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THE SOUND ENGINEERING MAGAZINE

FEBRUARY 1980
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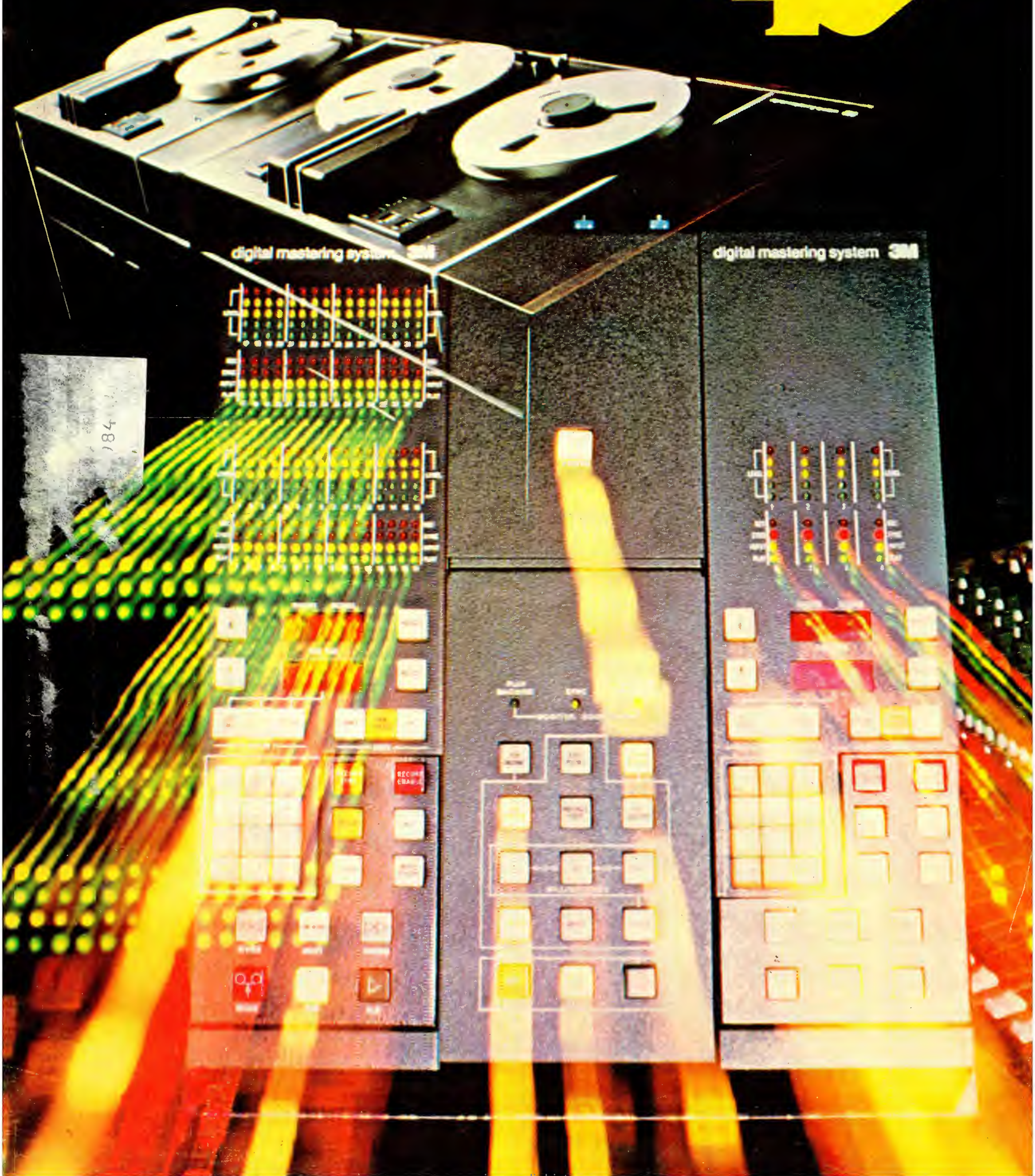

