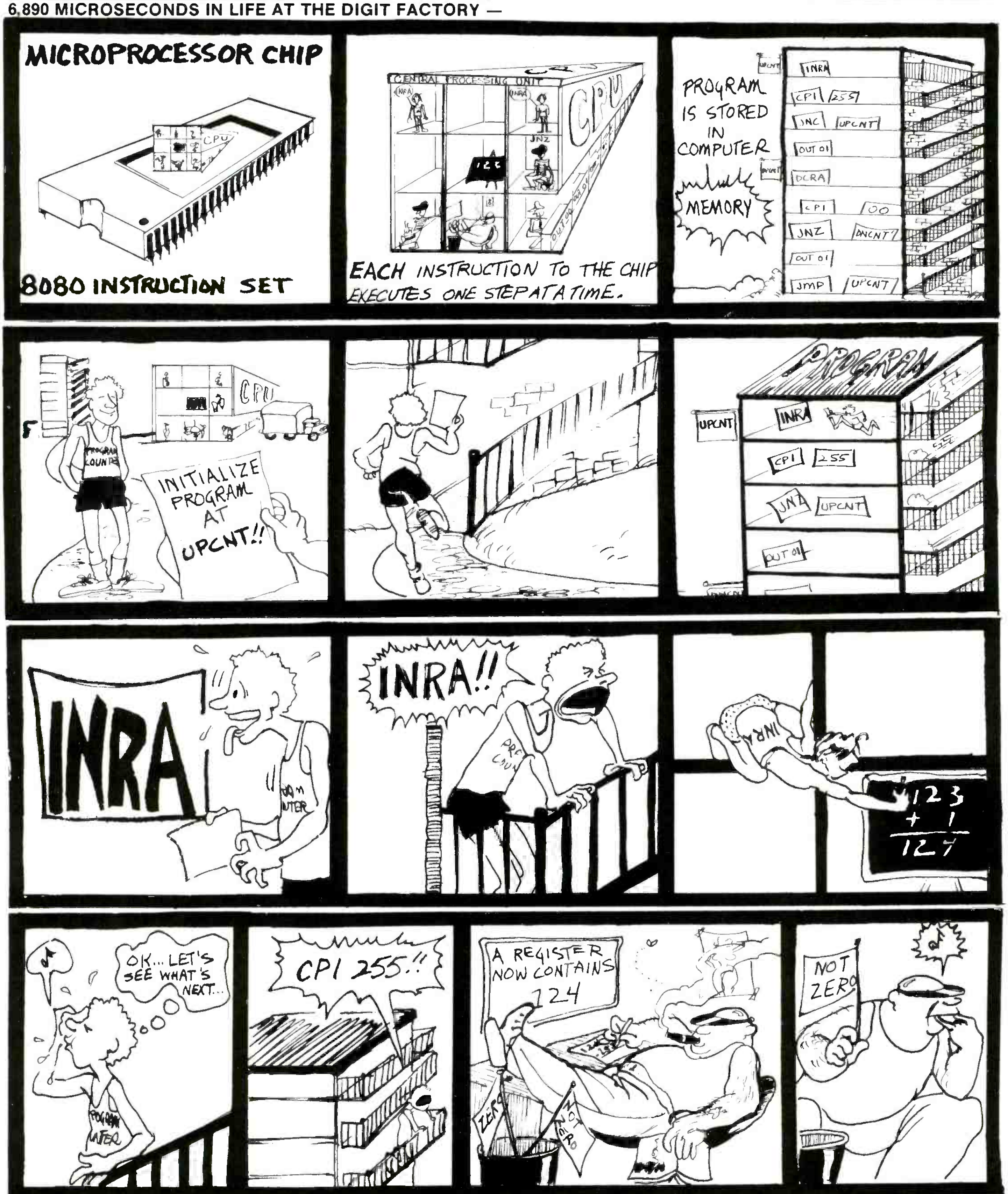

. . . continued on page 48

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stop our orderly procession through memory and execute the instruction contained in memory location XX. *But* to do so *only* if the *zero flag* is not set. If the zero flag is set we ignore this instruction and proceeed to the next instruction in memory.

OUT 01 — Output the A register contents to port 01.

DCR A — Decrement the A register.

JMP XX — Jump—unconditionally—to execute the instruction contained in memory location XX.

With these few instructions we will turn our microcomputer into an *audio frequency square wave generator.*

We initialize our program by re-setting the computer which sets the program counter to the start of memory. When we hit *run* we execute our first instruction which is INR A. Regardless of the contents of A we add 1 to it and store the new total back in A. Having completed this step we immediately

. . . continued on page 52

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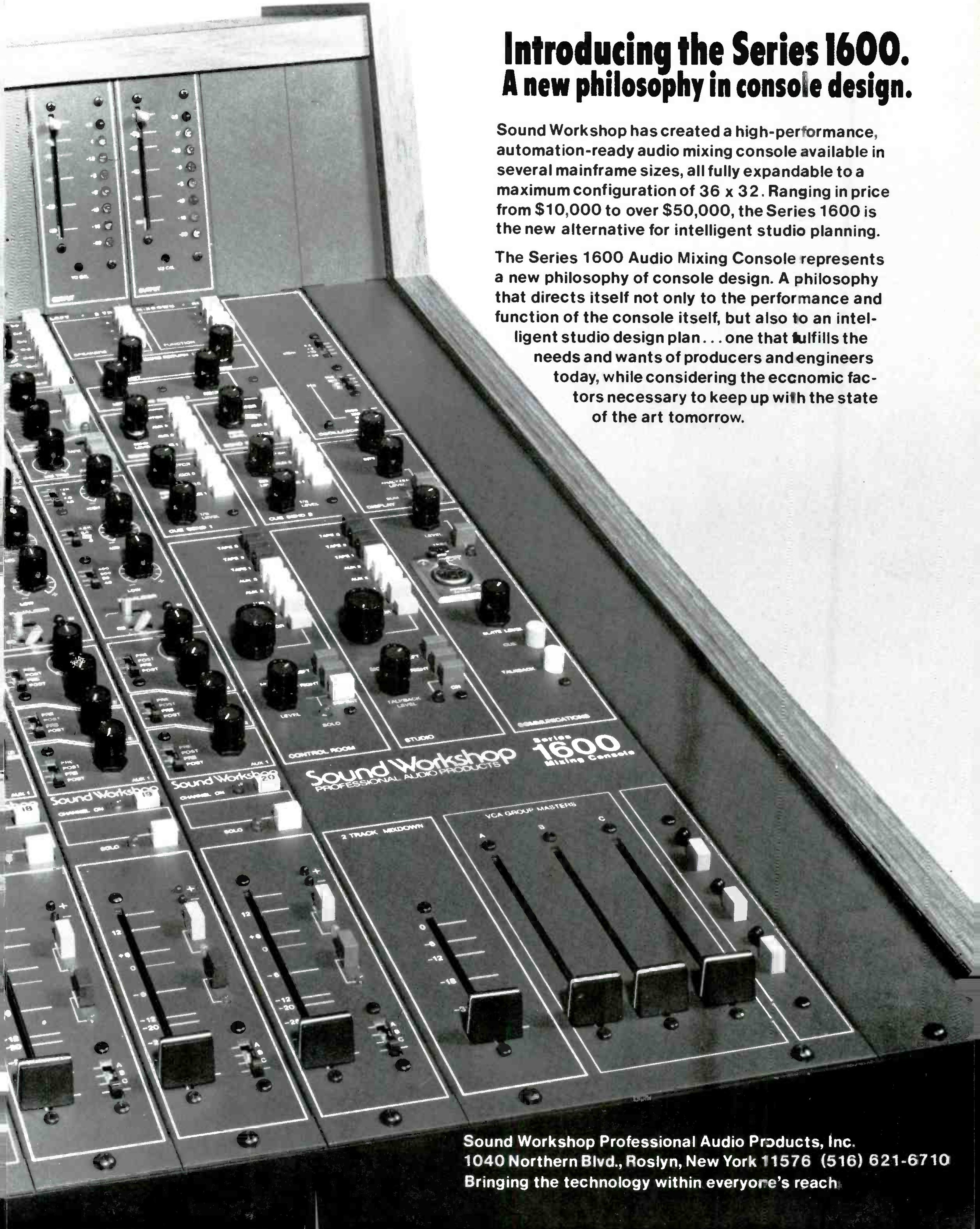

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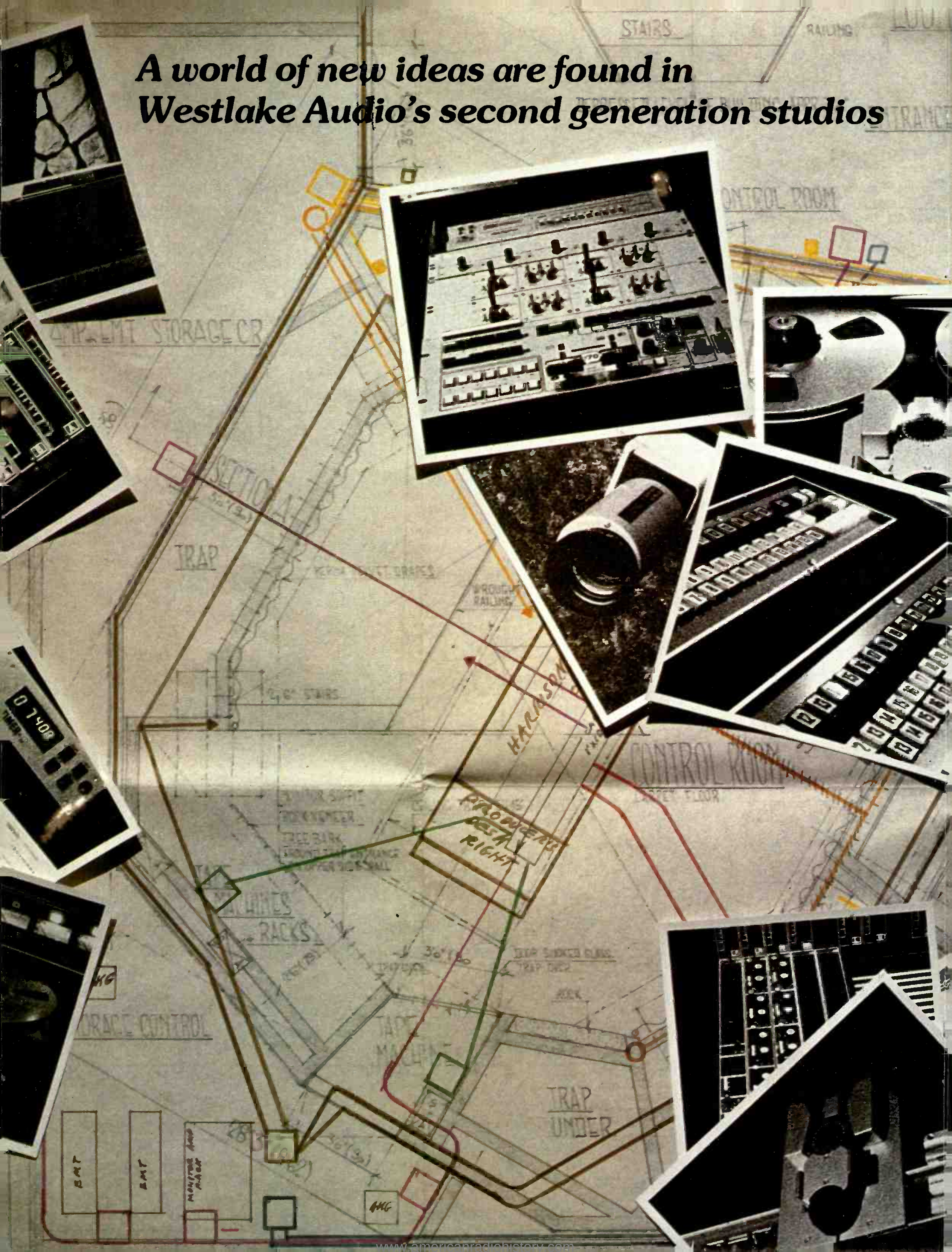
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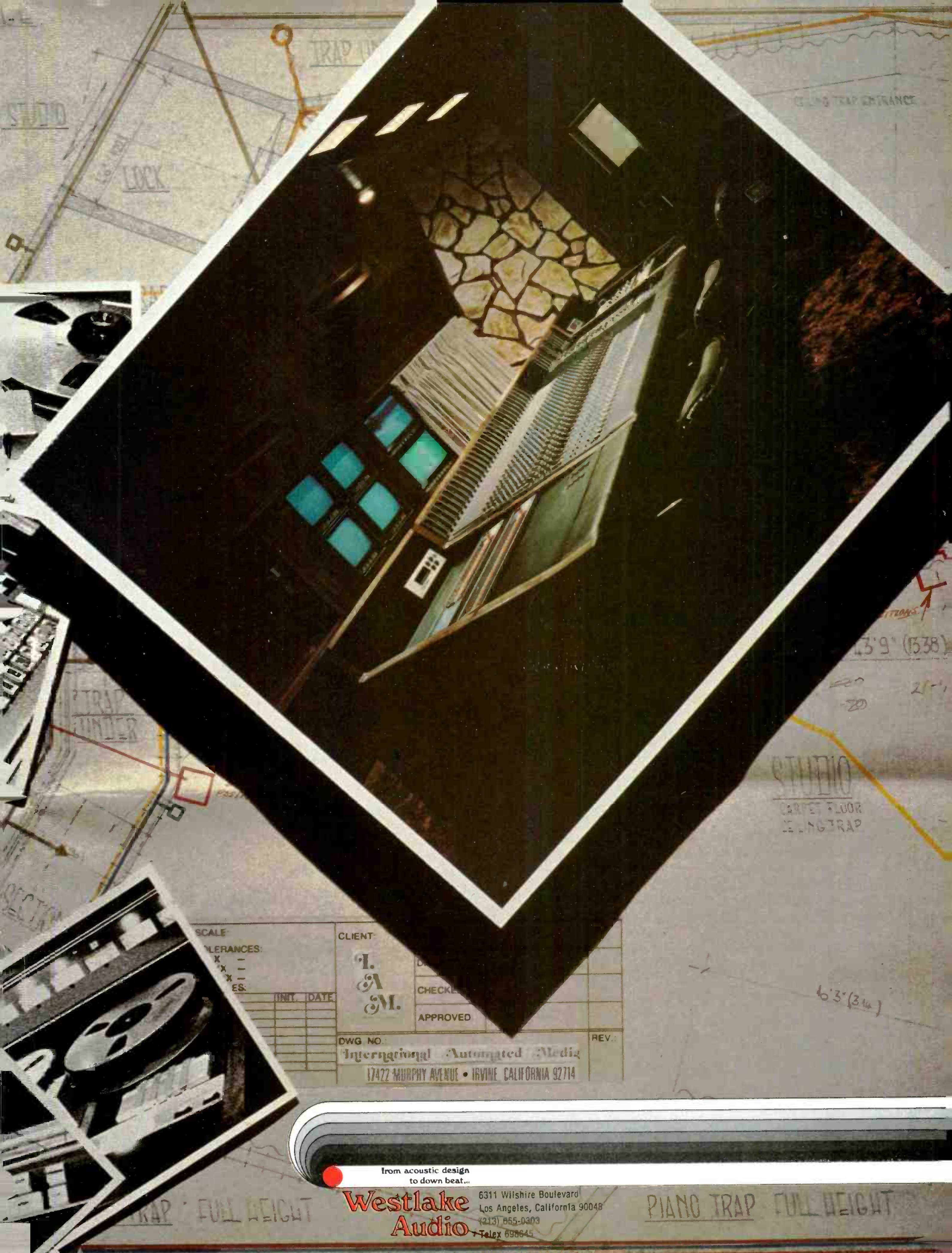

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continued from page 48 . . .

proceeed to the next instruction address. There we are directed to CPI with the value 255 decimal. The next instruction tells us to check the condition of the zero flag. Since subtracting 255 from the value in A does not equal zero the zero flag is not set, and we can jump to the address associated with this instruction. This brings us back to where we started. We are in a *loop* and cannot get out until we have raised the value of A equal to the value associated with the CPI instruction. When we finally get a zero flag we proceeed to the next instruction which is an OUT 01. Immediately we notify the output port to make a copy of the A register contents and give it to the outside world. Then on to our next instruction, DCR A. We find outselves once again in a loop, but this time we are looking for 00 in the A register. Eventually out count reaches 00 and we *fall through* to another *output* instruction. This time the A register contains 00 so we discard our previous output value and give 00 to the outside world. Once this is done we read the next instruction which is an unconditional *jump* to UPCNT where the entire process begins all over again. While it takes many words to describe this process, in real time it happens quickly:

The INR instruction executes in 5 microseconds (5/1,000,000 of one second).

The CPI instruction executes in 3.5 microseconds.

The JNZ instruction executes in 5 microseconds.

Total Loop Time: 13.5 microseconds.

Times 255 counts equals 3,442.5 microseconds.

When the timing of the positive phase (while we are outputting 255) is added to the timing of the negative phase (when we are outputting 00), we have a total of *approximately* 6,890 microseconds, which is roughly equal to 150 Hz. We have succeeded in producing a continuous tone that will go on until we instruct the machine to stop or pull its plug. In real computerese the program would look as it does below. Notice that we have created labels to help us locate the two primary *algorithms* which make up our program. These are UPCNT for our upward counting routine (INR A, CPI, JNZ) and DNCNT for our downward counting reoutine (DCR A, CPI, JNZ). The D after 255 is for decimal value. This must be identified since several different number base systems are used by computerites. Zero and 1 are not identified since they are the same in any number base.

```
UPCNT   INR A
            CPI 255D
            JNZ UPCNT
            OUT 01
DNCNT   DCR A
            CPI 00
            JNZ DNCNT
            OUT 01
            JMP UPCNT
```

While this is a very simple example, it serves to illustrate some of the important basics of microcomputer programming. Whether you are dealing with a one-board microcomputer that fits in the palm of your hand or a million dollar installation directing space flights, an algorithm is always an algorithm, incrementing the A register always adds 1 to the contents and stores the total back where it came from. Mnemonics may change from one assembly language to another, but these are just the langauges of the various machines. Now that you've started to get a handle on this mystical world of computer programming, I hope you agree that this, like other things we have yet to learn, are not as complicated as they looked from the outside.

In the event that there are reservations on the part of the reader as to the applicability of digital technology to audio as it is practiced here and now, we are about to present an actual application.

SMPTE TIME CODE STANDARD

The SMPTE Standard specifies a digital time code format and modulation method for use on video and audio magnetic tape recorders to be used for timing and control purposes.

80 Bits per Frame
32 User Binary Spare Bits (8, 4 bit groups).
16 Sync Word Bits
28 Assigned Address
4 Unassigned Address; always 0
(Assignment of these bits is reserved to the SMPTE Video Tape Recording Committee)
Bits are numbered 0 through 79.

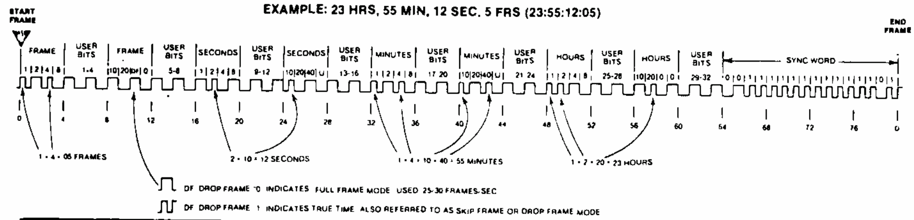
SYNC WORD
The Sync Word of 16 bits is a fixed format as shown. Its format permits rapid and accurate recognition of the tape direction. This information, when sensed by a SMPTE Time Code reader, determines if the reader should count UP or DOWN.

DROP FRAME
SMPTE DROP FRAME MODE is also referred to as: *TRUE TIME or SKIP FRAME Modes.* This mode is not required for monochrome recordings. It can be optionally used for color recordings. It is convenient to use this "Drop Frame" mode if the edited material has to be precisely timed for a "TRUE TIME" playback. If it is not used on a color recording, there is a resulting timing error of 108 frames (3.6 seconds) per hour.

The SMPTE DROP FRAME format is:
BIT 10 (Ref. "DF" in example) if a "1". Then: the first two frame numbers (1, 2) at the start of each minute, except every 10th minute (0, 10, 20, 30, 40, 50), are omitted from the count.

FULL FRAME
SMPTE FULL FRAME time code tapes can be used for color recordings that require precise TRUE TIME playback provided the resulting time difference adjustments are made by subtracting 3 seconds and 18 frames per hour from the FULL FRAME tape duration time.

1 FRAME WITH SMPTE EDIT CODE
EXAMPLE: 23 HRS, 55 MIN, 12 SEC, 5 FRS (23:55:12:05)



READING THE SMPTE EDIT TIME CODE WITH A MICROCOMPUTER

The problem of reading the SMPTE Edit Time Code is becoming a more common one as more and more studios undertake video sound track audio sweetening projects and/or use of the code as a basis for multi-machine syncing or simply as a means of logging program and take sequences. Currently, a single SMPTE time code reader will price out at around $4,000.00. This doesn't buy us anything in the way of sync capability nor does it provide us with a time code generation system. It is easy to see that several more thousands of dollars will be necessary to complete even a single SMPTE timing and synchronization system.

The remedy for this dilemma is at hand. The minicomputer, programmed with software to be described, has more than adequate capability to unscramble the digital data recorded on a master tape sync track.

The recorded electrical signals, as we all have at least heard, correspond to time-coded information as recorded on film or tape. With the code we imprint a continuous string of unique addresses describing hours, minutes, seconds, frames and up to eight additional characters which may be the reel number, Master number, Date or Take number. While the code was originally conceived for Film or Videotape it is totally applicable to Sound recording. Using a 60 Hz reference our tapes are defined in 1/30th of a second increments with a recoverable carrier usefull for multi-machine synchronization. With the code we have the ability to locate, to an accuracy of 1/30th of a second over an unlimited time period, virtually anything that can be recorded magnetically.

The implications of this new computer technology might be best appreciated if we view it in terms of numbers. The computer system requires about 1/8th of a second to do its job. Within this time some 80,000 data processing steps are executed. Given appropriate software now under development, SMPTE time code can be generated by the microcomputer and multi-machine synchronization accomplished with the same hardware system. Only a few minutes of time is necessary to perform a routine program load and sequence initialization.

All of these functions, and more, can be had from any one of several minicomputer systems available for about $800.00 or less. This means, here and now, that tens of thousands of dollars in dedicated hardware can be replaced by a single compact data processor and inexpensive interface equipment.

We cannot begin to speculate as to how far these applications will go. However, it is safe to say that *we have the technology* now at a price we can afford. All that is required is for us to write the applications software to usher in this next technological revolution.

SMPTE READ MINIMUM SYSTEM HARDWARE REQUIREMENTS SUMMARY

1. Fundamental 8080 Microcomputer system including: Microprocessor board; power supply; front panel or keyboard input

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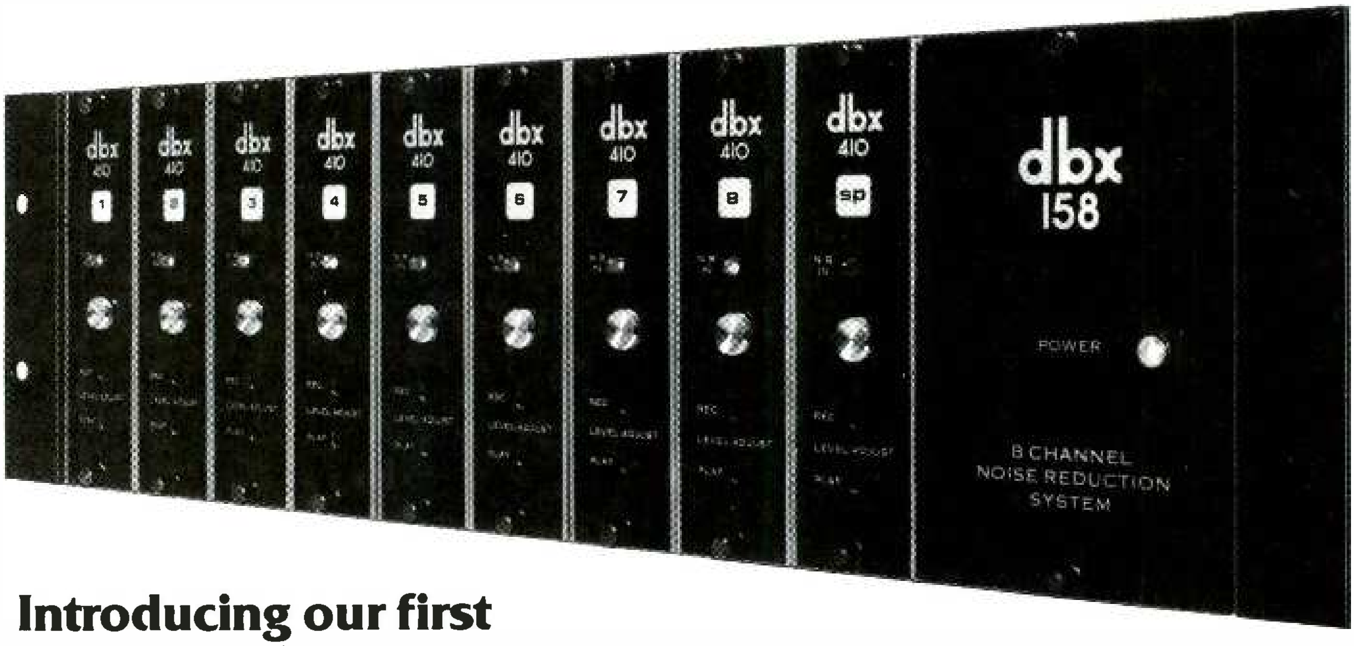

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The dbx 158 offers the semi-pro recordist or small studio all the advantages of dbx professional systems, including 30 dB of noise reduction, and 10 dB additional recorder headroom. It's a classic 2:1 mirror image compander which preserves the full dynamic range of program material without audible tape hiss. Each module contains separate record and playback noise reduction electronics. Its simultaneous record/playback capability permits the noise reduced, decoded tape to be monitored while recording without manual switching or remote control.

Requiring only 5¼" of rack space, the 158's light weight (17 lbs.) makes it easily portable for location dates. And naturally, tapes recorded with this system are compatible with any other dbx professional tape noise reduction system as well as on board dbx tape noise reduction in TEAC/TASCAM recorders. We'll be happy to send you further information and the name of your nearest dbx dealer. Just write us.

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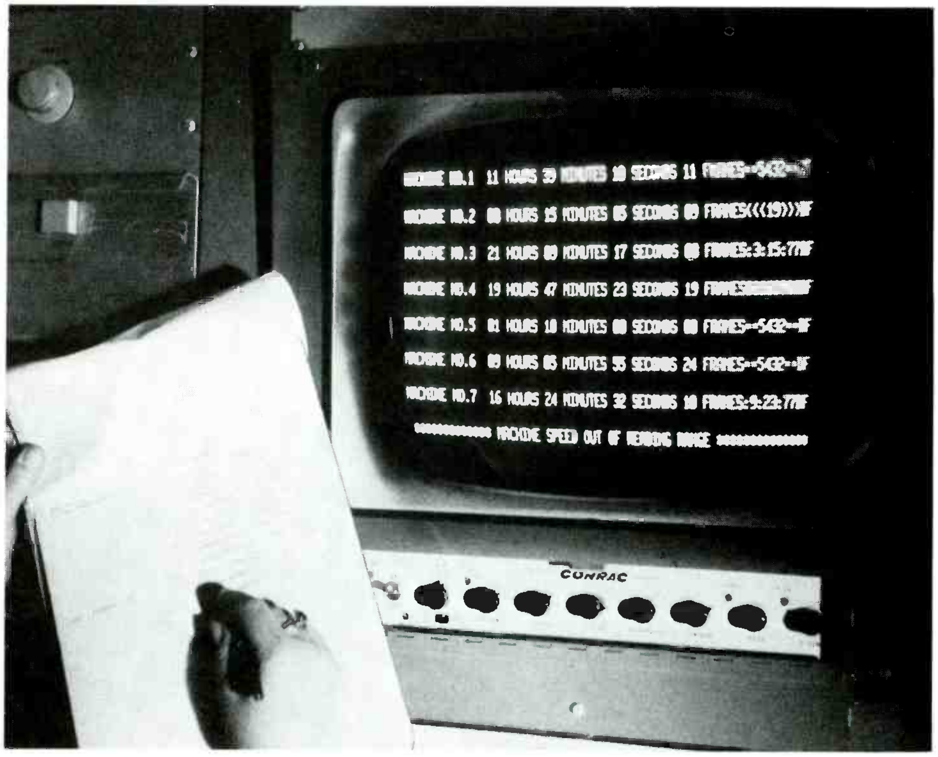

board.

2. Video Display Module, available from a varity of sources to meet these specifications: Minimum of 1K on-board memory, addressable by computer; Accepts ASCII characters for screen display. If flicker-free display is not built into hardware, unit must supply horizontal drive pulse for software synchronization.

3. Video Display Device: Modified TV set or composite video monitor.

4. Computer Memory: Minimum of 3K contiguous, addressed from D000 hex.

5. Parallel Input Port: Two inputs, 8 bits each addressed at 02 and 03 hex.

6. Parallel Output Port: Required only for generation of test signals. Addressed at 03 hex.

7. Analog/TTL Interface Module: One unit is required per signal source. A complete kit for one unit is supplied with software package. Additional kits are available from this writer at $7.50 each (seven dollars and fifty-cents).

INTERFACING INSTRUCTIONS

It is recommend that prior to loading SMPTE READ you follow these steps:

1. Assemble and check out, per manufacturers directions, all components of your system.

2. If necessary modify your VDM board to supply horizontal signal.

3. Assemble and connect an interface module.

4. Load SMPTE READ. Execute it from D000 hex.

Editor's Note:

Author Wiener graciously offered to allow R-e/p to reproduce the entire copyrighted SMPTE READ software program in this issue. Space limitations prohibited doing so. The complete SMPTE READ program is available from the author at modest cost by writing:

J. S. Wiener
4440 N. Kedzie Avenue
Chicago, IL 60625

GENERAL DESCRIPTION

SMPTE READ is a totally self-contained operating system. It is written for any microcomputer system which uses the 8080 instruction set. These include the 8080, 8080A, 8085, Z-80, and Z-80A microprocessors.

The program resides in 3K of contiguous memory beginning at D000 hex. An additional 1K of memory is provided by the on-board display storage of the Video Display Module.

It requires only one parallel input port to accept as many as eight data sources. This port is assigned address 03 hex. Provisions are included in the software to use input port 02 for outside world control of the error detect display routines. While not required for normal operation this feature is handy for initial start-up and maintenance of the magnetic tape hardware. Figure 1 is an actual photo of the system displaying its full compliment of 8 data inputs. This display is automatically up-dated approximately once per second.

The display shows the most recent VALID read unless instructed to display error messages by bringing low one bit of input port number 02. When a bad read is detected one of four error messages will be displayed.

BAD READ—TOO MANY DROP-OUTS will be displayed when the system knows that it is receiving data, but there are too many dropouts in the recorded signal.

***BAD READ—CODE DOES NOT

TRANSLATE PROPERLY***, or, ***CANNOT FIND SYNC STREAM*** are shown when errors are found in the reconstructed signal.

MACHINE SPEED OUT OF READING RANGE indicates just that. The machine is either running too fast or too slow for the tape machine playback electronics to produce a readable signal, or the machine is stopped altogether.

While all of these signals render the read useless, their importance will be appreciated if it becomes necessary to debug either the software or hardware.

Each of the displayed lines consists of the following data:

MACHINE NUMBER. The tape machine supplying the data on this line.

HOURS, MINUTES, SECONDS, FRAMES, (either tape or film) are self explanatory. To the right of the word FRAMES are eight character locations which will display any numeric data which has been entered into the USER BITS at the time of recording. Just to the right of this 8th character are two additional locations. If the timing generator was set in compliance with SMPTE DROP FRAME (color) format a DF is displayed here, otherwise it is blank.

INPUT CONDITIONING

To properly condition the tape recorder output very little circuitry is required. Most microcomputer input boards will recognize anything in the +2 to +5 volt range as a HI and anything near ground as a low. A simple open collector transistor driven by output audio should yield good results. By special arrangement with the program's author you may purchase completed input modules for $7.50 each, postpaid. The module is cased in a standard 2 circuit phone plug body, is powered by the computer's regulated supply and uses a voltage comparator input driving a TTL compatible output device.

TEST CODE

For those who wish to install the program but do not have current access to time coded materials, a special test program is included from DB40 to DB88. By connecting a standard tape recorder line level input to any one of the eight bits of output port 03 and recording at normal levels, a test tape can be prepared. Although the same information will be repeated for every frame it is otherwise identical to a normal SMPTE EDIT TIME CODE.

A Standard Digital Interface for Peripheral Equipment

by Richard Factor
Eventide Clockworks, Inc.

Hi there, automated studios! Greetings, peripheral equipment manufacturers. Salvete, automated console purveyors. How 'ya doin', youse tape recorder guys?

We all have a problem you know. No, I mean *BESIDES* those. What is it? Well, we all make or use audio equipment, and by the grace of two conductor shielded cable the equipment manages to send audio signals amongst its various selves. But that isn't enough. There is a lot more than audio information running around the normal studio. And I do mean running around.

Consider the mix: What is the first thing done? The tape recorder is started by remote control. After which, one or more people frantically finger faders, pan pots, equalizer adjustments, flanging knobs, delay settings, and compression controls. Despite the ingenuity of the automation people, the above statement is largely untouched. Most automated consoles control only the fader setting. Even the most recent, which control EQ, panning, and other console functions cannot run the myriad peripheral units in the average studio. Thus, if one wishes to change the setting of a parametric equalizer, a digital delay line, or whatever, one (or one's assistant, or the producer, or any of the semi-trained bodies pressed into service for a complicated mix) must be physically present at the offending device to change its setting. In these days of energy shortages and communication revolutions, it hardly seems appropriate to require physical scrambling to apply a few dyne-centimeters of force to a conic section attached to a mechanical transducer. The probability of error (and injury) increases drastically with the number of bodies, and as the number of control possibilities increases, the need for automation which initially became necessary to combat the spread of the console becomes necessary to combat the otherwise inevitable expansion of peripheral functions.

There are several possibilities for improving this situation. (We are all reasonable men. I do *NOT* propose that console manufacturers mount waldoes on the sides of their boards.) The first is that more of the so-called "peripheral" functions be included in the console itself, where they are under the direct control of the automation system. This is a distinct probability in some cases. (Witness the prevalence of multi-function equalizers which used to be elsewhere now mounted on each module.) Fortunately for us peripheral manufacturers, two things militate against this. The first is the fact that functions required in small numbers (delay, echo, special effects) simply are not practical or necessary in a one-per-channel basis. The second is that, given the free-market system combined with the selecivity of engineers and producers, somebody is always going to want something not provided as part of any given system.

Another possibility, and one which is far more practical, is to AUTOMATE THE PERIPHERAL EQUIPMENT! Well, I've finally gotten to the point. The question is: How the hell can you get equipment manufactured by perhaps a hundred different companies to interface with automation computers and consoles manufactured by about a half-dozen others? The answer, of course, is to design a standard interface system, upon which all manufacturers can agree, which will permit the diverse units to communicate. If you're politically oriented, you just threw up your hands and had a good laugh, and, just maybe, got to the rest of this paragraph. If, on the other hand, you've ever been involved with a standards committee, you just threw up your hands, had a good laugh, and tore the magazine in half. Now that we have a few of the politicians and all of you naive souls who don't know the problems of designing industry "standards" still with us, let me give you the good news: We don't

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need a standard. We already have one. Yes we do. But before we discuss the actual interface, let's discuss the requirements. They can be broken down into two categories, human and electrical. The human requirements are fairly simple: The ideal interface will be able to do everything a person can, and perhaps a bit more. It should be able to READ the control settings on the instrument, and it should be able to MODIFY them on remote command. Because it is electronic, it should be able to follow instructions virtually instantaneously, so that the instrument can be configured as rapidly as needed, even between notes. The electrical requirements needed to effect the human requirements imply a bidirectional transmission capability and a high enough data transmission rate to in effect set all the knobs on however many units are connected to the system in a tenth of a second or less. The system should be capable of connecting a large number of peripherals at minimal expense. It MUST be device independent: any piece of equipment, from any manufacturer, must plug in without the system knowing or caring that it's there, until some action is requested. It should be compatible with local control (much as most tape machines can be put in PLAY or REWIND either from a remote control or from the front panel).

Well actually, three categories. The third, possibly a sub-section of the human requirement, is that the system be politically acceptable. Without this we have nothing but committee meetings into the early morning and beyond. A politically acceptable solution must not give any manufacturer an unfair advantage, must be implementable by companies without exceptional technical skills, and must be reasonably economical. The interface standard I'm about to suggest embodies all these features. It is known cryptically as "IEEE Std 488-1975 (ANSI MC 1.1-1975)".

The Standard is a document published by the Institute of Electrical and Electronic Engineers. It is ligitimization of an interconnection scheme initially designed by Hewlett-Packard, an internationally known and highly regarded electronic instrumentation manufacturer. Because they make many instruments and the computers to control them, they invented a system to allow properly equipped instruments to plug into a common data/signal bus. This eliminated the requirement for their many divisions to design specific interfaces when putting systems together. The interconnect scheme was (and still is) call the "HP-IB". When the system became standardized, other instrumentation companies were loath to call their interface the "HP-IB", and so the term "GP-IB", for General Purpose Interface Bus came into use. This is a good term, is certainly easier to say than IEEE-488 1975, so let's use it from now on.

As you can see, the GP-IB is well established among electronic manufacturers in general. This leads us to our other criteria of acceptability: ease of implementation and economy. The standard applies to the entire electronic industry, not just the nearly insignificant (in dollar volume) branch calling itself "professional audio". This leads to economies of scale which, were they not present, would bankrupt most of the companies we know and love. Motorola and Intel, two large semiconductor manufacturers, make Integrated Circuits which in effect implement the standard without the user necessarily knowing the details of, or even understanding the theory behind, the GP-IB. These chips are presently available, reasonably priced, and will no doubt become more of both as competition and second sourcing sets in. These chips are designed to interface with the GP-IB on one end, and with a microprocessor data bus on the other. The microprocessor can employ its other capabilities to operate the peripheral equipment. The remainder of this article comprises an introduction to the GP-IB, an examination of the general requirements for peripheral equipment automation, and finally a fairly detailed example of an implementation of the interface, specifically a microprocessor controlled remote unit for the Eventide 1745M Digital Delay Line.

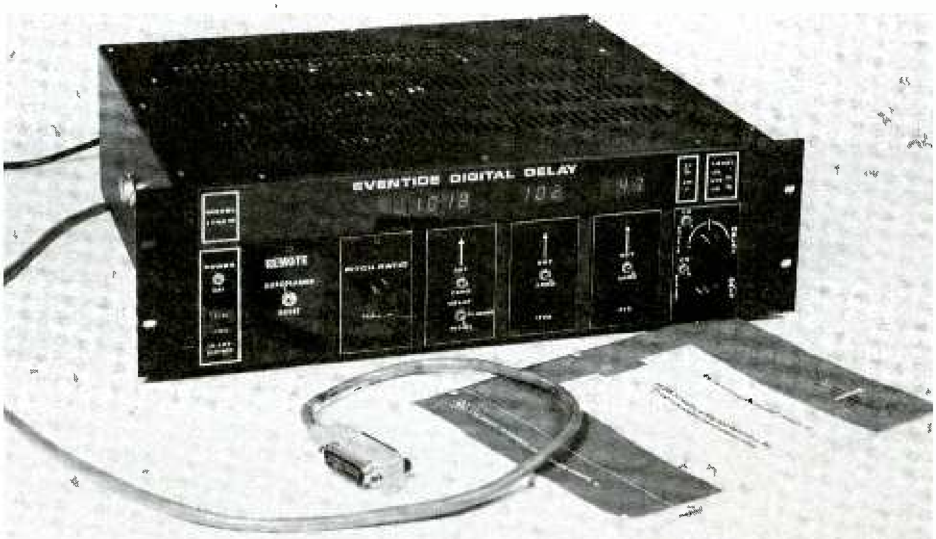
THE INTERFACE BUS

A GP-IB system consists of up to three types of elements. An instrument can be a controller, a talker, or a listener. A specific instrument can perform a combination of functions: it would be a talker when the console or computer is reading current settings; it would be a listener when the console is updating those settings. The console or computer would generally be the controller, although provisions are included for passing control from one instrument to another. In this simplified discussion, we will use the term console and computer interchangeably since they will probably be supplied as a system by one manufacturer and so whose interface is only a matter of academic interest.

The physical portion of the bus consists of special connecting cables (see photograph) which are mass produced, available normally 1, 2, and 4 meter lengths, and have rather cute piggy-back connectors so that each end of each cable is simultaneously a plug and a jack. All bus-compatible instruments have a 24 pin female connector. The cable mates to this connector and provides an identical connector immediately behind it. The number of cables that can be piggy-backed is limited primarily by the stress the chassis-mounted connector can tolerate.

Electrically, the bus consists of 16 information lines and 8 grounds. 8 of the lines, DI1 through DI8, are DATA lines which are used to transmit the actual details of desired operation, and which instrument should follow the instructions. Three of the lines, DAV, NRFD, and NDAC, are HANDSHAKE lines, which synchronize operation of the system. The remaining five lines are for BUS MANAGEMENT. The bus management lines coordinate the various peripherals connected to the bus so that they know when they are receiving address data, command data, are being asked to respond, etc. The following description is extremely abbreviated and, unfortunately, the standard document is virtually unreadable. Those of you vitally interested in this subject should probably write to any of the manufacturers of interface chips or bus-compatible equipment for literature.

The BUS MANAGEMENT LINES are the following: IFC, or INTERFACE CLEAR: This line is driven by the controller to place the interface in a known state. It sets talkers and listeners in a quiescent mode. SRQ, the SERVICE REQUEST line is driven by an instrument to notify the controller that it needs help. Typical instrumentation applications for this line are printers notifying the computer that they are out of paper, or that some error condition exists (broken print chain, open interlock, etc.) An audio SRQ message might include a notification of improper signal parameters (clipping, too low level), or an improper or illegal control setting, whether done from the fornt panel or remotely. The ATN (ATTENTION) line tells the peripheral how



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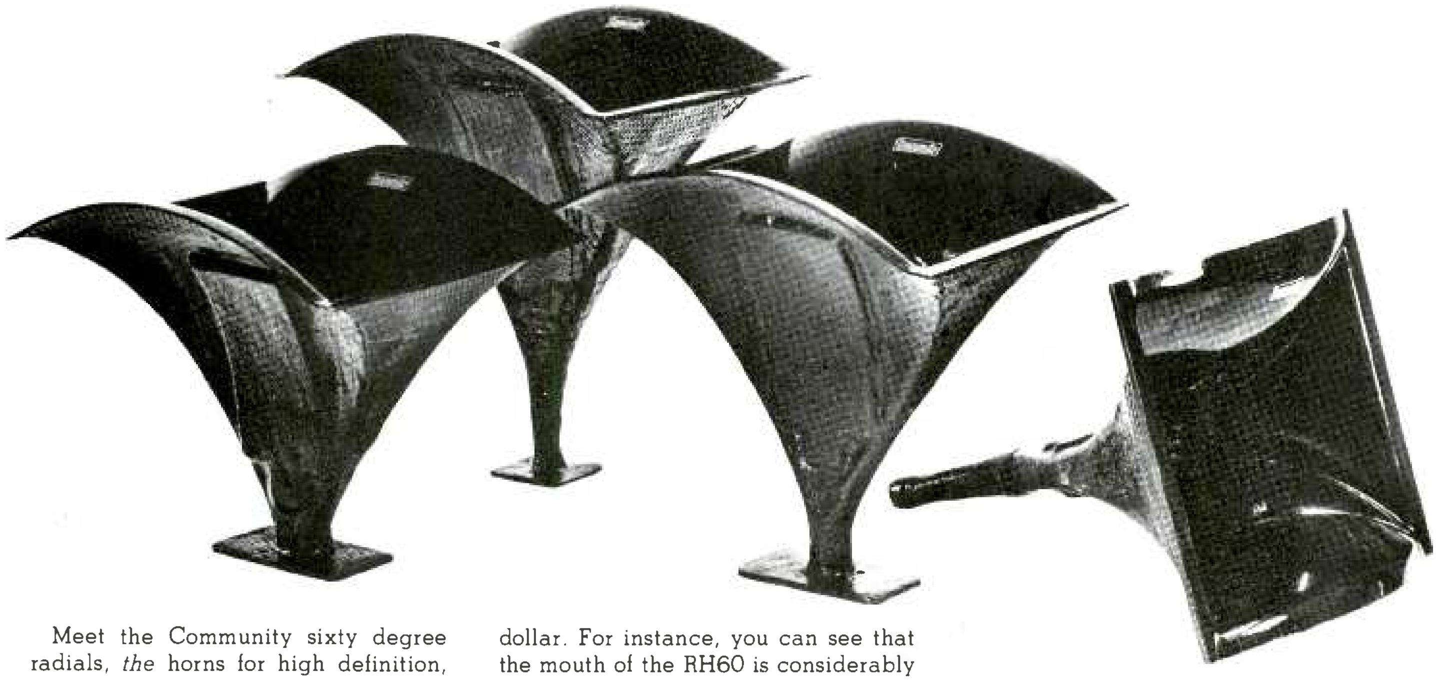

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									−12	−7	−6	−4	−3	−1	0	+1	−1	−2	−1	−3	−5	−5	−9	−9	−9	−13	−17

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

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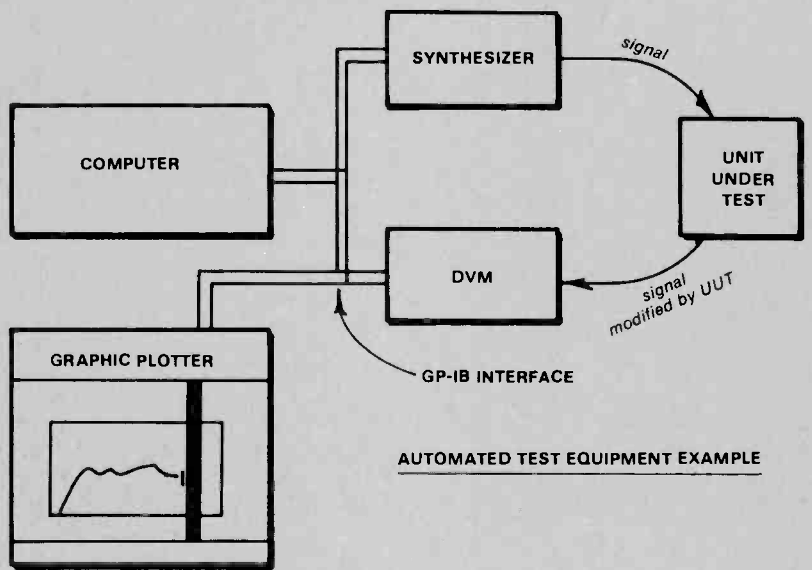
to interpret the data on the DATA lines. When it is true, commands or addresses are present. When false, device-dependent data is being transmitted. REN, REMOTE ENABLE, selects between two sources of programming data, such as local or remote. EOI (END OR IDENTIFY) tells the controller that the final byte of a multi-byte message has been transmitted. It should be emphasized that this is an abbreviated discussion. For instance, asserting SRQ doesn't tell the controller anything other than service is needed. Typically, the controller will then poll the devices on the bus to find out which one requested service (the SRQ line goes to all in parallel) and just what service is needed.

The HANDSHAKE lines serve to synchronize operation of all units connected to the bus. This is one of the more clever aspects of bus operation. First let's look at the problem. Suppose the controller wants to send a message to all the peripherals. One of the peripherals may require time to interpret the instruction, while another can accept it immediately. Although the second will be ready for the next datum immediately, the first is still considering what to do. The controller can't just go ahead and transmit because the first unit will ignore the data. The handshake procedure prevents this problem. The sequence is as follows: The controller sends DAV (DATA VALID). This indicates that the peripherals should accept the data and follow their orders. The peripherals accept data in their own time (usually microseconds) and when each has absorbed its byte, it asserts DAC (DATA ACCEPTED). This is a wire-anded line, so that all DAC lines must be true before DAC appears true at the controller. When it does, the controller knows that all the peripherals have received their data, and it no longer has to send DAV. It releases DAV and begins preparing the next data byte. Remember, though, that peripheral 1 is still thinking about its instruction. Although it has accepted the data byte (DAC true), it isn't ready for a new one. This is where RFD (READY FOR DATA) comes in. This is another wire-anded line. When peripheral 2 has digested its data, it sends RFD, but RFD doesn't appear true until peripheral 1 has also sent it. Thus, a new DAV cycle isn't initiated until all units have both accepted the old data byte and are ready for a new one. This system allows a tremendous variation in timing for different instruments. GP-IB afficionados will note that I have ignored the polarity of the handshake signals, assuming positive-true logic. This has been done to facilitate comprehension.

The GP-IB was originally designed as an instrumentation interface. It is not, and is not intended to be, a high speed data handling interface.When studios become completely digital, another arrangement will be necessary to actually carry digitized audio data. The bus is designed to transfer 8 bit data bytes at a maximum rate of 500,000 bytes per second. The maximum number of connections which can be made to the bus is 15. This maximum is determined by line loading considerations and is not a limitation on the number of functions which can be performed. The maximum cable length is 20 meters, or less if many devices are connected. Let's look at these limitations and see how they may affect studio peripheral interconnections. The first thing to look at is the data transfer rate, as if this is insufficient, then the entire scheme falls to the ground.

Let's assume that we have 8 peripheral devices connected to the bus as follows: 1 digital delay line, 2 flangers, 3 compressors, 1 8-channel equalizer, and a tape recorder. (Note that, except for the delay line no such bus compatible peripherals exist. Thus we are going to assume "worst case" requirements for everything.) The delay line (we assume) has 3 outputs and 2 other controls, double and repeat. There are other functions which can be controlled remotely, such as flanging and random doubling. As a practical matter, it is unlikely that every function will be employed simultaneously, but if they are, 1 byte will be transmitted for each digit of each output, which is 3 x 5=15 bytes, +1 byte each for the miscellaneous functions, say 4 more. This gives a maximum of 19 bytes for the DDL. The tape recorder is the only other "digital" instrument in the list. Assume 1 byte is required for basic operation (PLAY, FF, STOP). These instructions are mutually exclusive, so that a maximum of 1 byte will be necessary. An additional byte per track can be employed to define record, play, and sync functions, and, for safety, let's add another 5 bytes. The other bus devices are analog, and their control settings must be digitized in some way. The most obvious way to interface such devices is to arrange



AUTOMATED TEST EQUIPMENT EXAMPLE

The original and most widely accepted purpose of the GP-IB was to allow computers to control instrumentation in so-called "ATE", or Automatic Test Equipment configurations. A small system might include a controller, to output data to a frequency synthesizer, the "UUT", Unit Under Test receiving the output frequencies, and a digital voltmeter to measure the output voltage of the UUT. The DVM output would be communicated to the controller, which would then decide whether the UUT was within specification, and, perhaps, transmit the information to a graphic plotter to provide the final user with an error curve or quality assurance. Such systems can be expected to become less expensive as GP-IB usage grows and computer prices decrease.

At Eventide, we are developing methods of using remote control capabilities in conjunction with computers to reduce human error and labor requirements in the checkout of our products. Since the remote unit was developed, our checkout procedure has included plugging it into each delay line, and having the computer cycle through the various delay settings and modes. The output is continuously monitored on a spectrum analyzer to look for periodic artifacts indicative of a bad RAM, and the output voltage is monitored at various frequencies to check response. Since some of our test equipment isn't bus compatible, human intervention is still required, but most interpretation is left to the computer. A cute feature of the procedure is the readout test: The computer sets the readouts to a random set of values and requires a human to key in the reading. This makes it impossible to "pass" a unit which has counter or display problems, even by accident. Our newer products, such a the S1066 16 output DDL will have greater bus-compatibility and self-test capabilities which may be accessed by the remote control.

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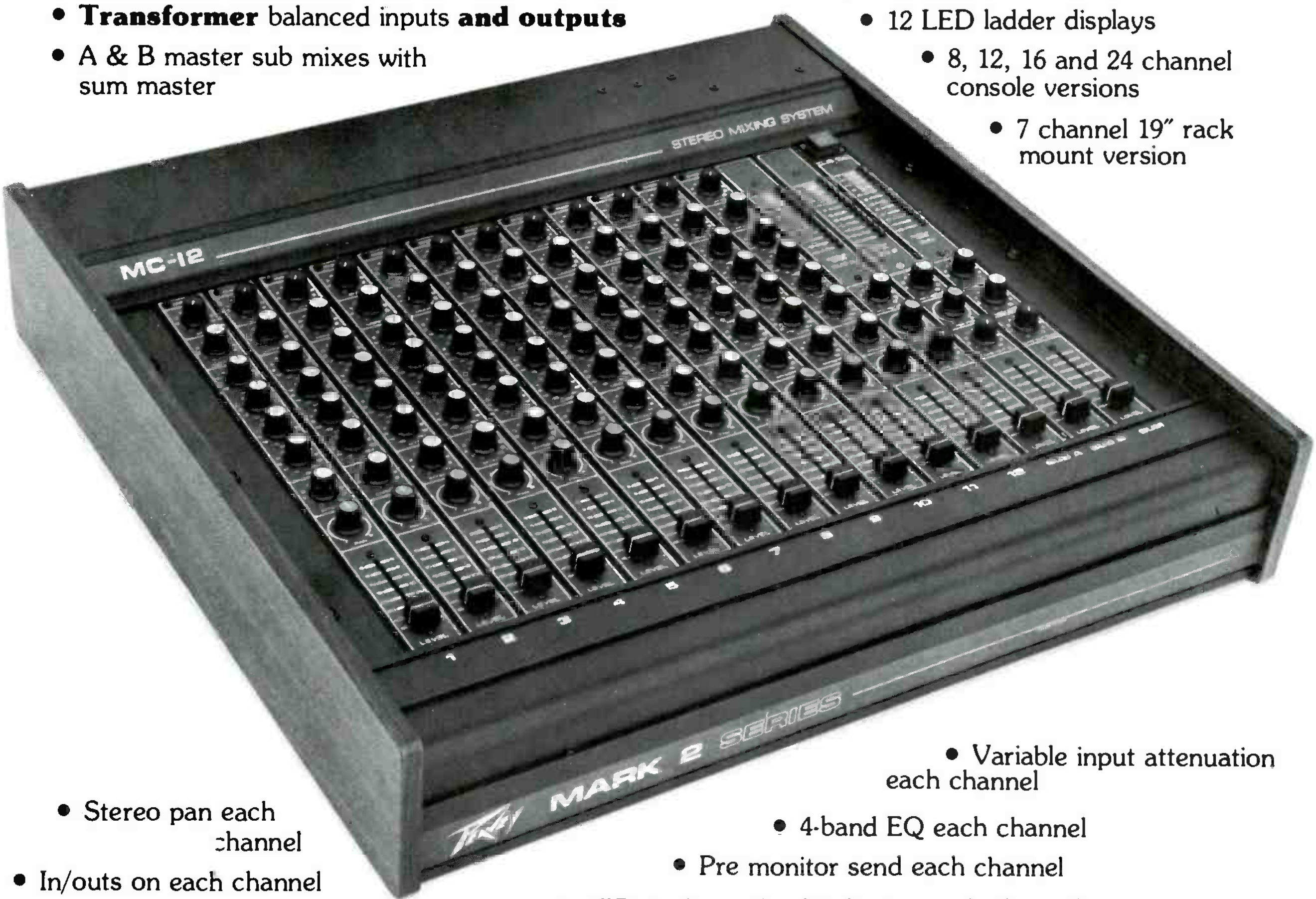

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for each control function to be controlled by a digital to analog converter which will take an 8 bit byte and convert it to any one of 256 analog values. This corresponds to a control resolution of about one degree of rotation, as most controls have just about 270 degree rotation. Of course, the manufacturers of the devices will have to provide the D/A interfaces. It is relatively simple to provide logarithmic and antilog characteristics with converters, but let's hypothesize that additional resolution is necessary, so we will allocate 3 bytes for each analog control, 1 to identify it, and 2 to set it. Thus, assuming 6 controls on each flanger, 6 on each compressor, and 4 on each of 8 equalizers, we come up with a grand total of 250 bytes to completely configure the system. Now, let's double the number to provide for future expansion. With a maximum data transfer rate of 500Kbyte/sec, this gives an update time of 1 millisecond for an expanded system. Our original cirterion was for a tenth of a second, so if everything is functioning optimally, we beat our goal by a factor of 100. Before we get too cocky, let's remember that devices may not digest data immediately, and have to be addressed as well. If the device on the bus uses a microprocessor such as the one to be described, the acceptance rate is largely a function of the software written for the peripheral device control program. To leave a large safety factor, a device should not require more than about 10 microseconds to complete its handshake with the bus. This is well within the range of realization for standard hardware/software combinations.

Actually, this 10 microsecond maximum may be optimistic for some units and pessimistic for others. The equalizer postulated above should be able to accept data almost immediately, since all that is required is to latch one byte into the D/A converter associated with its particular control. Other instruments may require longer. For instance, the 1745M delay line has "ZERO" command which sets all delays to zero and resets the 2X and REPEAT modes. Although this command is accepted immediately by hardware, it takes several times 10 microseconds to complete. If for some reason the command were issued successively several times, the third time would result in a much longer handshake. Of course, there is no reason to issue "ZERO" several times in succession, but if a peripheral does have several valid lengthy commands, the manufacturer should arrange to provide a command buffer to prevent tying up the bus.

The other bus characteristic should be no problem. The majority of control rooms are nowhere near 20 meters (over 65 feet) in any dimension, although circling the room with the bus cable will approach this length. The interface standard document implies that the cable can be lenthened at a sacrifice in data rate (which we are already assuming to be an order of magnitude lower), so there should be no problem there. The number of devices could be a problem as many studios have more than 15 peripherals. Note,

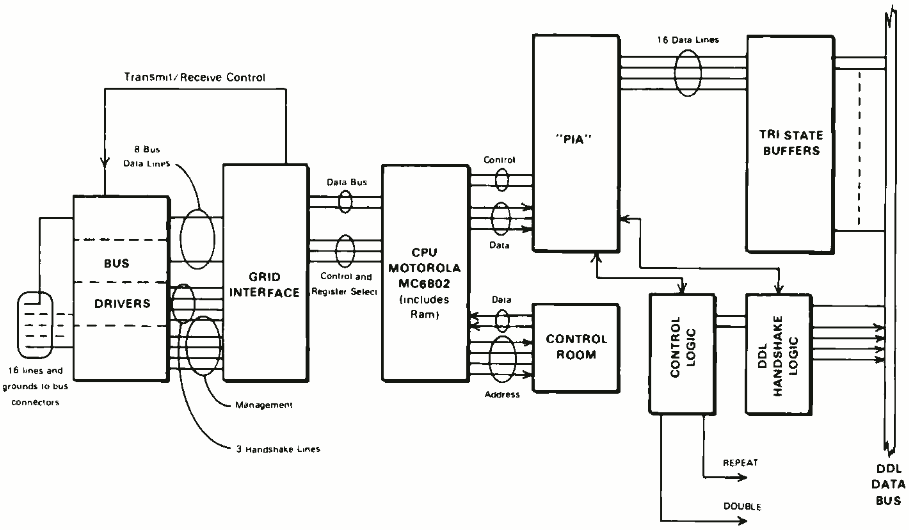

Figure 3: SIGNAL FLOW of the GP-IB (IEEE 488) DATA BUS INTERFACE

however, the previous assumption of an 8 equalizer unit. Economic considerations would dictate that a manufacturer would provide for multiple device control from each interface. Also, there is no limitation on the number of busses that many be used. Finally, the bus provides an "address pass through" feature which provides a convenient method to allow one interface connection, in conjunction with an expander "black box" to control additional busses!

BUS MESSAGES

The bus manages transmission of data from the console to the peripherals. All the peripherals see are "Device Dependent Messages". These messages can be anything which can be transmitted in 8, or any multiple of 8 bits. Anything from a control code to a copy of this article. As a matter of form, the committees in charge of such things recommend using ASCII (American Standard Code for Information Interchange) to transmit device dependent messages. This makes good sense in some cases and not in others. For instance, in the case of the equalizer, a binary control setting can be transmitted in 1 byte. If ASCII were used, it would require 3 bytes of numerical data to specify the control position (say "135") for 50% rotation. This is also inefficient in terms of control, as these bytes would have to be translated to binary again. On the other hand, if it is desired to adjust the cutoff frequency of the EQ in absolute terms, a message such as "H15", which could be internally translated as Hi-Cut 15 kHz could be sent. The manufacturer of a single, low cost remote EQ would probably prefer the binary system, the manufacturer of a multi-EQ unit would find ASCII control adding little marginal cost and might wish to advertise it as a feature. It would, of course, be very nice if everybody could get together and recommend an industry-wide device dependent message data format, but this is not nearly so vital as ascertaining that the electrical interface is compatible. Presumably any controller capable of driving the GP-IB will be able to transmit binary or ASCII messages. In the case of the 1745M Delay Line, ASCII was the obvious choice since all the input data is digital to begin with, and a microprocessor is available to process the ASCII data and control the DDL. An example command would be "Z,A123000,B22034,R", where the letters and numerals are the 7 bit ASCII version of each. The command shown first Zero's all functions, then sets output A to 123 milliseconds and output B to 220.34 milliseconds, and finally places the unit in the Repeat mode.

When we built the 1745M Digital Delay Line, we decided that automation was just around the corner, and, because we had no idea whether to make the unit compatible with a specific console or design our own remote unit, we opted for deferring a decision on specifics while designing the unit to be compatible with remote control. This was not too difficult because the DDL is a bus oriented machine itself, and there was a convenient way of presetting the various outputs built into the other necessary parts of the design. It was decided that the future remote control would reside in the leftmost slot of the mother board, and be able to control the delay settings of the other outputs. No provision was made initially for control of the 2X and REPEAT functions, but this can be done with a minor modification which brings the switch contact to the remote control unit through the printed wiring so that the DDL remains modular. The only other modifications require making a hole for the GP-IB connector and mechanically and electrically paralleling a low voltage transformer with the one already present to provide power for the new microprocessor. The job of the GP-IB remote control module is to form an interface between the bus in the external world and the DDL address bus, DDL control signals, and DDL handshake requirements. The balance of this article will describe the implementation of this remote control. As this is a digitally oriented issue, we shall delve as necessary into the microprocessor and associated chips, and the programming of them, without which any computer is about as useful as one tube of epoxy. And, I should state immediately,

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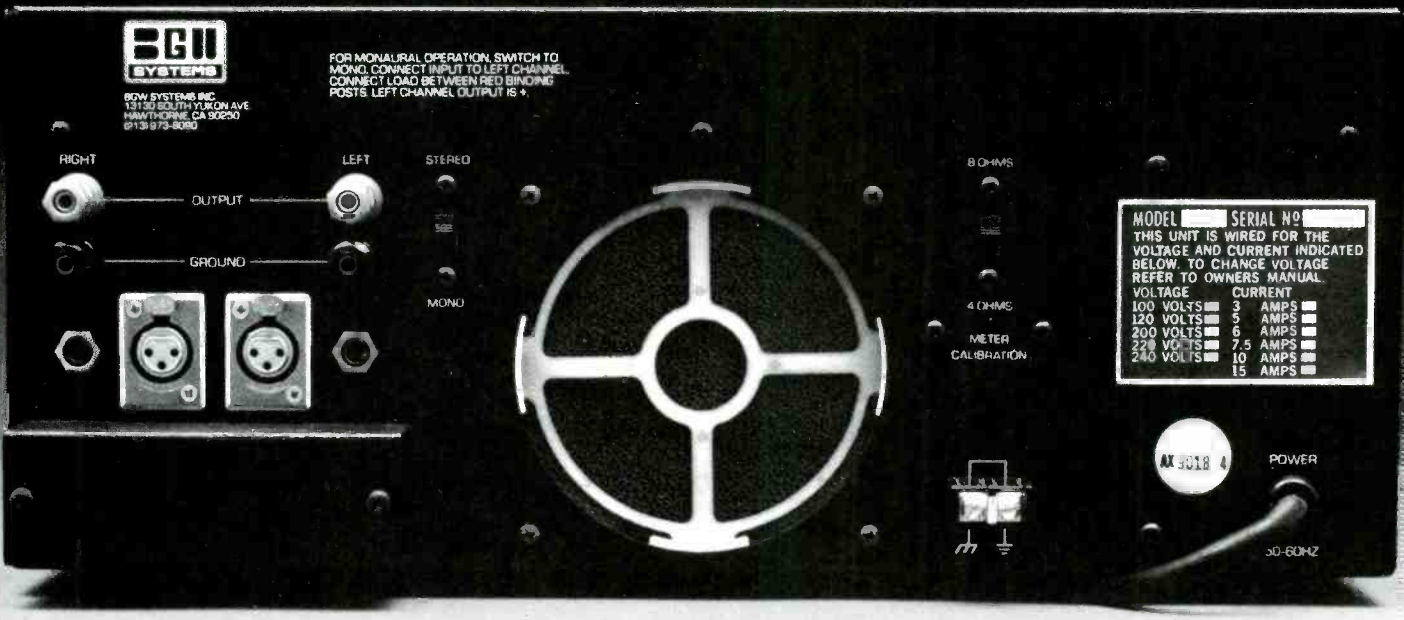

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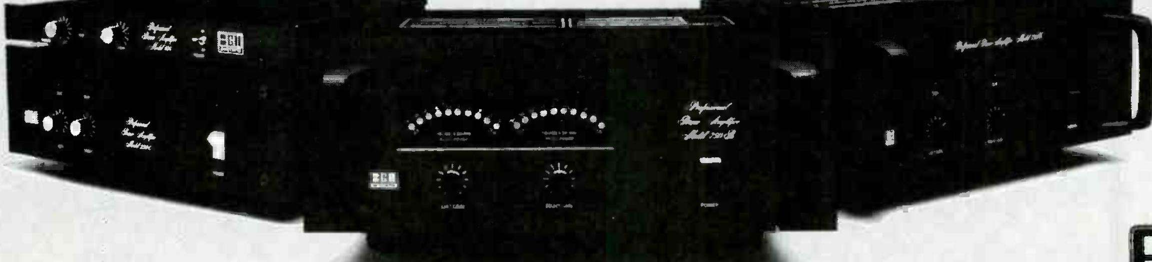

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the remote control system is, in fact, a complete computer. Don't let the lack of flashing lights fool you — the little card that plugs into the DDL could just as easily do your income tax with the proper programming and peripheral devices. (DO your taxes, not PAY them. That's next.)

The first decision to be made was: What did we want the remote control to do? As a remote control, it had to be capable of adjusting the user controls, specifically the delay settings, the DOUBLE (2X) mode which halves the clock frequency, and the REPEAT mode, which disables writing into the memory and thus "captures" a signal for continuous repetition. The other user controls are included by implication: setting the delay to zero is equivalent to pressing a reset switch. The switches were provided for convenience, but a controller can issue a command to ZERO as easily as any other, so it wasn't necessary. Also, we saw no need to provide a remote level control since this is rarely adjusted after initial setting. We mentioned earlier the possibility of reading the present settings of the peripheral. Although it could be done, the unit was not designed with this feature in mind and we decided that as it would require a significant modification of the main frame to do satisfactorily, we would not provide the capability. This is not too great a disadvantage since all outputs and modes announce themselves with readouts on the front panel. It should be noted that with the generalized computer architecture to be described, the remote control card can read and transmit data back to the controller with just a software change. Because of this general purpose structure, it is possible to write a program that will control the delay line even without a remote input. We identified two features which could be added to the DDL in this fashion. The first is a common one, automatic flanging. This can be done manually by setting one DDL output at a fixed point and varying another one about the same time delay. Mixing the two outputs generates the familiar comb filter effect called flanging. The other effect is a bit more subtle. One of the major uses of the DDL is for vocal and instrumental "doubling".

Doubling creates the illusion of the presence of multiple sources of the same signal, created by delaying the signal by a period of from 10 to several hundred milliseconds. The precise period depends primarily upon the type of program material; instruments with sharp transients such as guitars sound better with the shorter delays, sustained material such as violins and choral voices benefit from longer ones. Using delay lines to double (or triple) signals suffers to some extent from an artificiality caused by the fixed relationships of the delays. Live performers cannot match each other precisely, and so there is always a random variation between the original signal and its multiples. This can be simulated by randomly varying the delay of the various outputs. It must be done smoothly, quietly, and aperiodically to prevent worse problems than the artificiality thus eliminated. This is an ideal task for a microprocessor, which has plenty of computing capability, enough to generate random numbers, normalize them, and use them to control the DDL. Many other features could be added, mostly by modifying the software. For this reason, the remote control unit is supplied with the control program in an erasable ROM (Read Only Memory). If new software features seem desirable, the ROM can be erased and re-programmed. This is one of the best and cheapest ways of avoiding obsolescence known to audiodom.

The two new features, since they are new features, should be controllable without a computer. All features, new and remote, must be available to the computer. When power is turned on, the remote control automatically sets all delay line functions to their normal power-on states (all outputs 0 delay, no repeat, no 2X), and initializes itself to accept commands from the GP-IB. However, if certain switches on the remote are set, it will initialize to the flange or vocal double mode. To avoid switching power, a front panel RESET control is provided which will initialize the DDL in any of these states.

To see how the remote control works, refer to the block diagram (Figure 3). Note that most of the major circuit blocks, the CPU (Central Processing Unit), the PIA (Peripheral Interface Adaptor), and the GP-

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IB Interface are all individual integrated circuits, rather than groups of IC's as is common in block diagrams. These IC's are members of the Motorola 6800 microprocessor family. Intel also manufactures a set of chips which will perform the same job, as will, no doubt others. I'll explain why we chose the Motorola IC's in the software portion of this article. Block-by-block, we proceed:

THE BUS DRIVERS: These chips are the electrical interface between the actual external bus wiring and the MOS levels present within the remote card circuitry. These chips are basically bidirectional buffers, so that each section of the chip has an input and an output which may be interchanged on command from the GP-IB Interface chip. For instance, the data lines may be required to receive or transmit. The GP-IB Interface decodes the instructions from the CPU or from the GP-IB and instructs these chips in which direction to transmit.

THE GP-IB INTERFACE: This IC understands IEEE standard. It decodes the relatively complicated state transitions which occur on the bus and enables the CPU to read to, write to, and command the bus by writing bits into registers that form part of the address space of the CPU. Thus, if a data byte has been sent, the chip can notify the CPU, either by being interrogated, or by issuing an interrupt. The CPU can then determine the data and use them. The chip also has addressing logic built in. The bus address to which the remote control unit will respond is determined by the user by means of internal switch settings. The CPU reads this setting by interrogating the GP-IB interface, and then (after modifying the setting, if desired), sends it back to the chip. Thereafter, the Interface chip "knows" if a message is being sent to it, or to another device on the bus.

Beyond the scope of this article, but interesting to note in passing, are some of the other capabilities of this chip (in conjunction with the CPU). They include the ability to issue/respond to service requests, poll other peripherals to see if they need service, advise the CPU of the state of the bus, automatically complete (or refuse to complete) the bus handshake sequence, and choose to "interrupt" the CPU on any combination of conditions, such as whether data has been received, been sent, on end of data, and others more complicated. In this instrument, normal operation occurs when a data byte addressed to the Delay Line is received. Upon receipt, the GP-IB interface issues the accepted handshake and transmits the byte to the CPU. The CPU then determines its validity. Immediately after receipt, the interface is ready to handshake an another byte. Thus, assuming the CPU is free at the beginning of the sequence, two bytes can be accepted, in effect, immediately. If some procedure is taking a long time, the third byte in has no place to go, and handshake is delayed. This explains the somewhat cryptic reference to the third "ZERO" command earlier in the article. Much of the logic of the interface chip could be duplicated by the CPU and a few standard chips, or by a PIA (described later). The tremendous advantage of the IC is that it makes it unnecessary to understand the bus in great detail and even more important, unncesssary to write software to implement the bus functions, state transitions, etc. In other words, it enables any slob to implement a standard interface with which the entire world can communicate without prior arrangement.

THE CENTRAL PROCESSING UNIT (CPU): The CPU coordinates operation of the remote control by its ability to follow slavishly instructions which it is given. It has the ability to "fetch" an instruction from the ROM (Read Only Memory), and perform the instruction. The nature of some of these instructions is described under SOFTWARE, below.

The CPU is a 40 pin integrated circuit. The pins are employed thus: 16 pins, controlled by the CPU, form the ADDRESS. 16 bits can uniquely address 2 to the 16th power or about 65,000 bytes. Each of these bytes can contain an instruction, data, or a peripheral register, or perform other functions. The CPU can read or write into any of these locations. For instance, if it is reading a location containing an instruction, (and the CPU is looking for an instruction), it will perform the instruction. It could also read data, either from memory, or, as described earlier, from the GP-IB interface. It might be writing data, either to memory, to the bus, or to any other device. The data thus written will be interpreted according to the nature of the device. If it is written to memory, it will be stored. If it is written to the PIA, it will either configure the PIA (described later) or output data to the DDL. Eight additional pins on the CPU are the DATA BUS. This bus is bidirectional: depending upon the instruction the CPU is executing, data will be input or output (as described above). Another pin, R/W (Read/Write) tells the rest of the system what type of operation is being requested. The Motorola 6802 used in this unit contains enough Random Access Memory in the CPU to make an external RAM unnecessary. Typical microcomputer systems do contain RAM's and the R/W signal tells the RAM to read (CPU output, RAM listening) or to write (RAM output, CPU listening). This accounts for 25 out of 40 pins, and the power supply and timing signals account for 7 more. A few other signals of interest include: RESET, which unconditionally interrupts the CPU and tells it to fetch an instruction at a specified location; IRQ (Interrupt ReQuest), which tells the CPU "stop what you're doing now, save your status, and then look at a (different) specified location for an instruction; MNI (Non-Maskable Interrupt), which is identical to the IRQ line except that the instruction location is different, and the CPU must accept the interrupt. A few other lines such as HALT and BA (Bus Available) are primarily for larger system applications and we sha'n't go into them here.

The READ ONLY MEMORY (ROM) is the source of instructions for the CPU. A ROM contains many sequential addresses which contain, in the proper sequence of 8-bit bytes, complete instructions for the operation of the systems. The job of the programmer is to place the proper sequence of bytes into the ROM. As mentioned earlier, the ROM used in this unit is erasable. Data are stored "permanently" by charging certain physical locations within the chip. These charges are insulated by a dielectric and cannot leak away from their site. Unlike the other memories in the DDL, the ROM is "non-volatile", which means that if you turn off the power, the data don't disappear. They can, however, be removed by exposing the ROM, which has a quartz lid instead of the normal opaque one, to short-wave ultraviolet. This is invaluable in developing programs, since it is not necessary to throw away a chip (or worse, wait eight weeks) each time a change must be made. A program step may require from 1 to 3 bytes of ROM, depending upon the nature of the instruction. For instance, an instructrion telling the CPU to "add accumulater A to accumulater B" requires 1 byte because all that is needed is the instruction. The CPU knows where the accumulaters are located. A 2 byte instruction includes "load accumulater with hexadecimal data '2F'". The first byte says that the NEXT byte contains data to be loaded in the accumulater. Three byte instructions are memory references. The first byte is the instruction, the next two bytes contain the address (1 byte, 8 bits can only define 256 locations and there are 65,000 available). A typical instruction might be "store data in accumulater A into memory location 'A1BE'".

Again, the first byte is the instruction itself, and the next two bytes contain the address. Most 8 bit combinations represent valid CPU instructions, and, of course, data bytes can contain any combination of bits. For this reason, trying to read and understand the raw "machine language" code stored in a ROM (also called "object program") is an exercise in futility. Generating anything other than the simplest of programs requires the use of another program, called and "asembler". More on this shortly.

The PIA (PERIPHERAL INTERFACE ADAPTOR) is the last of the large chips used in the remote control. This chip connects to the CPU data bus and, upon instruction from the CPU, stores data in its internal registers and applies them to its sixteen output lines. The PIA has two sections, each 8 bits wide, and each section has three registers. Since both sections are similar, we will discuss only one. The DATA register stores data written by the CPU or forwards peripheral data to the CPU depending upon the data stored in the DATA DIRECTION REGISTER. In this unit, all data is going from the CPU towards the DDL. Changing the setting in the direction register changes the outputs to inputs and vice versa. The CONTROL REGISTER determines operational and handshake characteristics of the PIA. For instance, the control register can enable an automatic handshake upon receipt of data, or require the CPU to generate an output data available signal. In this system, the PIA is configured so that when data are written into the data registers, a strobe is immediately initiated. This strobe enables a synchronizing signal which enables the tri-state buffers on the next timing pulse from the DDL. Two output strobes are available, one for each section of the PIA. One is used as just described. The other strobe is used to store data in an auxiliary latch (block diagram "control logic") which controls the 2X, REPEAT modes, and the error indicator.

The TRI-STATE BUFFERS can best be explained in the context of the Delay Line operation. Delay is achieved by writing data into a Random Access Memory at a fixed rate and reading it at the same fixed rate, but at a different address. The write address is decremented every time a write operation is performed. Thus, if a read is performed two addresses higher than the write, the data read will be the same as the data stored two sample time intervals previously. The delay is numerically equal to the difference

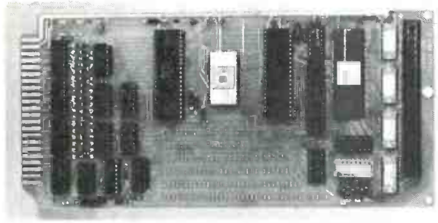


Figure 4: THE DDL INTERFACE

between the write address and the read address multiplied by the sampling period. The delay line can have up to five outputs (four if the remote control is installed). To avoid unnecessary wiring, a bus oriented addressing system was developed. The address bus is connected in parallel to counter registers on each output board, thru "TRI-STATE" buffers. These buffers have outputs which follow the input logic levls, unless they are OFF, in which case the output assumes a high impedance state. The buffers on the remote control card are identical to those on the DDL output cards. When data are written into the RAM, all of the buffers are off and pull-down resistors assure that all address bus levels are logic 0. After the write operation, a pulse is sent to the first output card, which enables its tri-state buffers. Thus, the contents of the first output's address register is placed on the address bus, and the DDL memory is read at this address. At the conclusion of this operation, the first output sends a pulse to the SECOND output, telling it to send out its address data. This proceeds sequentially until the pulse reaches the remote control. At this time, the remote card's tri-state buffers are enabled, and the address data placed on the bus. A strobe from the remote card to the selected output card loads the address data into the output card address register, replacing data previously stored manually.

PROGRAMMING

Having completed the electrical description (as we have just done, if you didn't notice,) of the remote control card, we would normally be able to say, "Now we UNDERSTAND." With microcomputer units, however, I'm afraid that's not the case at all. No, we still have to PROGRAM the system. Programming is determining, precisely and completely, what sequence of bytes to place in the control ROM so that the system will do its job when plugged in. And this, believe it or not, is more of a job than the hardware design. As an example, consider the fact that the electrical design of this unit was completed within a matter of a day or so. The designer simply sat down, drew a few blocks for the CPU, GP-IB, etc., and connected them as detailed in the Motorola Manuals. The only unusual (and by no means complex) circuitry involved the buffers between the 5 volt microcomputer logic and the 15 volt delay line logic. But the program! That's another matter indeed. As stated earlier, machine language programs are virtually impossible to generate by humans. The choice of the Motorola

Microprocessor family was dictated by the availability of a "cross-assembler". This is a program which will run on one computer and generate a program for another computer. Eventide possesses a Hewlett-Packard 9825 computer, which is a desk computer with rather nice scientific and input/output routines, and the capability of substantial storage on magnetic tape or disc. The cross-assembler is a program which runs on the 9825, and produces machine language programs for the 6800 series. The language used by the cross-assembler is, reasonably enough, "assembly language". If you are familiar with standard computer languages, you will appreciate that it takes from five to hundreds of assembly language statements to form instructions equivalent to one line of BASIC, FORTRAN, or whatever. Since the program that allows a larger computer to use a so-called "high level" language requires over ten times the total storage required by the remote control card operating program, it is obvious that there was little real alternative to assembly language.

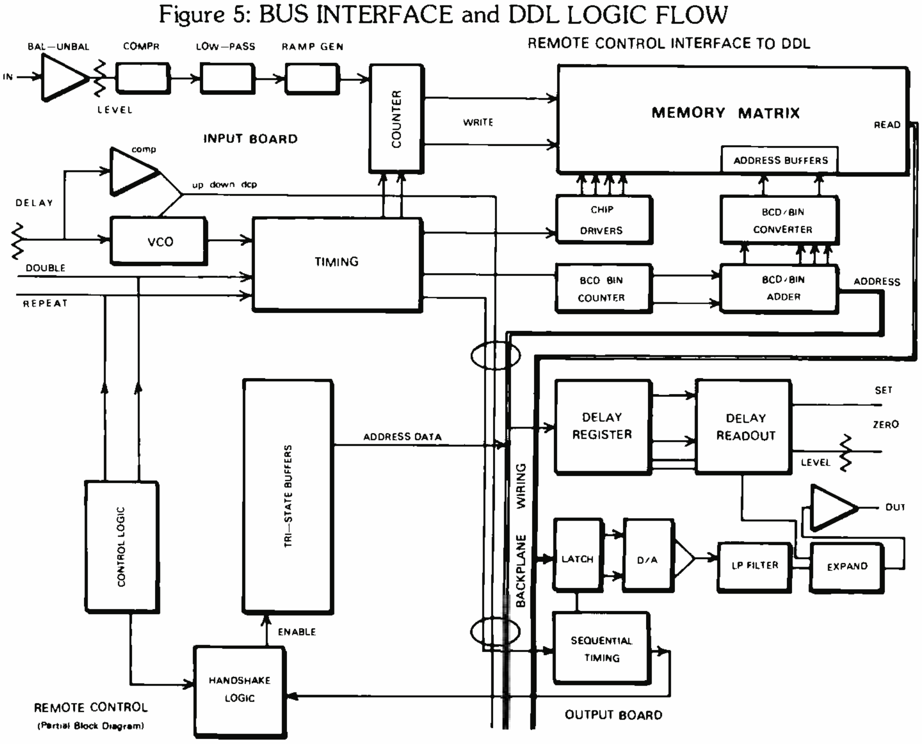
In the section describing the ROM, I used an example of a single byte instruction "add accumulater A to accumulater B". This is awkward to write or say, but, abbreviated "ABA" on the larger computer, the proper machine language code would be generated. Of course, there's no real trick to this: you could do the same thing by looking up the "op-code" for "ABA" in the tables furnished. The advantages of assembler over hand coding include the use of symbolic labelling, automatic computation of "branch addresses", automatic calculation and conversion of binary, octal, and hexadecimal constants. And, of course, it makes the program readable, since it is possible to distinguish instructions from data and, if good practice is followed, the program is "commented" so that other programmers can figure out what has been done. Other features include the capability to transfer the machine language program to a ROM programmer without human intervention or mistakes, and the ability to copy the program and link several programs on a new tape. Too much jargon in the above sentences? Symbolic labelling allows one to assign a name or "label" to a particular line of program, or to a particular address. For instance, the GP-IB address bus might have an absolute hexadecimal address of 8004. A real program has many registers, and an attempt to remember the numbers would be futile, and to use them would lead to mistakes. But, if you issue the statement "ADRREG EQU $8004" at the beginning of your program, then you can later say "STA A ADRREG", which means "store the contents of register A in hex location 8004".

More jargon: "Automatic computation of branch addresses". Let's say you want the remote control to cause the delay line to enter the REPEAT mode. You send the command "R" on the interface bus. The GP-IB Interface receives the "R" and transfers it to the CPU. The CPU knows it has received a command, but how does it know what command, or what to do with it? The (abbreviated) routine goes as follows: Load accumulater with data byte. Does data=ASCII "R"? If so, branch to the line labelled "REPEAT", otherwise continue. "REPEAT" is a symbolic address in the program located at a random location, and is the first step of the instructions that tell the DDL to enter the repeat mode. If the byte sent was "R", the program branches to this location. The assembler calculates the location from its knowledge of the symbolic addresses and the present location of the "branch if equal" instruction. If the byte was not "R", by the way, it would compare with the next valid instruction, "S", and branch to "S" if that were the correct byte. If not it would keep going until it reached the end of the valid instructions and branch to the error routine, which is the machine language code which turns on the front panel error light. Note a particularly important feature: If you make any change in the program, the probability is quite high that a large number of branch addresses would have to be changed. The assembler does this automatically. If it were done by hand, each change would engender more errors than it might correct! The automatic conversion of number bases is quite handy for the following reasons: By convention and convenience, microprocessors are programmed in a hexadecimal number base. It includes the digits 0 through 9, and A through F. A=decimal 10, and F=decimal 15, thus 10Hex=16Decimal, and FFHex=255 Decimal. But, let's say you want to use a binary number so that bit positions are clearly shown.

If you know that the bit that sets the REPEAT mode is the sixth bit, you could say "LDA A #%01000000", which shows at a glance that the sixth bit is set (they're numbered 0 through 7). You could also say "LDA A #$80" or even "LDA A #128", but the first indicates that you wish to set a bit rather than enter a constant, even though the final code is identical. The prefix "#" indicates "immediate mode" to the assembler, which means it should load the data from the ROM rather than the A memory location defined by the data byte. The prefix "%" indicates binary, "$" indicates hexadecimal. No prefix indicates decimal. One can also use ASCII, such as "LDA A #'R", which loads the accumulator with the value of ASCII "R".

Without drawing a complete flow-chart of the remote control program, I'll give an idea of how it is structured. When power is turned on or RESET depressed, the program goes to a routine which sets all the registers to their cleared state, except for a few which require other settings. It reads the address switches as the user has set them and tells the GP-IB that this is its address. It then waits for a command. When a command is received, it follows the branching routine described earlier to the particular segment of machine code designed to effect the command. If a command involving setting an output is received, the program jumps to an input routine that expects a group of digits. Depending upon the digits or lack thereof, it decides how to set the output indicated. If the number is too high, it will set delay for maximum. If the command makes no sense at all, it will branch to error routine without doing anything. Once a numerical command is complete, it will perform an arithmetic routine which will convert the ASCII Binary Coded Decimal information received to the mixed format of absolute Binary/BCD required by the DDL address bus, and finally begin the handshake

Figure 5: BUS INTERFACE and DDL LOGIC FLOW



sequence with the DDL.

Refer to Figure 6 for a brief section of commented commented assembly language. Readers interested in more details of the various type of computer languages are referred to their local hobby computer shop, the library, and the old stand-by, manufacturers' literature. Of the three, the hobby shop will have the best selection, beholden as it is to no one semiconductor maker, and, of course, the manufacturers' literature will be the best bargain!

SEMI-DISCLAIMER

There are small (I hope) factual errors and omissions in this article. Those familiar with the Motorola Microprocessor family and GP-IB structure will note that I have totally ignored signal inversions and bus states. This was done strictly for clarity, as this is clearly not a description for the design engineer. Likewise, in the programming section, I have called the program group commonly known as the Editor/Assembler the Assembler, again to avoid unnecessary jargon and to assure comprehension.

The main purpose of the article is to promote consideration of the IEEE 488-1975 interface standard as a universal control interface between separate manufacturer's units. While the standard is perhaps not the best that can be defined, and is certainly not instantly comprehensible, it is STANDARD, of international scope, and widely supported by multi megabuck companies. I know it is customary to write Letters to the Editor following publication of articles of this nature. I would hope that those criticizing this suggestion remember that any counter suggestion they propose must engender agreement amongst many groups whose ECONOMIC interests are at variance. Anything which gives any company in this highly fragmented industry a significant advantage will be fought claw and pseudopod.

Figure 6:

EXAMPLE OF TIMING LOOP USING SYMBOLIC ADDRESSING:

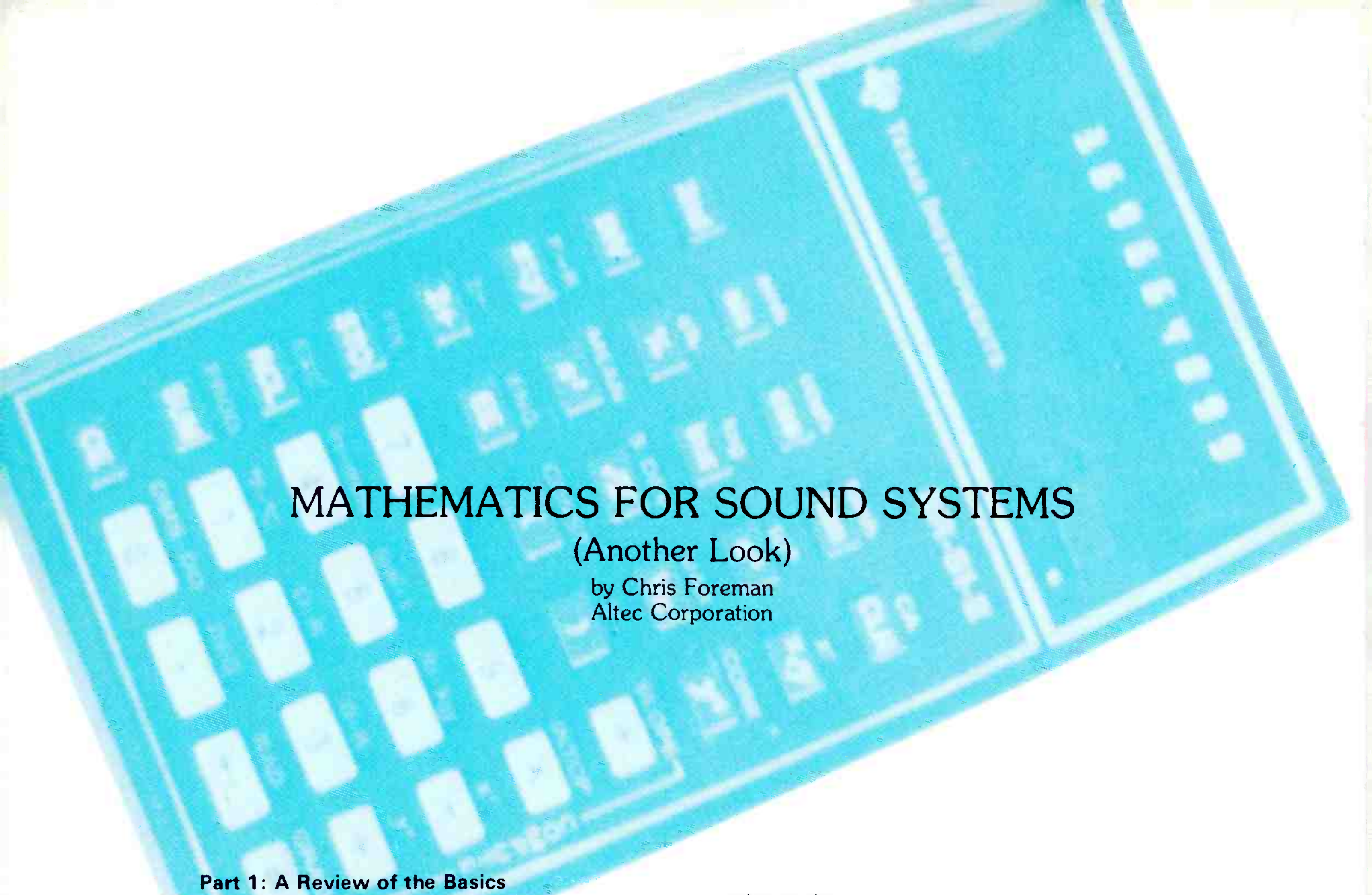
LINE 1010 Loads Register B with all ones
LINE 1020 Labelled "TMLOOP" subtracts 1 each time it is executed
LINE 1030 Compares Register B with data 00 A status bit is set if they are
LINE 1040 The status bit is checked and the program branches back to TMLOOP if it isn't set (BNE) means 'Branch if Not Equal'
This routine must be executed 256 times before the program can continue causing a deliberate time delay

EXAMPLE OF ASSEMBLY LISTING

LINE NUMBER FOR SEQUENTIAL NUMBERING OF PROGRAMMING STEPS
ACTUAL ADDRESS AT WHICH BEGINNING OF INSTRUCTION APPEARS
MACHINE CODE INSTRUCTION (OP CODE)
DATA OR ADDRESS OF DATA TO BE OPERATED UPON OPERAND

```
                          .. ORZX    (NOTE SIL
                         CLR A
0960 E096 BA A000        ORA A  DBLREG    DOUBLE MODE TO ACCU..
0970 E099 BA A001        ORA A  RPTREG    REPEAT MODE TO ACCUMUL,
0980 E09C 8A 80          ORA A  #%10000000 ENABLE THREE STATE TO AL
0990 E09E B7 800A        STA A  PIARB     STROBE LATCH, ENABLE OUTPU
1000 E0A1 88 80          EOR A  #%10000000 OROP THREE STATE ENABLE
1010 E0A3 C6 FF          LDA B  #$FF      ALL ONES IN B
1020 E0A5 5A      TMLOOP DEC B
1030 E0A6 C1 00          CMP B  #$00      HAS B REACHED ZERO YET?
1040 E0A8 26 FB          BNE    TMLOOP    IF NOT DECREMENT AGAIN
1050 E0AA B7 800A        STA A  PIARB     DISABLE THREE STATE
1060 E0AD 7E E02B        JMP    LBL5      WAIT FOR NEW DATA
1070 E0B0 7F A002 ZERO   CLR    ZERREG
1080 E0B3 7F 8008        CLR    PIARA     ZERO FIRST 8 BITS OF OUTPUT
1090 E0B6 B6 A002 NXTOUT LDA A  ZERREG    WHICH OUTPUT?
1100 E0B9 B7 800A        STA A  PIARB     CLEARS REPEAT AND DOUBLE
1110 E0BC 86 3C          LDA A  #%00111100
1120 E0BE B7 800B        STA A  PIACRB    DISABLE CB2 STROBE
1130 E0C1 7F 800A        CLR    PIARB     ZERO MOST S
1140 E0C4 86 34          LDA A  #%00110100
1150 E0C6 B7 8009        STA A  PIACRA
  60 E0C9 86 3C          LDA A  #
     CB B7 8009          ST
```

COMMENT FOR CLARITY OR FUTURE REFERENCE
DATA OR ADDRESS, CONVERTED TO ONE OR TWO BYTE OPERAND BY ASSEMBLER
LINE LABEL FOR FUTURE REFERENCE
INSTRUCTION MNEMONIC-CONVERTED TO OP CODE AT LEFT BY ASSEMBLER



MATHEMATICS FOR SOUND SYSTEMS
(Another Look)

by Chris Foreman
Altec Corporation

Part 1: A Review of the Basics

Why review mathematics for sound systems? Haven't we all seen enough explanations of "the dB" and such? Probably! But consider that a great deal of sound system design and analysis ends up being done by an artistic process. We often call this "engineering judgement", or less formally, "seat of the pants". Most of us would agree that engineering judgement gets better when the engineer understands the quantitative as well as the qualitative side of engineering. Obviously, nothing is more basic to quantitative engineering than mathematics.

In addition, most mathematics reviews lately are calculator-oriented and may not do much more than tell us which buttons to push. The handbook that comes with our calculators should tell us that much.

Thus, this review assumes that we already know how to use our calculators. I will concentrate on explaining why we are pushing the buttons, and how to get to the point where button pushing is the only thing left to do!

The review progresses in a logical order, starting with exponents of numbers, then covering logarithms and the dB, and finishing with a review of basic trigonometry. Each subject, except the trigonometry review, can be considered to be a prerequisite for the following subject. Thus, they should be covered in the order presented. I have numbered important formulas so that you can refer to them as necessary. The entire review and all the examples given are as oriented towards sound system design and analysis as I could make them.

EXPONENTS OF NUMBERS

In the expression Y^a, the number a is an exponent. Examples:

$Y^1 = Y$; $Y^2 = Y \times Y$

$Y^3 = Y \times Y \times Y$ and so on.

For negative, zero and fractional exponents, the following rules apply:

1) $Y^0 = 1$ for any value of Y

2) $Y^{1/a} = \sqrt[a]{Y}$

3) $Y^{-a} = 1/Y^a$

4) $Y^{-1/a} = 1/\sqrt[a]{Y}$

Example #1:

$Y^{1/2} = \sqrt{Y}$

$Y^{1/3} = \sqrt[3]{Y}$

$Y^{-2} = 1/Y^2$

$Y^{-1/2} = 1/\sqrt{Y}$

The following are rules for manipulation of expressions involving exponents:

5) $Y^m \times Y^n = Y^{(m+n)}$

6) $Y^{(m)^n} = (Y^n)^m = Y^{(mn)}$

7) $(XY)^n = X^n Y^n$

8) $Y^n/Y^m = Y^{(n-m)}$

9) $Y^m/Y^n = 1/Y^{(n-m)}$

10) $(X/Y)^n = X^n/Y^n$

Example #2:

$Y^2 \times Y^3 = Y^5$

$(Y^2)^3 = Y^6$

$Y^5/Y^3 = Y^2$

SCIENTIFIC AND ENGINEERING NOTATION

These are two similar methods for expressing a large number as a smaller number times 10 to some exponent (times a power of ten). For Scientific Notation:

$2{,}900{,}000 = 2.9 \times 10^6$

$28{,}500{,}000{,}000 = 2.85 \times 10^{10}$

$342 = 3.42 \times 10^2$

Introducing the first 16 track 1" recorder/reproducer in the world. The new 90-16.

Now you can make money while you save money.

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And here's how you save. You don't pay studio time. You don't pay for 2" equipment. And you don't pay the heavy hit of "outboard" DBX which usually cost about $300 per channel. Our suggested retail price of just $16,000* *includes* integral DBX interface.

If you're an engineer, the 90-16 Function Select panel will make a lot of sense to you. One button operation simultaneously switches three interrelated functions: tape/source, playback/record, and DBX decode/encode thus eliminating the need for three arms. If you're a musician, this human engineering factor means you won't have to wait for the engineer and lose the spontaneity of your music in the process.

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(714) 645-8501

Sound Genesis
2001 Bryant St.
San Francisco, Ca. 94110
(415) 285-8900

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Nashville, Tenn. 32703
(615) 256-1650

Audio By Zimet, Inc.
1038 Northern Blvd.
Roslyn, New York 11576
(516) 621-0138

Lebow Labs
424 Cambridge St.
Boston, Mass. 02134
(617) 782-0600

SPECIFICATIONS

Tape Speed: 15 ips

Line Input: −10 dB (0.3V) impedance: greater than 20K Ohms, unbalanced*

Line Output: −10 dB (0.3V) load impedance: greater than 10K Ohms, unbalanced*

Record Level Calibration: O VU referenced to 3 dB above 185nWb/m tape flux

Wow And Flutter
0.03% RMS (NAB), weighted

Overall Frequency Response, Sync Mode
40 Hz – 18 KHz, ± 3 dB

Signal To Noise Ratio (With DBX)
95 dB weighted, 90 dB unweighted, referenced to 3% T.H.D. level (9 dB above O VU) at 1 KHz

Distortion
1% at 1KHz, O VU

Cross Talk
greater than 45 dB at 1000 Hz

Dimensions, Overall
24⅞" (W) x 47¼" (H) x 28 5/16" (D) (Console included)
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$0.000045 = 4.5 \times 10^{-5}$
$0.051 = 5.1 \times 10^{-2}$

Engineering Notation follows the same rules as Scientific Notation using only powers of 10 that are divisible by three:

10^{-12} = pico (as in picofarads)
10^{-9} = nano (as in nanovolts)
10^{-6} = micro (as in microfarads)
10^{-3} = milli (as in millivolts or millimeters)
10^{3} = kilo (as in kilohertz or kilometers)
10^{6} = mega (as in megahertz)
10^{9} = giga (as in gigahertz)
10^{12} = tera

Example #3, Engineering Notation:

$2{,}900{,}000 = 2.9 \times 10^{6}$
$28{,}500{,}000{,}000 = 28.5 \times 10^{9}$
$342 = 0.342 \times 10^{3}$ or $= 342 \times 10^{0}$ but probably $= 342$
$0.000045 = 45 \times 10^{-6}$
$0.051 = 51 \times 10^{-3}$

Note that the difference between these and the previous examples for Scientific Notation is that all exponents are divisible by three.

LOGARITHMS

Given a **base** B and a **number** Y, the **logarithm** of Y is some number, a, chosen to solve the following equation:

11) $Y = B^{a}$

A common way to express the solution to this equation is:

12) $a = \log_B Y$ read "a equals the log to the base B of Y".

The base B can be any real number, but is usually either 10 or "e", an irrational number approximately equal to 2.718.

If the base is 10, the equation is usually written:
$a = \log Y$ (when no base is given, base 10 is assumed)

If the base is e, the equation is usually written:
$a = \ln Y$ (base e, also called natural logarithms).

The **antilog** of a number is the number represented by the logarithm. Thus in the above equation, $a = \log_B Y$, Y is the **antilog** of a.

The following are rules for manipulating logarithmic expressions:

13) $\log MN = \log M + \log N$
14) $\log M/N = \log M - \log N$
15) $\log N^{c} = c \times \log N$
16) $\log N^{1/c} = (1/c) \log N$
17) $B^{(\log_B N)} = N$

Many years BC (before calculators) we used tables of logarithms to multiply and divide large numbers. For completeness, the following formulas are presented:

18) $M \times N = \text{antilog}(\log M + \log N)$
19) $M/N = \text{antilog}(\log M - \log N)$

According to legends, there was an ancient device called a "slide-rule" that consisted of two sliding scales which added or subtracted logarithms and thus multiplied or divided real numbers (at least its batteries wouldn't wear out).

THE DECIBEL (dB)

The term dB, which stands for decibel, expresses a **ratio** in terms of a base 10 logarithm. The dB notation allows us to represent very large ratios with small numbers. For example, the sound pressure level of the loudest sound we can tolerate is approximately 1,000,000 times the sound pressure level of the softest sound we can hear. This same statement using the dB notation would read: The loudest sound we can tolerate is approximately 120 dB higher in sound pressure level than the softest sound we can hear. It is obviously easier to work with and to comprehend the smaller number expressed in dB. This ability to represent large ratios with small numbers is the primary reason we use the dB notation in audio work.

Power RATIOS in dB

The ratio in dB of two power levels (acoustic power, electrical power, horsepower or any type of power) is equal to 10 times the base 10 logarithm of their simple numeric ratio:

20) $\text{dB} = 10 \log P_1/P_2$

Voltage or SPL RATIOS in dB

The ratio, in dB, of two voltages or sound pressure levels is equal to 20 times the logarithm of their simple numeric ratio:

21) $\text{dB} = 20 \log V_1/V_2$
22) $\text{dB} = 20 \log SPL_1/SPL_2$

10 Log vs 20 Log dB Formulas

It may be useful to explain why there are both "10 log" and "20 log" dB formulas. The dB notation was originally defined for power type quantities. Neither voltage nor SPL are power type quantities. Power can be calculated from voltage, however:

$P = V^2/R$ (where R is resistance in ohms)

Placing this back into formula 19), we find:

$\text{dB} = 10 \log (V^2/R)_1/(V^2/R)_2$

Cancelling the R value in the numerator and denominator leaves:

$\text{dB} = 10 \log (V^2)_1/(V^2)_2$

From rule 15), this becomes:

$\text{dB} = 2 \times 10 \log V_1/V_2$

Which is the same as saying:

$\text{dB} = 20 \log V_1/V_2$

A similar discussion would apply to SPL. This shows that both of the formulas for dB are actually based on **power ratios**. When we cancelled the R in the equation above, we assumed that the value for R in the numerator and in the denominator were the same. If they are different, the 20 log equation for voltages in dB doesn't actually apply. However, we often ignore any difference in R values and use the 20 log equation anyway. Provided that we do not try to relate the dB values gained in this manner back to power ratios, we won't get into trouble.

Example #4, Power RATIOS in dB:

Find the ratio in dB of 100 watts to 50 watts.

$\text{dB} = 10 \log 100/50$
Answer: +3 dB

Note that this means that 100 watts is 3 dB above 50 watts. If we had compared 50 watts to 100 watts ($\text{dB} = 10 \log 50/100$), the answer would have been −3 dB. Similarly, any time the ratio of two powers is 2:1, their ratio in dB is +3 dB. When the ratio is 1:2, the ratio in dB is −3 dB.

Example #5, Voltage RATIOS in dB:

Find the ratio in dB of 100 volts to 50 volts.

$\text{dB} = 20 \log 100/50$
Answer: +6 dB

Note that a ratio of 2:1 in voltage means a ratio in dB of +6 dB. If the ratio is 1:2, the ratio in dB is −6 dB.

Example #6, SPL RATIOS in dB:

SPL ratios, expressed in dB, are similar to voltage ratios. For example, two SPL levels (in dynes per square centimeter) with a numeric ratio of 2:1 would have a ratio in dB of +6 dB.

RATIOS vs Specific LEVELS

All of the above examples concern **ratios**. The term "dB" **always** expresses a ratio. To express a single, specific level in dB (such as 95 dB SPL or +4 dBm) there must be a **reference** level. This reference level takes the place of the denominator in each of the above dB equations. Specific power **levels** (as opposed to power **ratios**), voltage levels and

SPL levels may be expressed in dB notation when the appropriate **reference** level is understood. Several accepted reference levels for power, voltage and SPL are discussed below.

Power LEVELS in dBm:

dBm is an accepted way of expressing a specific power level. **The reference quantity for dBm is 1 milliwatt.** If we are given a value in watts and wish to find the value in dBm, we use the following formula:

23) dBm = 10 log ? watts/1 milliwatt

Example #7, Power LEVELS in dBm:

Find the level in dBm of 1 watt:

dBm = 10 log 1 watt/1 milliwatt
Answer: +30 dBm

Voltage LEVELS in dBV:

dBV is an accepted way of expressing a specific voltage level. The reference quantity for dBV is usually 1 volt. If we are given a value in volts and wish to find the value in dBV we use the following formula:

24) dBV = 20 log ? volts/1 volt

Example #8: Voltage LEVELS in dBV:

Find the level in dBV of 10 volts.

dBV = log 10 volts/1 volt
Answer: +20 dBV

Sound Pressure Levels in dB SPL:

dB SPL is an accepted way of expressing a sound pressure level. The reference quantity for dB SPL is 0.0002 dynes per square centimeter. If we are given an SPL value in dynes per square centimeter and wish to find the value in dB SPL, we use the following formula:

25) dB SPL = 20 log ? dynes/cm^2/0.0002 dynes/cm^2

Example #9: SPL Levels in dB SPL:

Find the value in dB SPL of 2.0 dynes/cm^2.

dB SPL = 20 log 2.000 dynes/cm^2/0.0002 dynes/cm^2
Answer: 80 dB SPL

How To Go Backwards

So far, we have discussed methods for finding dBm, dBV, or dB SPL when the value in watts, volts or dynes per square centimeter was known. Now suppose that we have a value in dBm, dBV or dB SPL and want to convert backwards to watts, volts or dynes per square centimeter. How do we go about this?

The way we find the answers to this question, and the general method for finding the answer to any type of dB related problem when the answer is not clear, is to return to the appropriate original dB formula and manipulate it as needed.

Finding Watts When the RATIO in dB is Given:
Example #10:

If P_1 is +5 dB above (higher in power than) 50 watts, find P_1.

In this example, we are given a value in watts, and we are told that the unknown P_1 is +5 dB above the known value. It is important to realize that the term "+5 dB" implies a **ratio** (+5 dBm would have implied a specific **level**). Thus, we will use the original formula for the **ratio** of two powers:

20) dB = 10 log P_1/P_2

Now insert all of the known values into formula 20):

+5 = 10 log ? watts/50 watts

Now we will manipulate this equation to isolate the unknown value on one side. First, divide both sides of the equation by 10. This gives us:

1/2 = log ? watts/50 watts

Now, use each side of this equation as an exponent of 10:

$$10^{1/2} = 10^{(\log ?\ \text{watts}/50\ \text{watts})}$$

Next, apply formula 17):

17) $B^{(\log_B N)} = N$ thus:

$$10^{(\log ?\ \text{watts}/50\ \text{watts})} = ?\ \text{watts}/50\ \text{watts}$$

and therefore: $10^{1/2} = ?$ watts/50 watts

Multiplying both sides of this equation by 50:

$$50 \times 10^{1/2} = ?\ \text{watts}$$

Thus, ? watts = $50\sqrt{10} = 158.1$ watts

It is now possible to develop a formula that will allow us to find the answer to problems of the above type without going through all the mathematical manipulations.

26) $?\ \text{watts} = P \times 10^{(dB/10)}$

Where ? watts is the unknown power level, P is the known power level and dB is their ratio in dB. If the unknown power level is **below** the known power level, the value in dB must be negative (such as −6 dB).

Example #10 Worked Again Using Formula 26):

If P_1 is +5 dB above (higher in power than) 50 watts, find P_1.

26) $?\ \text{watts} = P \times 10^{(dB/10)}$

Inserting all the known values:

$?\ \text{watts} = 50 \times 10^{(5/10)}$

Thus, $?\ \text{watts} = 50 \times 10^{(1/2)} = 50 \times \sqrt{10} = 158.1$ watts

Obviously, formula 26) makes Example #7 much easier to work. The reason we worked Example #7 the hard way the first time was to understand the derivation of formula 26). As we will see, there are a great number of possible formulas that can be applied to a great number of different problems. However, if we understand the original formulas for dB ratios and for dB levels (formulas 20 through 25), and understand the basic algebraic manipulations discussed under Exponents and Logarithms, we can **derive** whatever formulas we need to solve a given problem. In fact, it should actually be easier for us to derive any modified formulas we need than to carry around a copy of this article and try to decide which formula (or which modified formula) to use with which problem!

In order to simplify the next few examples, however, several more modified formulas are presented. Each of these formulas was **derived** from the appropriate original dB formula using some algebraic manipulation of the type done in Example #10.

Finding Watts When the LEVEL in dBm is Known:

Formula 27), below, allows us to find a value in watts for a given value in dBm. Note that the known power P in formula 26) is replaced with 0.001 watts (1 milliwatt).

27) $?\ \text{watts} = 0.001 \times 10^{(dBm/10)}$

Also note that 0.001 watts is the **reference** level for dBm as explained for formula 23) earlier.

Example #11, Finding Watts from dBm:

Find the value in watts of: a) +24 dBm; b) +50 dBm; c) −50 dBm.

a) Inserting the known values into formula 26):
$?\ \text{watts} = 0.001 \times 10^{(24/10)}$
Answer: 0.25 watts

b) $?\ \text{watts} = 0.001 \times 10^{(50/10)}$
Answer: 100 watts

c) $?\ \text{watts} = 0.001 \times 10^{(-50/10)}$
Answer: 10 nanowatts (10×10^{-9} watts)

Finding Volts When the RATIO in dB is Known:

Formula 28), below, allows us to find an unknown value in

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volts when we know another value in volts, and we know the ratio between these two values in dB.

28) $? \text{ volts} = V_1 \times 10^{(dB/20)}$

Example #12:

If V is −4.5 dB below 2.5 volts, find V.

Inserting all the known values into formula 28):

$V = 2.5 \times 10^{(-4.5/20)}$

Answer: V = 1.49 volts

Finding Volts When the LEVEL in dBV is Known:

Formula 29), below, allows us to find a value in volts for a given value in dBV. Note that the known voltage V in formula 28) is replaced with 1 volt.

29) $? \text{ volts} = 1 \times 10^{(dB/20)}$

Example #13:

If V is equal to −56 dBV, find the value of V in volts.

Inserting all the known values into formula 28):

$V = 1 \times 10^{(-56/20)}$

Answer: V = 1.58 mV (1.58×10^{-3} volts)

Finding Dynes per Square Centimeter When the RATIO in dB SPL is Known:

Formula 30), below, allows us to find an unknown value in dynes per square centimeter when we know another value in dynes per square centimeter, and we know the ratio between these two values in dB.

30) $? \text{ dynes/cm}^2 = SPL_1 \times 10^{(dB/20)}$

Since this formula will probably not be used too often, no example is included.

Finding Dynes per Square Centimeter When the LEVEL in dB SPL is Known:

Formula 31), below, allows us to find an unknown value in dynes per square centimeter for a given value in dB SPL.

31) $? \text{ dynes/cm}^2 = 0.0002 \times 10^{(dB/20)}$

Example #14:

If L is equal to 94 dB SPL, find the value of L in dynes/cm^2.

Inserting all the known values into formula 30):

$L = 0.0002 \times 10^{(94/20)}$

Answer: L = 10.0 dynes/cm^2

Variations on a Theme:

In the above equations for dBm, dBV or dB SPL, if we place the reference quantity in both the numerator and the denominator, the value in dBm, dBV or dB SPL comes out to be 0 (zero):

dBm = 10 log 1 milliwatt/1 milliwatt or 1 mW = 0 dBm
dBV = 20 log 1 volt/1 volt or 1 volt = 0 dBV
dB SPL = 0.0002 dynes/cm^2/0.0002 dynes/cm^2
or 0.0002 dynes/cm^2 = 0 dB SPL

THE dBm DILEMMA

Now that we know all about dBs, we ought to be able to decipher some of the specifications we read on catalog sheets for products such as mixers and other electronics. What, for example, is the maximum output of a mixer rated at "+18 dBm maximum output level into 600 ohms"? From formula 27), this must be:

$? \text{ watts} = 0.001 \times 10^{(18/10)}$

Answer: 63 mW

Let's try this again for a mixer rated at "+18 dBm maximum output level", but also rated to drive a "minimum 10,000 ohms load impedance".

$? \text{ watts} = 0.001 \times 10^{(18/10)}$

Answer: 63 mW

But how can this be? Can a mixer which can only drive a 10,000-ohm load have the same power output as a mixer that can drive a 600-ohm load? Not likely! The problem is that the term "dB" is often used to specify a **voltage** output level. This may happen because the scales on a lot of voltmeters are calibrated in dB referenced to 0.775 volts. Thus, on these meters, +10 dB means 10 dB above 0.775 volts. Now, it just so happens that if the circuit impedance is exactly 600 ohms, and the voltage level is 0.775 volts, the power level is dBm is exactly 0 dBm! Thus, **these voltmeters read accurate dBm when they are connected to a 600-ohm circuit.**

This means that for the mixer with an output of "+18 dBm into a 600-ohm load", a voltmeter could accurately measure the power output in dBm if the load impedance was 600-ohms. However, for the mixer with an output of "+18 dBm into a 10,000-ohm load", the voltmeter would not read accurate dBm.

The question, then, is how to interpret these specifications, and what to do about the inaccurate dBm specifications. The answer may be to interpret these ratings exactly as they were measured, as voltage! The following formula allows us to find the actual value, in volts, for a rating in dBm which was measured with a voltmeter calibrated in dB referenced to 0.775 volts.

32) $? \text{ volts} = 0.775 \times 10^{(dBm/20)}$

The corresponding conversion to dBm from volts is:

33) dBm = 20 log ? volts/0.775 volts

Remember; these conversions are accurate only when the circuit impedance is 600-ohms. However, it is also useful to remember that many specification sheets use these conversions whether or not the impedance is 600-ohms. Thus, we must read the sheets carefully. Actually it is unusual to see an audio input or output level specification in dBm where the conversions in formulas 31) and 32) will **not** work.* Thus, we are assured of a good chance of coming up with useful information if we interpret these specifications as voltage, not power ratings.

Several manufacturers have begun rating the input and output levels of their low-level electronic products in dB referenced to 0.775 volts. For our mixer output, the specifications might appear as follows:

"Output Level: +18 dB (6.16 volts)",or,
"Output Level: +18 dB (re: 0.775 volts)"

This type of rating gives us the same information as the dBm rating, but it is accurate regardless of the circuit load impedance (the minimum load impedance should be noted in another specification). I believe that in most cases, this type of rating for an input or output level is actually preferable to a rating in dBm.

It should be noted that there are at least two dB-voltage ratings which are referenced to 0.775 volts. These are the so-called "dBv" and "dBe" ratings. However, these ratings are not widely accepted in the audio industry.

FIGURING dB's IN YOUR HEAD; MANIPULATING NUMBERS EXPRESSED IN dB

Sometimes it's useful to be able to convert simple ratios into dB values in your head. By memorizing a few common ratios and their dB equivalents, you can quickly estimate the dB equivalents for more complex ratios.

Ratios to Memorize for Power-Type Quantities:

The dB equivalent of a 2:1 ratio is +3 dB.
The dB equivalent of a 10:1 ratio is +10 dB.
The dB equivalent of a 1:2 ratio is −3 dB.
The dB equivalent of a 1:10 ratio is −10 dB.

---

*Noise specifications are often given in true dBm, and must be interpreted as power-related dBm ratings.

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Example #15:

What is the dB equivalent of a 4:1 power ratio?

We have already memorized that a 2:1 power ratio is equivalent to +3 dB. We remember that to multiply two numbers we simply add their logarithms (formula 17)). Since 4 is 2 x 2, we will add the dB equivalents (the logarithms) for a 2:1 ratio and a 2:1 ratio to get the dB equivalent of a 4:1 ratio.

Answer: a 4:1 power ratio is equivalent to +6 dB.

What is the equivalent of a 20:1 power ratio?

20 = 10 x 2. Therefore we will add the dB equivalents (add the logarithms) for a 10:1 ratio and a 2:1 ratio to get the dB equivalent of a 20:1 ratio.

Answer: a 20:1 ratio is equivalent to +13 dB.

Ratios to Memorize for Voltage of SPL-Type Quantities:

The dB equivalent of a 2:1 ratio is +6 dB.
The dB equivalent of a 10:1 ratio is +20 dB.
The dB equivalent of a 1:2 ratio is −6 db.
The dB equivalent of a 1:10 ratio is −20 dB.

Example #16:

What is the dB equivalent of a 4:1 voltage ratio?

Answer: a 4:1 voltage ratio is equivalent fo +12 dB. (add the dB equivalents for 2:1 and 2:1 ratios.)

What is the dB equivalent of a 20:1 voltage ratio?

20 = 10 x 2. Therefore we will add the dB equivalents (add the logarithms) for a 10:1 ratio and a 2:1 ratio.

Answer: a 20:1 ratio is equivalent to +26 dB.

SUMMING LEVELS EXPRESSED IN dB

If we are asked to sum two levels expressed in dB, we cannot simply add the two levels directly. We must either: 1) convert each of the two dB levels back to their absolute values, add the absolute values and convert back to dB or; 2) add the two dB levels using a chart such as the one in Figure 1 below. Note that this chart should be used only to add SPL levels, power levels or non-coherent voltage levels.

Summing Coherent Signals, Summing Non-Coherent Signals

The most common example of two coherent signals is two sine waves of the same frequency which are perfectly in phase (they do not have to be the same level). Similarly, any two signals which have exactly the same wave-shape and are exactly in phase are coherent. Any two signals which bear no resemblance to each other are said to be non-coherent. Thus the signals from two different microphones, each with a different talker, are non-coherent. Two pink noise signals from different generators are non-coherent.

Coherent signals (when not expressed in dB) can be added directly, that is:

Sum X and Y = X + Y

Non-coherent signals must be added as follows:

Sum X and Y = $\sqrt[2]{X^2 + Y^2}$

The signals from two loudspeakers can usually be considered to be non-coherent. Thus, they can be added as indicated in the chart of Figure 1.

Example #17:

Find the total SPL level produced by two loudspeakers, each producing 100 dB SPL.

Answer: From Figure 1, the total is 103 dB SPL.

Example #18:

Find the sum of +24 dB (12.3 volts) and +18 dB (6.16 volts) if: a) the signals are coherent; b) they are non-coherent.

a) Coherent: 12.3 volts + 6.16 volts = 18.5 volts.

18.5 volts in dB = 20 log 18.5/0.775

Answer: +27.5 dB (18.5 volts).

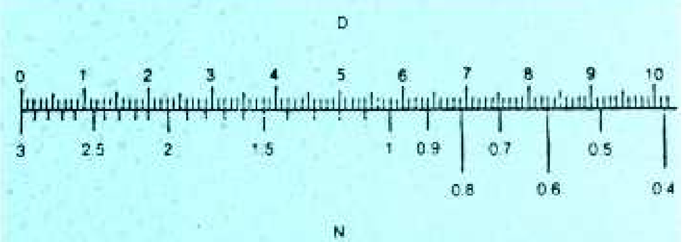

Figure 1:
SUMMING SOUND LEVEL OUTPUT OF TWO LOUDSPEAKERS, WHERE 'D' IS THEIR OUTPUT DIFFERENCE IN dB SPL. 'N' IS ADDED TO THE HIGHER LEVEL TO DERIVE TOTAL LEVEL.
(courtesy of Altec)

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b) Non coherent: $\sqrt[2]{(12.3 \text{ volts})^2 + (6.16 \text{ volts})^2} = 13.8$ volts and 13.8 volts in dB (re: 0.775 volts) = +25.0 dB (13.8 volts)

Note that Figure 1 predicts the answer for the non-coherent case. In the coherent case, the sum is exactly twice that predicted by Figure 1.

TRIGONOMETRY REVIEW

Trigonometry is the study of triangles and all their attributes. A triangle has three sides. If we draw a triangle, we see that there are also three angles. The sum of these angles is always 180 degrees. If we know two angles and the length of a side of a triangle or two sides and one angle of a triangle, we can calculate the other parts.

STANDARD TRIANGLES

a) There are a number of standard triangles. An "isosceles" triangle has two equal sides and two equal angles. An "equilateral" triangle has three equal sides and three equal angles. A "right" triangle is any triangle with one "right" angle (with one angle equal to exactly 90 degrees).

b) The right triangle is the most useful in sound system design although the isosceles and equilateral triangles can be helpful. The trigonometric functions: sine, cosine, tangent, cosecant, secant, and cotangent, plus their inverse functions: arcsine, arccosine, arctangent and so on can be defined from a right triangle as follows:

sine θ = opposite side/hypotenuse
abbreviation: sin θ

cosine θ = adjacent side/hypotenuse
abbreviation: cos θ

tangent θ = opposite side/adjacent side
abbreviation: tan θ

1/sin θ = cosecant θ or csc θ = hypotenuse/opposite side

1/cos θ = secant θ or sec θ = hypotenuse/adajcent side

1/tan θ = cotangent θ or cot θ = adjacent side/opposite side

c) The inverse trigonometric functions are defined as follows:

arcsin x = $\sin^{-1}$ x = The angle whose sin is x.

arccos x = $\cos^{-1}$ x = The angle whose cos is x.

arctan x = $\tan^{-1}$ x = The angle whose tan is x.

Note that $\sin^{-1}$ x **does not** equal 1/(sin x). Thus the notation $\sin^{-1}$ x could be confusing, and it may be better to use arcsin x instead.

PYTHAGORAS' THEOREM

Pythagoras' theorem for right triangles states that the square of the length of the hypotenuse is equal to the sum of the squares of the lengths of the two sides:

$H^2 = S_1^2 + S_2^2$

Thus, $H = \sqrt[2]{S_1^2 + S_2^2}$

FINDING TRIGONOMETRIC FUNCTIONS OF ANGLES

The easiest way to find the value of a trigonometric function is to use a book of tables or your pocket calculator! The definitions above give the trigonometric functions for angles between 0 degrees and 90 degrees. For angles between 90 degrees and 180 degrees, subtract the angle from 180 degrees. For angles between 180 degrees and 270 degrees, subtract 180 degrees from the angle. For angles between 270 degrees and 360 degrees, subtract the angle from 360 degrees. Then calculate the function on the new, smaller value and use the following chart to determine whether the function is positive or negative.

Second Quadrant (90 degrees to 180 degrees) sine is positive cosine is negative tangent is negative	First Quadrant (0 degrees to 90 degrees) sine is positive cosine is positive tangent is positive
Third Quadrant (180 degrees to 270 degrees) sine is negative cosine is negative tangent is positive	Fourth Quadrant (270 degrees to 360 degrees) sine is negative cosine is positive tangent is negative

(Note: Most scientific calculators will do all this work for you; just enter the angle and press the function.

Example #19:

sin 150° = sin (180° − 150°) = sin 30° = 0.50

cos 195° = cos (195° − 180°) = cos 15° = −0.97 since 195° is in the third quadrant

tan 310° = tan (360° − 310°) = tan 50° = −1.19 since 310° is in the fourth quadrant

Fortunately, even though we often deal with angles greater than 90 degrees, we seldom have to deal with their trigonometric functions in sound system design. In sound system design we almost always work with right triangles, and an occasional isosceles triangle, and the angles in these triangles are 90 degrees or less.

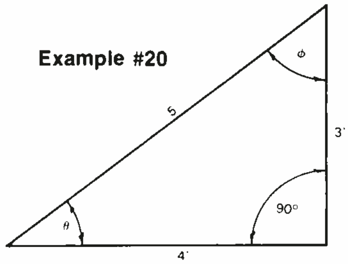
Example #20:

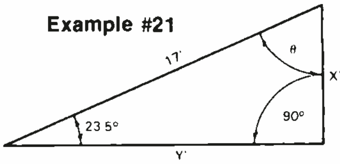
A certain right triangle has a hypotenuse of 5′, and two sides of 3′ and 4′. Draw the triangle and find the angles.

To find the angles, we use the original equations defining sin θ. (We could have used the cosine or tangent equations.)

sin θ = 3/5 therefore θ = arcsin 3/5 = 36.9°

sin ϕ = 4/5 therefore ϕ = arcsin 4/5 = 53.1°





Example #21:

A certain right triangle has a hypotenuse of 17′ and one angle = 23.5°. Find the other angle and the length of the two sides.

Since one angle of a right triangle must be 90°, and one of the other angles is given as 23.5°, the third angle can be found by simple subtraction.

θ = 180° − (90° + 23.5°) = 66.5°

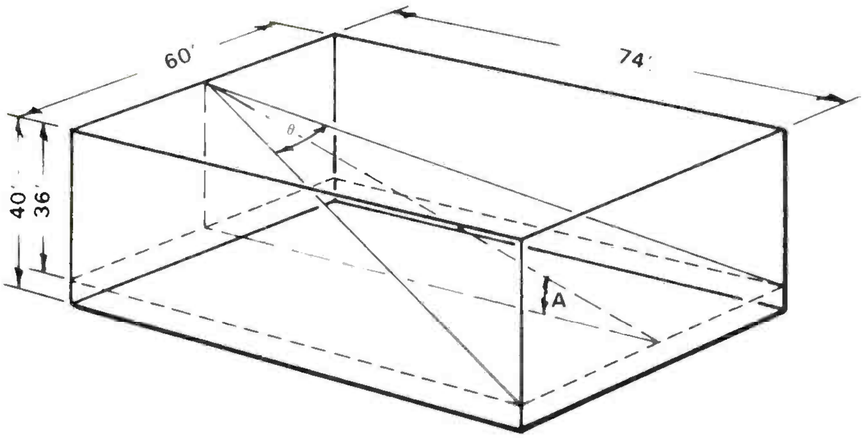
To find the lengths of the two sides, we use the equations for the definition of sin and cos:

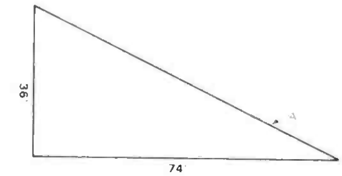
sin 23.5° = X/17; therefore X = 17 sin 23.5° X = 6.78′
cos 23.5° = Y/17; therefore Y = 17 cos 23.5° Y = 15.6′

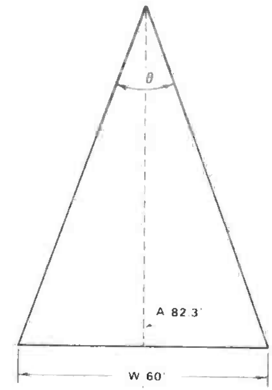
If we had used the 66.5° angle as a reference, instead of the 23.5° angle, the answers would have been the same, although the sin and cos would have been reversed as follows:

cos 66.5° = X/17 and sin 66.5° = Y/17

for additional information circle no. 55







A SOUND SYSTEM APPLICATION

A rectangular room has dimensions of 40' high by 68' wide by 74' long. What is the coverage angle at the back of the room?

First we build an imaginary platform 4' above the floor (about head height) and aim our horn at that platform.

From the drawing, we find that line A is the hypotenuse of a right triangle with sides equal to the height (now 36') and length of the room.

We can now find A from pythagoras' theorem:

$$A = \sqrt[2]{H^2 + L^2} = \sqrt[2]{36^2 + 74^2} = 82.3 \text{ feet}$$

To find θ, we draw the triangle shown. Since it is an isosceles triangle, we can divide it into two right triangles using the line A, just calculated, as the center line.

We know the length of the two sides, and the right angle in each of these triangles, therefore, we can calculate the remaining information as follows:

$$\tan \theta/2 = (W/2)/A \text{ or } \theta/2 = \arctan (W/2)/A$$
$$\theta = 2 \times 20.0° = 40.0°$$

Part 2 of this review will cover basic electrical circuit analysis techniques including Ohm's law, voltage and current division, parallel and series impedance calculations and so on. Part 2 also presents a method of modeling many of the devices we use in sound systems with simple, electrical circuits using resistors and voltage sources. From these models, we will find that we can analyze the progression of signal levels in a sound system with a minimum of calculations and a maximum of accuracy.

CONSOLE
NOISE SPECIFICATIONS

(Part 2)

by ***Paul C. Buff***
Allison Research

My article "Console Noise Specifications . . . Fact or Fiction" *(Dec. 1977 R-e/p)*, raised some strong controversy, as to the validity of certain methods of specifying equivalent input noise or E.I.N.

While the article was extremely well received, many readers have expressed a degree of bewilderment over my rather firm statement that E.I.N. figures beyond -124.8 dBme are fantasy. This bewilderment is certainly justified, in view of the almost universal notion that -130 dBme is the theoretical minimum noise level for mike pre-amps.

In one case, it was suggested to me that one of the formulas (thermal noise power = 4KTB) which I used in proving my case, was in error, since some previous authors had stated that thermal noise power = KTB.

Now, out of courtesy to those I may have left bewildered, and to those who may doubt the validity of the formulas presented, it seems necessary to take a more scientific approach in examining the origin and the validity of my statements, as well as the statements with which I take issue.

The intent of this paper is to offer undisputable proof, which may be verified, not only on paper, but on real equipment, by actual measurement.

ASSUMPTIONS

It is first assumed that the term E.I.N., when expressed in the notations dBm, or dBme, means what is implied, to wit:

1. A noise power, relative in decibels to .775 vrms, across 600 ohms (1 mw), which appears to be located at the input of the amplifier, or other device, under test or specification.

2. This equivalent input noise power is presumed to exist with the amplifier or other device, at its normal operating temperature and terminated in the same fashion, as it will be operated in, with the following exception:

   A. A resistor of equal value, to the specified input source, may be substituted for the actual transducer, during measurement, for the preclusion of acoustic noise effects.

3. The E.I.N. rating of an amplifier is assumed to be directly applicable in the computation of what signal-to-noise ratio might be expected from a transducer of known output level and impedance.

It is further assumed that the specified E.I.N. should be capable of measurement, on real devices, with real test equipment and should not be the result of processes provable only with imaginary parameters.

E.I.N. IN BRIDGING AMPLIFIERS

Since modern microphone pre-amplifiers are, by generic classification, bridging amplifiers, offering little or essentially no load to the transducer, it would be misleading to define the equivalent input noise power as that present in said load. (If this were the norm for specification, the Valley People Trans-Amp™ could legitimately be defined as having an E.I.N. of -157 dBme, due to its high (100 K) input bridging impedance.) It is then assumed that when the E.I.N. of a bridging amplifier is stated in terms of power, it refers to that power level which exists in the source/load combination, i.e., that power which exists at the input terminals. Remember, the implied meaning of E.I.N. power: Equivalent Noise Power at the amplifier input.

THERMAL NOISE VOLTAGE

The universally accepted formula for the thermally generated noise voltage across any electrical conductor, or resistor, is:

$E_T^2 = 4\ KTRB$, where:

K = Boltzmans constant, 1.38 x 10^{-23}

T = Absolute temperature in degrees kelvin

R = Resistance in ohms

B = Absolute bandwidth in Hz

It is important to note that the voltage thus expressed is a function of the square root of the resistance. This is due to the fact that the electron agitation throughout the physical resistor is non-coherent, or random, or "not in phase".

It is the non-coherency of thermal electron stimulation, which precludes the physical size, or number of series/parallel resistor combinations from affecting the outcome of the formula.

Thus, a single particle of material whose terminal resistance equals 600 ohms, produces the same thermal noise voltage and power, as a cubic foot of resistors whose series/parallel combination yields a terminal resistance equal to 600 ohms.

THERMAL NOISE POWER

There are two valid formulas for thermal noise power. Both are provable (which I shall do below), and both have their applications. It is unfortunate that at some point in time, some misinformed soul chose the wrong one (the more attractive one, of course), in the specification of his equipment. What is more unfortunate, is the fact that instead of correcting the mistake, subsequent manufacturers and "audio experts" simply followed suit, either out of a mental inability to comprehend their own specifications, or out of simple passiveness. Keep in mind that I do not include the entire industry in these allegations . . . only the guilty ones.

The applications of the two formulas are as follows:

1. PT = KTB is applicable in the illusionary process of calculating the amount of power which may be unilaterally transferred from a thermally excited resistor to an imaginary, noiseless resistor of like value.

2. PT = 4KTB is applicable in the real process of determining the power level which exists within a thermally excited resistor.

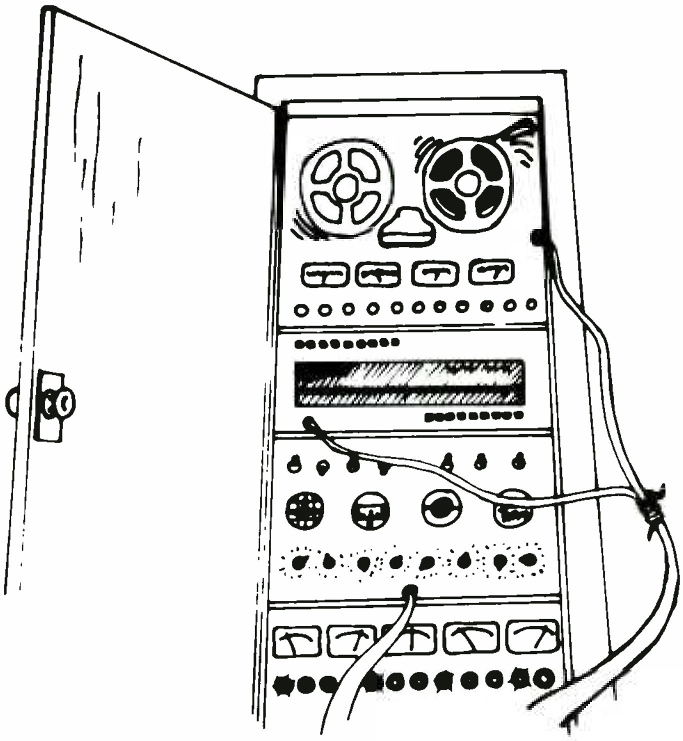
PROOF OF PT = KTB

Since the process to which the formula PT = KTB applies is one of conversion of thermal energy to electrical energy, with the subsequent distribution of said electrical energy to a previously non-energized load, the thermally activated resistor must be classified as a heat engine. As such, the mechanism is governed by the second law of thermodynamics, which states:

1. Heat energy may not be converted to work energy, unless the heat energy is made to flow from a high temperature source to a

. . . *continued*

Allison Research presents:
3 CONCEPTS FOR CONSOLE AUTOMATION

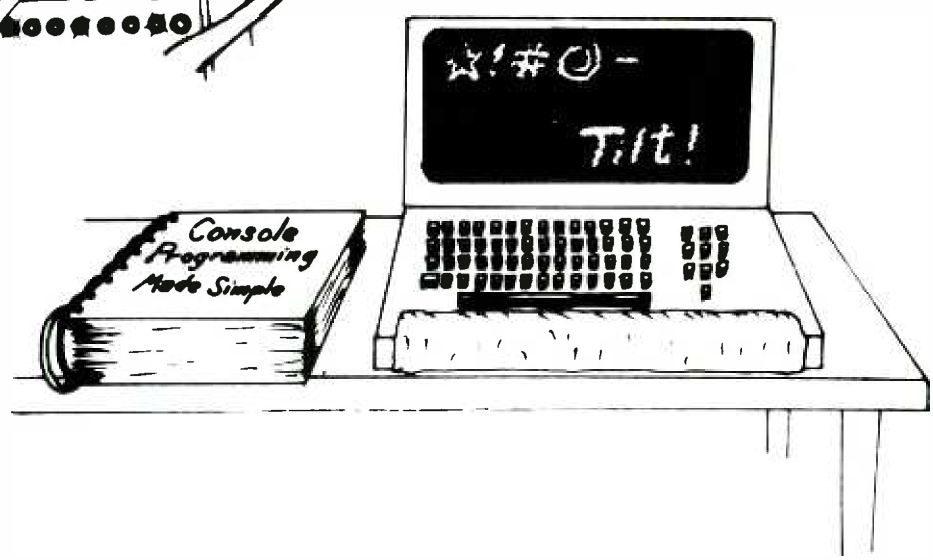


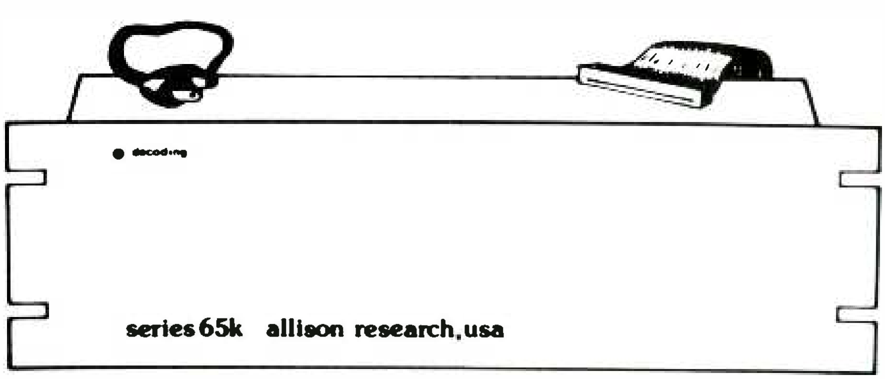
THE "CLOSET" CONCEPT

This concept involves housing your sensitive computer equipment in a specially constructed, environmental space. Your new room will need a special thermostat to keep the temperature constant year 'round. The room also serves to isolate you from annoying machine clatter and protects you from overexposure to heat generated by the equipment. COST: $70,000.00 (excluding its new home)

THE "T A L K - T O - M E" CONCEPT

This concept utilizes a combination engineer/typist. Mixing information is typed into a terminal. Your errors are displayed on the screen, and all you have to do to fix a goof is go back and read the instruction book. Meanwhile, the producer can have lunch! COST: $20,000.00





THE "UNCOMPLICATED" CONCEPT

This concept involves Allison Research's 65K Programmer. It does not involve any other peripheral equipment. You simply plug one end into your API, Harrison, Helios, Multi-Track, Sphere, or Trident console and plug the other end into the wall. You are now ready to automate! COST: $4,246.00 (for 32 channels)

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Maybe that's why about 90% of all automated consoles use allison automation!

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low temperature sink.

2. The portion of heat energy which may be converted to work energy (efficiency) is governed by the theoretical Carnot Cycle, which states that:

$$\text{A. Efficiency} = \frac{THI - TLO}{THI}$$

THI = The absolute temperature of the heat source
TLO = The absolute temperature of the heat sink

Thus governed, a thermally excited resistor (source) may not convert heat energy into electrical energy and unilaterally transfer that electrical energy to a second resistor (load), in a circuit wherein both source and load are in thermal equilibrium. A circuit which could perform that function would enable one to heat up a resistor by the simple expedient of connecting it across a second resistor. This, of course, would constitute a perpetual motion machine.

Hence, the unilateral concept of source to load transfer for thermal noise power is disqualified in equal temperature systems.

The requirement for proof of PT = KTB, which demands that the load resistor be noiseless, defines the temperature of the load resistor, by use of the formula $E_T^2 = 4$ KTB, as being zero degrees kelvin (absolute zero).

The requirement that the resistance of the load resistor be finite and equal to that of the source resistor is impossible to meet at absolute zero temperature.

Hence, the whole process is illusionary and has no practical value in real systems. Nevertheless, I will conduct a proof of the illusionary formula PT = KTB.

PROOF OF PT = KTB

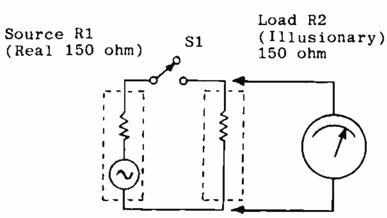

Figure 1
Model For Illusionary Power Transfer Theory

Assuming that K = Boltzmans constant (1.38×10^{-23}), T = 300° kelvin (80.6°F) and B = a 20 Hz to 20 kHz bandwidth, a mathematical solution of the formula PT = KTB will show PT (Thermal Noise Power) to be 8.27×10^{-17} watts, or -130.8 dBme. This is the amount of power delivered to the illusionary load resistor R2.

In modeling the KTB formula, (Figure 1), R1 is assumed to be a voltage generator with an internal impedance of 150 ohms. The voltage generated by R1 is assumed to be governed by the formula E_T^2 = 4KTRB.

1. With S1 open, the voltage across R1 calculates to be 2.23×10^{-7} volts rms, or -130.8 dBv.

2. With S1 closed, the voltage is divided equally across R1 and R2. The voltage appearing across R2, then, equals 1.115×10^{-7} vrms, or -136.8 dBv.

3. The power level which now exists in R2 may be calculated by ohms law ($P = E^2/R$) and will calculate to be 8.27×10^{-17} watts, or -130.8 dBme. The formula is thus proven, though it has no practical application.

Some proponents of the real world use of the PT = KTB formula will argue its validity by stating that no power can exist in a circuit unless said power is delivered to a load. This theory would offer some credibility if a resistor could be literally taken to be a voltage generator in series with a resistor. Such, as I shall proceed to prove, is not the case. As a matter of physical fact, a resistor *is a load* whose electrons may be stimulated to a given level of power by the application of heat. The fact that the stated level of power exists within the physical body of the resistor may be attested to by noting that the resistor ultimately becomes as warm as the heat source to which it is subjected. Thus, the source of power is not the resistor, or a mysterious internal voltage generator. Power is supplied to the resistor, from its surroundings, via the thermal conduction process.

Once the resistor reaches thermal equilibrium with its surroundings, the unilateral transfer of power cannot continue, lest the resistor become hotter than the applied source of heat. This concept, of course, is easily proven by the second law of thermodynamics, as well as by the use of common sense.

However, a random, bilateral transfer of energy continues to exist between the physical parts of the resistor as well as between the resistor and the heat source. It is this random, but predictable rate of energy transfer (power) which is heard as white noise in the speakers, and which may be calculated by use of the formula PT = 4KTB.

Once the reader is satisfied that the power actually exists within the physical resistor, its magnitude may be calculated by measuring the voltage across the resistor and applying ohms law ($P = E^2/R$).

PROOF OF THE EXISTENCE OF THERMAL NOISE POWER PROOF OF PT = 4KTB

It must first be considered that a resistor is not a single element, but rather is a lattice network of microscopic particles, each of which, if isolated, may be properly termed a whole resistor.

Thus, a 150 ohm resistor may be broken in half, with the physical result being two whole 75 ohm resistors, or two whole 300 ohm resistors, depending on which plane it is broken in. What we term a resistor, is actually an infinite series/parallel arrangement of microscopic resistive particles.

In order to prove the existence of power within a thermally activated resistor, as well as to prove the validity of the formula PT = 4KTB, let us make the simplified assumption that a resistor consists of two parallel components. (The same proof may be made with more complex arrangements — I have chosen a two component model for ease of explanation.)

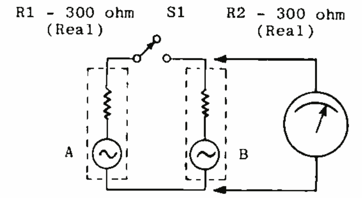

Figure 2
Model For Real Thermal Noise Power

In referring to Figure 2, one may make any of the following assumptions:

1. R1 and R2 are two separate resistors to be connected in parallel to form one 150 ohm resistor.

2. R1 and R2 are two halves of a broken 150 ohm resistor to be rejoined.

3. R1 and R2 are two of an infinite number of resistive particles within the lattice network of a physical resistor.

It is assumed that both R1 and R2, in their isolated form, are represented as voltage generators, having internal impedances of 300 ohms each.

It is further assumed that each voltage generator will produce a random voltage as dictated by the accepted formula E_T^2 = 4KTRB.

With S1 open, assuming a temperature of 300°K (80.6°F) and a 20 Hz to 20 kHz bandwidth, each generator will produce an rms voltage of 3.15×10^{-7}v, or -127.8 dBv. The two voltages will be non-coherent, with respect to one another.

When S1 is closed, a complete circuit will exist, in that each generator will cause current to flow through both resistors. This current flow will cause the voltage produced by each generator to be divided equally, with one half being dropped across each resistor.

In looking at the distribution of voltages separately, it may be stated that the voltage produced at the meter terminals, as a result of generator A (R1) will equal 1.575×10^{-7}vrms, while generator B (R2) produces an equal, but non-coherent, meter terminal voltage of 1.575×10^{-7}vrms. The accepted formula for the summation of non-coherent voltages is $E_{total}^2 = E_1^2 + E_2^2 + E_n^2$. Thus the summation of the two non-coherent voltages appearing at the meter terminals is equal to 2.228×10^{-7}vrms, or -130.8 dBv.

Since R1 and R2 are paralleled when S2 is closed, they are in fact, one physical resistor of 150 ohms resistance, which has a terminal voltage of 2.228×10^{-7}vrms. The preceeding paragraphs have shown that, indeed, current flow exists within the physical confines of the resistor. Thus, ohms law ($P = E^2/R$) may be employed to calculate a power level of 3.31×10^{-13}mw, or -124.8 dBme, within the resistor.

If the reader wishes to exercise his calculator, he (she) will find the same power level to exist with any number of resistor/voltage generators of any resistance, arranged in any lattice network which allows current flow. He (she) will also arrive at the same answer if magnitudes of current flow are calculated, or if the power levels of the individual resistor particles are summed.

Certainly, one also cannot argue that the individual particles of a physical resistor are not arranged in the series/parallel lattice network required for the internal flow of current.

Thus, it is proven that the power level *within any resistor,* at room temperature and over a 19,980 Hz bandwidth, is 3.31×10^{-13}mw, or -124.8 dBme.

Since the solution of the formula PT = 4KTB also indicates 3.31×10^{-13}mw, when applied to any resistor at the same temperature and bandwidth, its validity is also proven.

I believe, then, that I have undisputably shown that the equivalent input noise power of any electronic device cannot be less than -124.8 dBme (3.31×20^{-13}mw), when *operating* at room temperature, and measured over a 20 Hz to 20 kHz bandwidth.

In regard to the second law of thermodynamics, this power level which exists within the resistor is bilateral, or give and take, between the individual particles and the surrounding heat source and is a part of the process which causes good electrical conductors to be good heat conductors. While the long term integration of thermal noise power must add up to zero (in accord with the second law of thermodynamics), a point by point analysis will yield a random (white noise) noise power of the magnitude indicated by the formula PT = 4KTB. This noise power, when connected to the input of an amplifier, will make an output noise of exactly the same magnitude as if a signal source of -124.8 dBme were connected to the input instead. Case closed.

SUMMARY

Since a mircophone pre-amplifier is generically a bridging amplifier, or voltage responsive device, its equivalent input noise should, to be entirely correct about it, be stated in terms of noise voltage at a specified impedance (dBv), or as noise figure (the amount of noise added to that existing in the source).

If such an amplifier's E.I.N. is stated as power (dBm or dBme), that power should be the real (not illusionary) power which effectively appears to be located at its input terminals, in *actual use conditions.* This properly defined E.I.N. power level *cannot* be less than -124 dBme over a 20 Hz to 20 kHz bandwidth.

A statement of E.I.N. power less than -124.8 dBme indicates that the specified E.I.N. relates to other than *in use conditions* and must, if it is to be of any value to the user, be accompanied by a formal statement of what is actually implied by the specifications.

New Products

JBL INTRODUCES THREE NEW SOUND REINFORCEMENT LOUDSPEAKER SYSTEMS

The JBL 4662 (two-way) and 4663 (three-way) reinforcement loudspeaker systems are said to provide high acoustic output and high power handling capability while producing uniform and accurate sound over a controlled dispersion pattern.

Rugged and reliable, they are designed for either indoor or outdoor applications. Outdoors, both systems will produce 100 dB at 5 m (16.4 feet) when driven at their rated power of 125 watts continuous sine wave. A pair of systems can produce up to 6 dB more SPL than a single unit, as well as improving peak power capabilities. SPL will be even greater when the systems are used indoors.

The two-way 4662 delivers outstanding performance from 40 Hz to 9 kHz; the three-way 4663 extends the top end performance to beyond 20 kHz.

Bass is provided by the 380-m (15-inch) K 130 loudspeaker, which reproduces the rich fundamental tones of lead or rhythm guitar, electric piano, organ and vocals. Energized by a 5.4-kg (12 pound) Anico V magnet, the loudspeaker also features a 100-m (4-inch) edgewound aluminum ribbon voice coil. Close construction and precise tolerances of the assembly concentrate a magnetic field of 1.2 T (12,000 gauss).

The low frequency section of both the 4662 and 4663 have an exponential horn flare designed for high efficiency about 90 Hz and are reflex loaded for extended bass response, to 40 Hz.

The model 2461 high frequency compression driver generates high sound pressure levels, while providing clear, crisp and natural reproduction of voice or instrumentals. The magnetic assembly is heavy cast iron and the diaphragm contained in this loudspeaker is constructed of pheonolic-impregnated linen, ensuring durability and reliability for prolonged periods of use.

Model 2345, a compact radial horn unit, produces a sound distribution pattern of 90 degrees horizontal and 40 degrees vertical. Output through these angles diminishes no more than 6 dB relative to output on axis.

The additional ultra-high frequency loudspeaker built into the 4663 is engineered to deliver exceptional clarity and accuracy of overtones above 8 kHz. The unit consists of a compression driver and integral diffraction horn, which will provide efficient reproduction and wide dispersion at the extreme high end of the audio spectrum.

Both systems feature the 3110 frequency dividing network, which provides 800 Hz crossover between the bass and horn compression drivers. The 4663 is also equipped with a 3106 network for the high to ultra-high frequency transitions at 8 kHz.

The cabinet is an adaption of the basic model used by most sound touring companies. The enclosures are constructed of 7-ply 16-mm (5/8") plywood, and baffle panels are fitted with 14-20 threaded T-nuts to facilitate loudspeaker mounting.

The 4662 and 4663 enclosure dimensions are 36 inches x 30 inches x 23-7/8 inches deep. The 4662 weighs 60 kg (132 lbs.) and the 4663 weighs 64 kg (140 lbs.).

The enclosures are finished in utility black.

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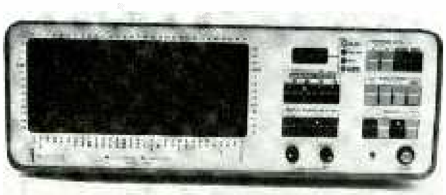
ONE-THIRD OCTAVE SOUND LEVEL AND REVERBERATION ANALYZER ANNOUNCED BY INOVONICS

In the real-time mode, the Model 500 Acoustic Analyzer displays wideband or weighted sound pressure levels in each 1/3 octave from 25 Hz to 20 kHz on a 13 by 31 LED matrix display with either peak or selectable averaging response. The reference level, which is indicated on a digital readout, can be varied manually over a 100 dB range in 1 dB steps, or the unit will seek a proper reference level automatically. The range (resolution) of the matrix display can be set by the user.

In the RT60 mode, the unit displays reverberation time up to 10 seconds with 10



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U.S. Equipment Brand Usage Survey

This special Billboard survey of recording studio equipment usage was compiled from questionnaires returned by 569 U.S. studios from June through August 1977. Though this is a representative portion, it does not necessarily reflect the exact total situation in the U.S. Studios not available for custom recording or mastering and studios not providing brand name information have been excluded. The charts show the leading brands of various types of standard equipment. The calculations for most types of equipment are a weighted figure based on both the number of studios using the equipment and on the total number of items *as reported to us*.

Consoles

Custom	14.5%
MCI	14.3%
Tascam	8.6%
API	6.3%
Auditronics	4.6%
Neve	4.3%
Electrodyne	2.8%
Langevin	2.8%
Spectra Sonics	2.6%
Opamp	2.0%
Quad-Eight	2.0%
Harrison	1.9%
Audio Designs	1.8%
RCA	1.7%
Sphere	1.6%
All others	28.2%

Tape Recorders*
(16 or more tracks)

MCI	36.3%
Ampex	23.3%
3M	23.2%
Scully	10.2%
Studer	3.1%
All others	3.9%

Tape Recorders*
(Fewer than 16 tracks)

Ampex	37.2%
Scully	15.6%
Teac	7.7%
Sony	6.0%
MCI	5.2%
3M	4.7%
Revox	4.4%
All others	19.2%

Microphone

Neumann	
Electro-Voice	
Shure	
AKG	
Sony	
Sennheiser	
RCA	
Beyer	
Altec	
Telefunken	
All others	

*NOTE: Tape Recorders calculations are based on 460 returns. Cartridge or Cassette Units may be included

Amplifiers -- Monitor

Crown	34.7%
McIntosh	19.9%
Dynaco	5.1%
Spectra Sonics	4.6%
BGW	4.4%
Phase Linear	3.7%
Altec	2.4%
Marantz	2.3%
	2.3%
	20.6%

table

	25.1%
	14.7%
	9.4%
	6.8%
	6.3%
	3.9%
	3.9%
	3.7%
	3.5%
Gately	3.0%
All others	19.7%

In consoles, MCI is the No. 1 manufacturer in the U.S.

In 16 tracks or more, MCI is the No. 1 manufacturer in the U.S.

AND NOW, MCI ANNOUNCES IT WILL BECOME THE NO. 1 MANUFACTURER OF 8-TRACK RECORDERS...

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ms resolution for either 15 or 30 dB decay. The actual decay plot is graphed on the LED matrix, and reverberation time shown on the digital display.

The Model 500 is equipped with a built-in pink noise generator that produces broadband noise for response measurement or octave-band noise for reverberation analysis. Other features include a keyboard control panel for simplified data entry, two independent memories, data and oscilloscope outputs, and a choice of microphone or line input.

The Model 500 Acoustic Analyzer lists for $2,750.00. Dealer inquiries are invited.

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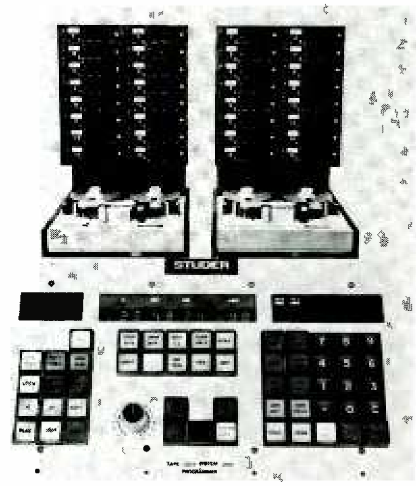
STUDER INTRODUCES NEW SYNCHRONIZING SYSTEM

The Studer Tape Lock System 2000 is a universally applicable synchronization system with a wide range of applications. It can be used both for the synchronization of professional Studer multi-channel machines of the Studer A80 type (audio-audio) and for synchronizing a Studer A80 multi-track machine through a video recorder (audio-video).



The SMPTE time code is used as the electrical link between the units being synchronized, and is recorded on an audio or cue track.

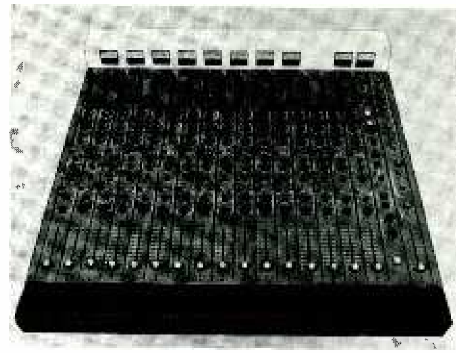
Any make of equipment can be used as master in conjunction with the Studer Tape Lock System 2000 as long as it has the ability to read the code.

STUDER REVOX AMERICA
1819 BROADWAY
NASHVILLE, TN 37203
615/329-9567

for additional information circle no. 63

SPECK SP800C

The SP800C is a 16 input 16/8 output, stereo out console which can easily be expanded to 24 track operation by the addition of the Speck 01 or 02 options. The SP800C was primarily designed to operate with MCI, 3M, Stevens, Ampex and other professional multi-track recorders, but will



also work well with semi-professional tape machines.

The input modules feature a 4½" conductive platic slide fader, 6 knob 3 band parametric equalization, 8/16 track assignment buttons, post echo send, monitor send control, 2 cue sends, solo button which allows stereo panning when engaged, a mike/line switch, program/sync switch, and a attenuation switch of –10 or –20 dB.

The output section contains everything needed to do a professional and efficient recording session, from the stereo master fader to the 8 submaster level controls. It also includes stereo control room and studio level controls, cue 1 and cue 2 level controls each of which can be soloed, slate and talk buttons with level control, 2 cue prompts, 2 cue returns, 2 two-track playback controls, 2 echo returns, and self contained microphone.

The console is completely modular, painting and silkscreening to Class A specifications, 8 submaster meters, 2 stereo meters, and for easy and convenient wiring and patching, barrier strip connections are located on the rear of the console.

Specifications: Mike input impedance 150 ohms balanced, line input impedance 10 kohms, signal-to-noise –72 dB, output level (normal) +4 dBm above 0 VU, maximum output level +20 dBm, headroom +16 dBm, equivalent input noise –127 dBm. Dimensions: Width 38", depth 37", height 7½" at rear.

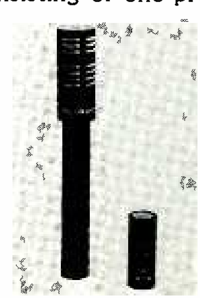
Price: $,6,500.00, F.O.B. factory.

SPECK ELECTRONICS
5642 LANKERSHIM BOULEVARD
NORTH HOLLYWOOD, CA 91601
213/769-7090

for additional information circle no. 64

BEYER DYNAMIC CONDENSER SERIES

Beyer Dynamic announes the availability of its 48V Phantom Powered Condenser Series. Consisting of one preamp/shaft



(HV710) and four interchangeable head capsules (CK711-714), the system is designed to give maximum versatility. The capsules include two omnidirectional and two cardioid patterns, with one of each pattern incorporating a windscreen. For those who don't have access to a 48V line, Beyer provides external power supplies for both balanced and unbalanced operation. Also available is a lapel clip-on condenser (MC715) and its associated 18V power supply (MSB18).

Beyer will introduce two shotgun attachments in the near future, and will add further accessories to satisfy the requirements of their customers.

BEYER DYNAMIC
155 MICHAEL DRIVE
SYOSSET, NY 11791
516/364-1900

for additional information circle no. 61

MCI 3-SPEED, ONE INCH 8 TRACK RECORDER INTRODUCED

Using the same series JH-110A transport as the time-proven MCI 24 and 16 track 2″ machines, the new Model JH-110A/8HP was developed as a scaled down version for 8 track applications where quality and track conformity cannot be compromised, but where budget is a consideration. The JH110A/8HP records 8 tracks on one inch tape at 7.5, 15, or 30 ips.

The DC servo systems for tape handling as well as the crystal controlled capstan servo systems are identical to those used in the most advanced professional systems available.

Automatic monitor switching from Cue (sync) mode to Input mode when going into RECORD makes overdubbing and editing simple. Punch-in and punch-out noise has been virtually eliminated.

Plug-in interchangeable cards make

servicing simple and convenient. Performance specifications are as rigid as any multi-track system available.

A return-to-zero (JH136) tape position locator is included in the basic unmounted unit cost of $8,200.00. The cabinet is an extra option.

MCI, INC.
4007 N.E. 6TH AVENUE
FT. LAUDERDALE, FL 33334
305/566-2853

for additional information circle no. 67

AUDIOMARKETING INTRODUCES NEW A&H LIMITER

Audiomarketing, Ltd., exclusive U.S. distributor for Allen & Heath audio equipment, has introduced A&H's new Mini Limiter. This single-channel limiter is designed for live recording. Inputs are provided for line and low-Z balanced mike.

Front-panel controls include an input level slide and five pushbuttons; one for line or mike selection while another activates limiting. Other buttons include one for attack (slow or fast) and two for release (one for slow or fast, the other for slow or medium). The control panel also has an LED indicator that is activated when 3 dB or more of limiting occurs. The mike input, at the rear, is an XLR connector. Other single jacks are standard ¼ inch phone jacks including the line input, stereo link connection, output for zero dBm and an output for -30 dBm.

The compact device, which measures only 12¼" wide x 1½" high and 4½ deep, weighs only 4 lbs.

The Mini Limiter is economically priced at $250.00.

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

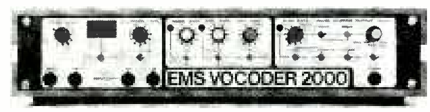
for additional information circle no. 68

VOCODER-2000 ANNOUNCED

EMS, Ltd., of Oxford, announces production of Vocoder-2000, a compact speech synthesizer whose prinicpal application is to impose articulation of any spoken sound onto another "excitation" sound derived from an audio source — organ, guitar, orchestra, oscillator bank, noise, etc. The effect is to make the excitation sound talk or sing.

The new product, according to the company, is more compact, less expensive, but not significantly less versatile than the original EMS Vocoder announced last year. New features include: voltage-controlled slew, "pause stuffing" and pedal-controlled panning — not hitherto available in these instruments, and designed to enhance live-performance capabilities of the device.

Vocoder-2000 analyzes input speech with 16 sixth-order active bandpass filters, while at the same time distinguishing voiced from unvoiced sounds. Then, in a separate 16-channel filterbank, it reconstitutes speech by treating the excitation source on the basis of the input filter information. Unvoiced sound can be restored with a noise source provided by the Vocoder, which also contains a 0 to 1 kHz pulse-wave oscillator as a simple monotone excitation source. Whatever the excitation, the result of the process is a well-formed and intelligible reconstruction of speech bearing the timbre of the excitation.



Demo cassettes are available for EMSA at $5.00.

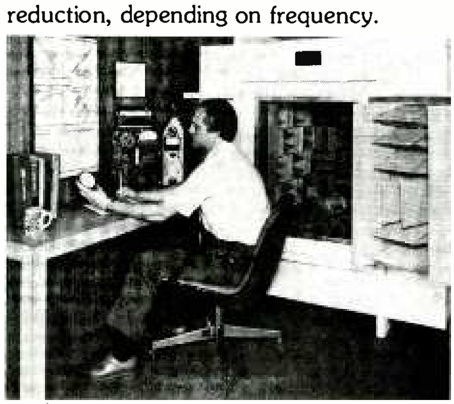
EMSA
269 LOCUST
NORTHAMPTON, MA 01060
413/586-3777

for additional information circle no. 69

PORTABLE ANECHOIC CHAMBERS PROVIDE ECONOMIC ENCLOSURES FOR DETERMINING PRODUCT PERFORMANCE

The portable An-Eck-Oic® Chambers, developed by Eckel Industries, Inc., offer a suitable environment for conducting acoustic studies on a variety of small electronic, electric, and mechanical equipment.

Standard units are available with low frequency cutoffs from 150 Hz and free field volumes up to 69" x 69" x 69". The Model 545-250-2, for instance, ideal for determining performance of small products has a low frequency cutoff of 250 Hz and a free field volume of 33" x 25" x 33". These chambers can provide from 32 to 68 dB noise reduction, depending on frequency.



The Eckel chamber is lined with pretested An-Eck-Oic® wedges. A special track mounting system is used for the factory installation of these wedges. This design, combined with a non-reflective floor system, allows a sound absorption level of between 99% and 100% to be maintained within the chamber.

The chamber can be quickly installed and put to use. The sections, which fit through an average size door opening, are shipped with the lining attached and fitting with accessories. Two 1" ips sleeves and one light fixture are standard. (Options include vibration isolators, instrumentation supports, microphone calibration rig, ventilation, etc.) Bolting the sections together is all that is required to complete the assembly on-site. After assembly, the enclosure can be safely moved to any area where it is needed, without damaging acoustic or structural integrity.

The portable An-Eck-Oic® Chambers are supplied with a frequency green urethane enamel finish or a paint grip galvanized finish, depending upon the particular model. For complete information contact:

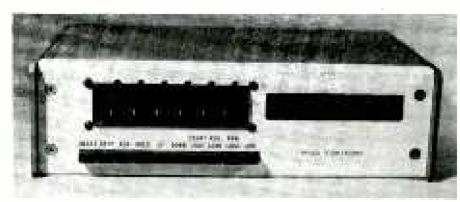
ECKEL INDUSTRIES, INC.
155 FAWCETT STREET
CAMBRIDGE, MS 02138
617/227-8581

for additional information circle no. 70

IMAGE FORMATIONS TIMER

Designed to fill a multitude of timing needs, the newly introduced unit can be used as a stop clock or an event counter.

As a stop clock it can upcount or downcount, be preset to any time with front panel thumbwheel switches, hold current display while continuing to count. The clock can be ordered to county in hours:minutes: seconds, or minutes:seconds:tenths-of-seconds; or in six decades. It can also be ordered with a crystal time base, or 60/50 mains referenced.



The controls, thumbwheel preset switches, and display can all be remoted through a connector at the rear of the clock. Multiplexed BCD display data and a logic flag from the compare register also appear on this connector. The compare register can be loaded to any value by the thumbwheel switches and will present a logic flag when the register and counter are equal.

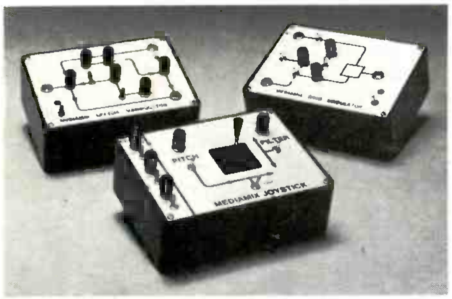
An option can be ordered which allows use of the clock as a film footage counter or clock. This option automatically determines which direction to count or stop.

IMAGE FORMATIONS
P. O. BOX 4227
BURBANK, CA 91503
213/994-2430

for additional information circle no. 71

MEDIAMIX RING MODULATOR

Mediamix has introduced a new voice and instrument manipulation device called the Mediamix Ring Modulator. The AC powered totally self-contained unit features a built-in mike pre-amp, a variable symetry



audio oscillator (used in conjunction with mike to produce talking computers, androids, etc.),an LFO (for tremelo and stereo spatial effects — applicable to a Rhodes Stage piano, for example), a Squaring function for synthesizer pitch doubling and stereo spatial effects, and an external input enabling the user to sing along with a synthesizer, producing a melodic yet electronic-sounding singing or speaking voice.

Kit price is $85.00. "Finished" units are $120.00.

This is the latest addition to a line of specialized devices, among them: a stereo spatial effects unit, a Joystick for manual pitch bend on a synthesizer, and a series of add-on modifications for Oberheim, Arp and Moog synthesizers. A 30-minute stereo demo tape illustrating the features of all of the Mediamix products is available for $2.50.

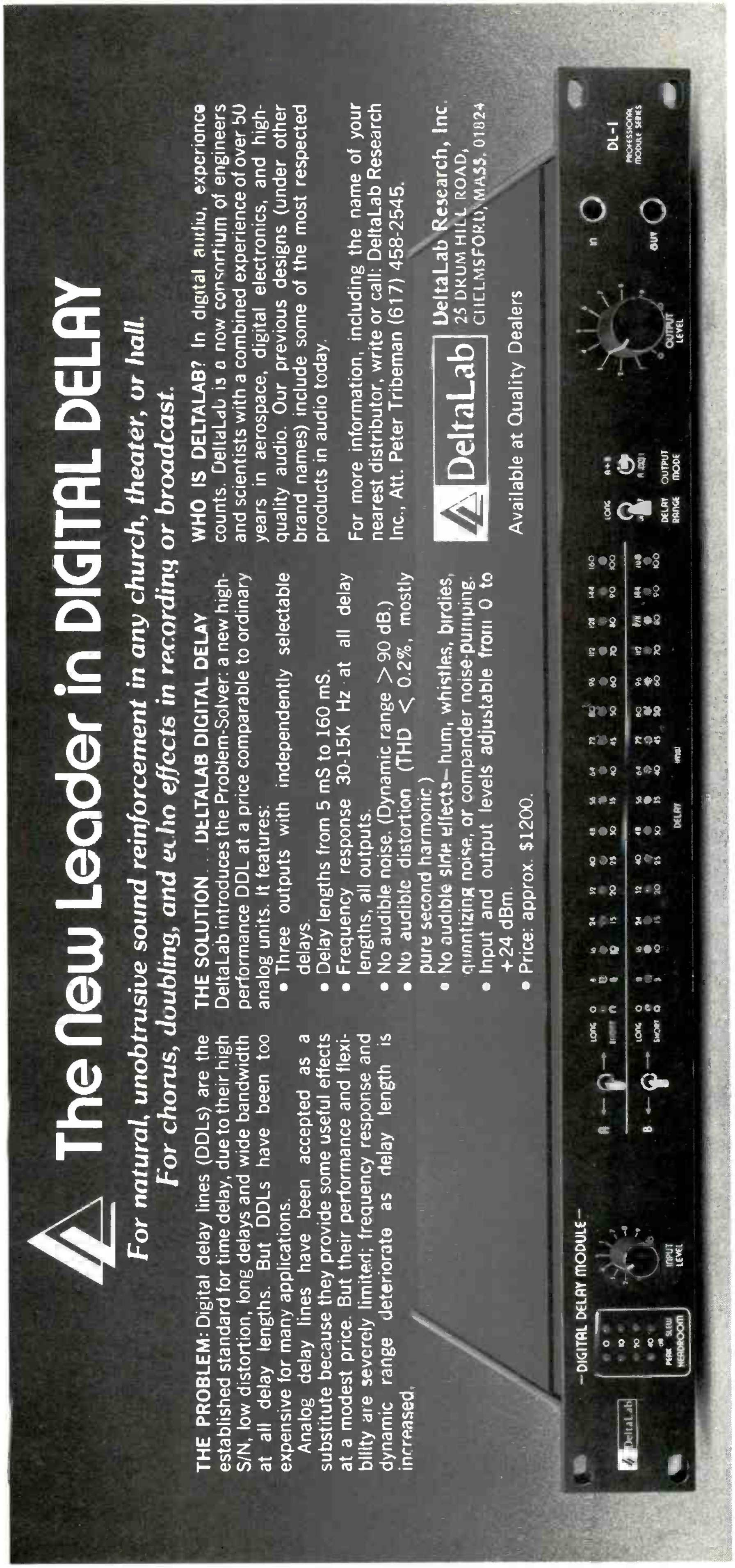
MEDIAMIX
4060 STANFORD
DALLAS, TX 75225
214/368-6846

for additional information circle no. 72

LINEAR AUDIO (L.A.) SYSTEMS STUDIO MONITORS INTRODUCED

The new line is an attractive assortment of moderately priced recording studio speakers; with each configuration designed for studio monitor mixdown, as well as the exacting home listener, where precision high level playback equipment is required.

L.A. Studio Monitors are said to be unmatched in linearity, efficiency, power handling capacity and, with elegant walnut veneer cabinetry. All components are professionaly hand-sealed and are installed in finely tuned enclosures.



for additional information circle no. 73

No inductors are used within the crossover network; the drivers natural impedance characteristics providing all the necessary roll offs, the result being the elimination of ringing, phase-shifting, insertion loss and dampening impairment, normally associated with LC networks.

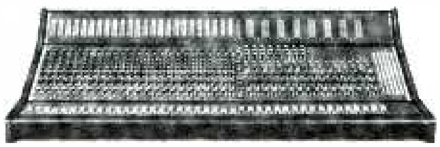
Linear Audio Systems offers three popular configurations: The three-way L.A. Studio Monitor; the two-way L.A. Midi-Monitor; and the full range "Bigmouth"™ mini-monitor cubes.

LINEAR AUDIO SYSTEMS
15210 VENTURA BOULEVARD
SHERMAN OAKS, CA 91403
213/986-9111

for additional information circle no. 74

SOUNDCRAFT SERIES THREE CONSOLES

The Series Three, which was exhibited at the 58th AES Convention in New York last November, hails the natural progression for Soundcraft into the 16 and 24 track market. A fully modular approach has been taken, and for the first time the company has used IC's throughout as a result of the availability of devices which perform as well and better than the discrete components of the Series



Two. The facilities offered include a four band equalizer, each band sweepable frequency; auto-solo on inputs, groups and auxiliary sends, which is switchable pre or post fade; LED array or conventional metering; eight auxiliary busses; patch bay utilizing Bantam (TT) jacks.

SOUNDCRAFT ELECTRONICS
5 GT. SUTTON STREET
LONDON, EC1V OBX, ENGLAND
01-251 3631

for additional information circle no. 77

AUDIOMARKETING LITTLE RED STUDIO MONITORING SYSTEM NOW AVAILABLE

In response to critical listeners who required the same Big Red sound but in a more compact size, Little Reds are now available. The very same criteria used for Big Red for frequency response, transient response, and phase correlation were used in the design of the Little Red Speaker Array and Crossover.

The Crossover incorporates the same honeycomb wound air coils and precision mylar capacitors as in the Mastering Lab Frequency Divider. A mid-frequency and high-frequency equalizer are included, allowing minor room compensation and tuning to suit individual taste.

Little Red is 24" high x 16" wide and weights in at only 45 lbs. Like its Big Red counterpart, Little Red is constructed of ¾" of low resistance composite board and is

finished in epoxy formica. It comes not only in deep red, but also in a variety of other colors. For those who prefer wood finishes, Little Reds come in walnut, too.

Little Red's price is $440.00 a pair.

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

for additional information circle no. 78

INDUSTRY STANDARD INTERFACE AVAILABLE FOR EVENTIDE 1745M DELAY LINE

A new Remote Control card uses the IEEE standard interface to permit computer automation control and introduce new effects for the Eventide 1745M Digital Delay Line.

The Remote Control card is a modular PC card which plugs into an unused connector in the Delay Line. It allows delay setting, and control of the 'repeat' and 'double' modes of operation, as well as adding two special features: automatic flanging with digital delay quality, and an extremely effective method of vocal or instrumental 'doubling'. Both features are obtained by allowing a microcomputer to vary the delay of one or more of the Delay Line's output modules. In the 'doubling' mode, the time delay is varied in a pseudo-random manner, similar to the natural effect caused by the inability of human musicians precisely to duplicate a previous performance. The note-for-note exact doubling produced by most delay equipment sounds much more mechanical than the new Eventide effect.

Heather Wood, Eventide's Marketing Manager, admits that the computer control may not find instant use in many studios. When the Delay Line was in its design stages, it was decided that automation compatibility was a 'must'. For this reason, even units which have been in the field for two years will require only minor modification. After much searching for an interface scheme, it was decided to use a standard, the ANSI/IEEE 488/1975 Interface Bus. One automation manufac-

turer already uses this system, and it will be easy for other manufacturers to make adapter boxes. The interface system is also compatible with many computers, including those made by Hewlett Packard, and a very low-cost unit, the Commodore 'PET'.

The remote unit is priced at $500.00. Another option available for the 1745M Delay Line is a Pitch Change module, which allows wide-range pitch and tempo variation, and musical harmonizing. This module cost $850.00.

EVENTIDE CLOCKWORKS, INC.
265 WEST 54TH STREET
NEW YORK, NY 10019
212/581-9290

for additional information circle no. 79

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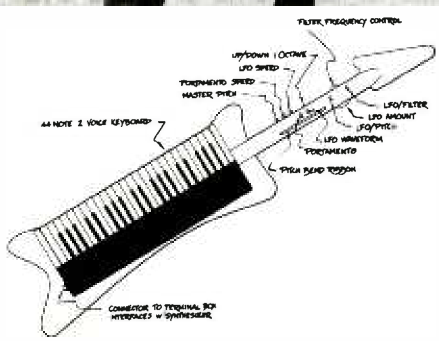
Custom built performance syntheiszer equipment is offered by the Davis Company of Hollywood. Specializing in interfacing the performing musician with modern synthesizers and accessories, the company designs and produces custom instruments. An example is the performance synthesizer keyboard shown, designed for use with any standard synthesizer equipment and providing all the most useful controls in a remote unit.



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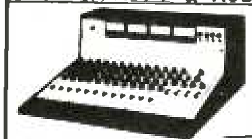

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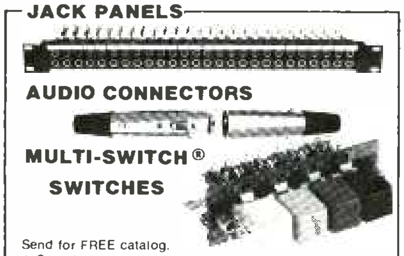

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FOR SALE: Wow and flutter meter, FL-3D with manual, $175. ***GR*** sound level meters, 1551C w/AC supply and calibrator, $195; 1565A w/1562A calibrator and case, $340. ***Altec*** "lipstick" power supply, $40. ***Ampex 354*** in stand-up console w/ new heads and and rebuilt, $1,400. ***Revox A700*** 2-track w/edit mode, portable case, and warranty (demo), $1,550. ***dbx 154,*** $450. ***dbx 157*** $400. ***Schoeps*** CMP series 12 to 48 V phantom powered transformerless mike preamps with a variety of one, two, and three pattern capsules. ***AKG 414EB's*** and 451 EB's w/CK 2 capsules matched curve pairs in stock.
Contact Wes Dooley at (213) 578-1314 or 85 Arroyo Annex, Pasadena, CA 91109.

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P.A. and custom touring sound systems, studio equipment and turn-key installations, theatre and disco sound. Representing over 100 audio lines including: AKG, Allen & Heath, Alembic, Altec, Ampex, B&W, Beyer, BGW, Cetec, Cerwin-Vega, Celestion, Community Light & Sound, dbx, Denon, Dokorder, Dynaco, Emilar, ESS-Pro, E-V, Forsythe Audio, Fons, Furman, Gallien-Kruger, Gale, Gauss, Goldring, Grace, J&H Formula 4, Kelsey, Koss, Lamb, Langevin, 3M, 3A, Marantz, Meteor, Mitsubishi, Maxell, Malatchy, MXR-Pro, Otari, Russound, Revox, SAEC, Sennheiser, Scotch, Shure, Sonab, Sound Craftsman, Soundcraft, Sound Workshop, Sony, Switchcraft, Sescom, Stax, Supex, Tapco, TDK, Tascam, Technics, TEAC, Thorens, Uher, West Penn.
ALL EQUIPMENT ON DISPLAY IN A WORKING ENVIRONMENT. COMPETITIVE PRICING & COMPREHENSIVE SERVICE.
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Electro Voice Sentry III Speakers; Sound Workshop Model 220—Vocal Doubler; AKG BX 20E Reverb—Echo Unit; Tascam Model 10 Custom designed—14-in.
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ECHO—MicMix Super "C"
New unit. We are overstocked and you take advantage. One only. *Sell at $1,700,* down from suggested price of $2,095.
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Seeks work in SAN FRANCISCO area studio/production house. Five years experience in commercials, music, A/V; strong music background. Responsible, articulate. Resume, tape:
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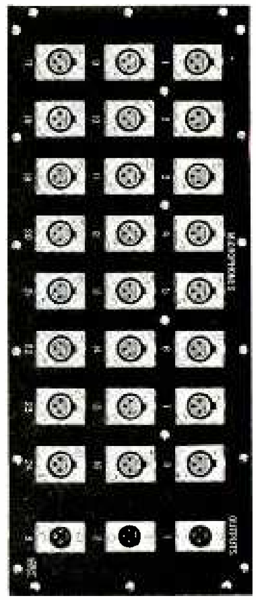

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The first Time Aligned™ Control Room Speaker System

Unbelievably Clean...from a whisper, to the threshold of pain.

The UREI 813 Monitor Loudspeaker System brings impressive new realism and clarity to recording control room listening. This first Time Aligned™ professional monitor employs the efficient Altec 6048-G duplex 15" driver with a UREI custom horn for extended and more uniform H.F. response, plus an added 15" direct radiating driver for extended L.F. response and higher power handling. Add to this the UREI 3-way TA™ network in a unique pressure controlled enclosure and you have unbelievably clean reproduction from low levels to the threshold of pain! Bring your aspirin and hear it at your UREI dealer.

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Gotham Export Corporation, New York

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video storage systems. He will be located at corporate headquarters in Redwood City, California.

Ide replace Richard Sirinsky, who has been named area manager of Europe, Africa and the Middle East (EAME) for Ampex International. He will be based in Reading, England.

He came to Ampex from TeleMation, Inc., where he was vice president-sales of the corporate marketing division. He was previously with Sarkes Tarzian, Inc., as marketing manager and, in 1959-60, was assistant and then acting national sales manager of the professional products division of Ampex.

VICE PRESIDENTS APPOINTED AT MCI

As announced by "Jeep" Harned, president of MCI, the Ft. Lauderdale based manufacturer of equipment for the professional recording and broadcast industries, Tom Hay has been elevated to the position of Vice President, Engineering, and Lutz Meyer to the position of Vice President, Marketing.

NEW TAPE MANUAL NOW AVAILABLE FROM R. K. MORRISON

A new data book for the audio tape recordist, engineer or designer is now available through R. K. Morrison Illustrative Materials.

Titled Standard Tape Manual, this is not a textbook but rather a data reference source for sophisticated users of magnetic recording equipment . . . those who are familiar with the fundamentals of magnetic recording and simply require a quick source of reference data to "plug in" to their routine endeavors. It is compiled by Robert K. Morrison, an international authority in this field and the founder of Standard Tape Laboratory, producer of test tapes.

R.K. MORRISON
ILLUSTRATIVE MATERIALS
819 COVENTRY ROAD
KENSINGTON, CA 94707

FUNDAMENTALS OF RECORDING AT BANFF TO BE CONDUCTED BY STEPHEN TEMMER

The one-week seminar, which runs from Monday May 29th to Friday, June 2nd, 1978, will again be taught by Stephen F. Temmer, President of Gotham Audio Corporation.

The extremely intensive course will consist of six hours daily of scheduled class and hands-on recording work. In addition to that, there are voluntary attendance discussion sessions every evening, probing the philosophy of recording and exchanging ideas on future technology.

In view of the artistic environment of The Banff Centre, this course, of necessity, looks at all technical aspects of recording from the viewpoint of the end product: music in the home. Musicians lend a hand as guinea pigs for experimentation in microphone and musician placement. It follows that special emphasis is placed on the fundamentals of microphone design and technique.

The seminar will satisfy most of those already involved in the recording/broadcasting/audio fields at any level, who are anxious to gain a greater appreciation and grounding in those aspects of recording which many of the texts and courses available today overlook.

Reasonably priced on-campus accommodations with private bath are provided. The view of the snow capped mountains, the outdoor hot springs pool, the heavy schedule of excellent evening musical offerings; all of these make the learning experience more vital and lasting.

GOTHAM AUDIO CORP.
741 WASHINGTON STREET
NEW YORK, NY 10014
212/741-7411

AUDIO-TECHNICA WILL DISTRIBUTE RCA DIRECT DISCS

Audio-Technica U.S., Inc., has added another direct-disc label to its roster of specialty records for sound purists.

Audio-Technica has become the U.S. distributor for the direct-to-disc recordings produced on the RCA label by RVC Corporation of Japan, according to Jon R. Kelly, A-T vice president and general manager. RVC, a joint venture of RCA and JVC, is a major factor in the Japanese record industry.

Releases announced thus far include six direct-to-disc LP's encompassing jazz and classical music. A seventh LP, "Audio Symphony", is a conventionally recorded disc, specially composed and recorded to demonstrate the potential of a modern audio component system.

The RCA direct discs are recorded at 45 r.p.m. Many experts say the faster speed results in lower distortion and a wider dynamic range.

The RCA discs are nationally advertised at $14.95 and are available through Audio-Technica dealers.

AUDIO-TECHNICA
33 SHIAWASSEE AVENUE
FAIRLAWN, OH 44313

ACI RENTALS ANNOUNCED BY AUDIO CONSULTANTS

Claude Hill, President of Audio Consultants, Inc., in Nashville, Tennessee, is pleased to announce the formation of their rental divison, ACI Rentals, Inc.

ACI Rentals will have available a wide range of professional tape recorders, noise reduction equipment, signal processing units, monitor systems and test gear. All units will be available on a daily or weekly basis.

For further information, contact **Richard Adler**, Vice President:

ACI RENTALS
1200 BEECHWOOD AVENUE
NASHVILLE, TN 37212
615/256-6999

NOW AUTO·SET

WHAT IS AUTO-SET?

AUTO-SET is not a single piece of equipment. Rather it is a system of process control micro-computers designed for the entertainment industry, and manufactured by Harrison Systems.

The first implementation of the AUTO-SET system is the 864 AUTO-SET version 1.0. This version of AUTO-SET is currently available from Harrison Systems and is for use with the Harrison 24 series, 32 series and 32B series consoles.

Additionally, the 864 AUTO-SET V1.0 can be used in any application where control signals must be stored and recalled with data management capabilities. This includes, but is not limited to audio, video, lighting and special effects.

WHAT MAKES AUTO-SET DIFFERENT?

There are four basic differences between AUTO-SET and previous automation "programmers". They are:

- **Physical Presentation**
- **Data Management**
- **Software Control**
- **Open-ended System**

Physical Presentation

AUTO-SET's obvious difference is the physical presentation of the system to the operator. The physical package appears to be a small computer terminal.

Data Management

Data management is the not so obvious difference between AUTO-SET and most previous automation systems.

Data management, in simple terms, is the ability to manipulate the data. This includes the ability to merge or separately use individual components of various data sets.

Data management in the 864 AUTO-SET V1.0 is extensive but is presented in such a way that even a novice operator can beneficially use the system with a few minutes instruction.

The data management capability includes the ability to store up to four independent mixes or dynamic sets of data on one track of an audio recorder.

Data management also includes the ability to store "Snapshot mixes" or static sets of data on a data cartridge machine included in AUTO-SET. Up to 630 individual sets of data can be stored on each cartridge.

Software Control

Internally AUTO-SET is a software or more correctly, a firmware driven machine. This means that there are many features and refinements of operation that could not economically be offered with a traditional "hard logic" design.

Open-ended System

AUTO-SET is modular. Future hardware and software modules will be available to perform many new functions.

AUTO·SET

Part of the
NO COMPROMISE
philosophy at

Harrison

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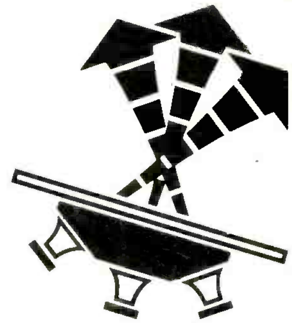
fact: the 702 stage monitor cuts through!

Made for you. The 702 was designed from scratch after a careful analysis of on-stage performance requirements — Shure tested it on stage, and Shure refined it on stage. Here's what we found:

You need . . . Clear, clean sound from your stage monitor without extra bass, and with smooth, high-end dispersion. You must hear yourself . . . above the super-amplified instruments, above the brass.

You want a monitor that cuts through! We've solved the problems, so you hear YOU — no more and no less. Where innovation was necessary, our engineers rose to the challenge. For example, the 702's unique tweeter array with three tweeters mounted in a concave, cross-firing arrangement dramatically increases high-end dispersion. This array eliminates high frequency beaming commonly found with single and double flush-mounted tweeters. This means more freedom of movement for you on stage.

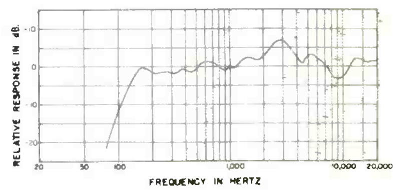
You also get . . .



Unique "wide-angle" concave tweeter array — cuts through on-stage volume, eliminates "beaming" on-axis and muddy sound off-axis.

Super Intelligibility. Shaped response — boosted mid-range, controlled bass rolloff. Lets vocals cut through on stage.
Super Output. 114 dB sound pressure level at four feet (1.2m) with only 50 watts.
Exceptionally Wide Dispersion. 90° horizontal, 110° vertical dispersion for broad stage coverage.
"Roadie" Proof. 5/8" plywood, corner protectors, rubber feet. Built to last.
Stage Versatility. Close-up (30°) or long-throw (60°) set up positions. Great for both roomy and confined stage areas.
Portability . . . Looks. All these features in a fine-looking, low profile, and lightweight cabinet.

702 Frequency Response Curve — with enhanced mid-range for outstanding intelligibility.



It's compatible with voltage — or current — source amplifiers, and is highly efficient. Handles 50 watts continuous at 16 ohms. The Model 702 Monitor is a necessary part of your act. Put it up front and you'll like what you hear.

Shure 702 Monitor Speaker

SHURE

Outperforms the best of the

Shure Brothers Inc., 222 Hartrey Avenue, Evanston, IL 60204, In Canada: A. C
Manufacturers of high fidelity components, microphones, sound systems

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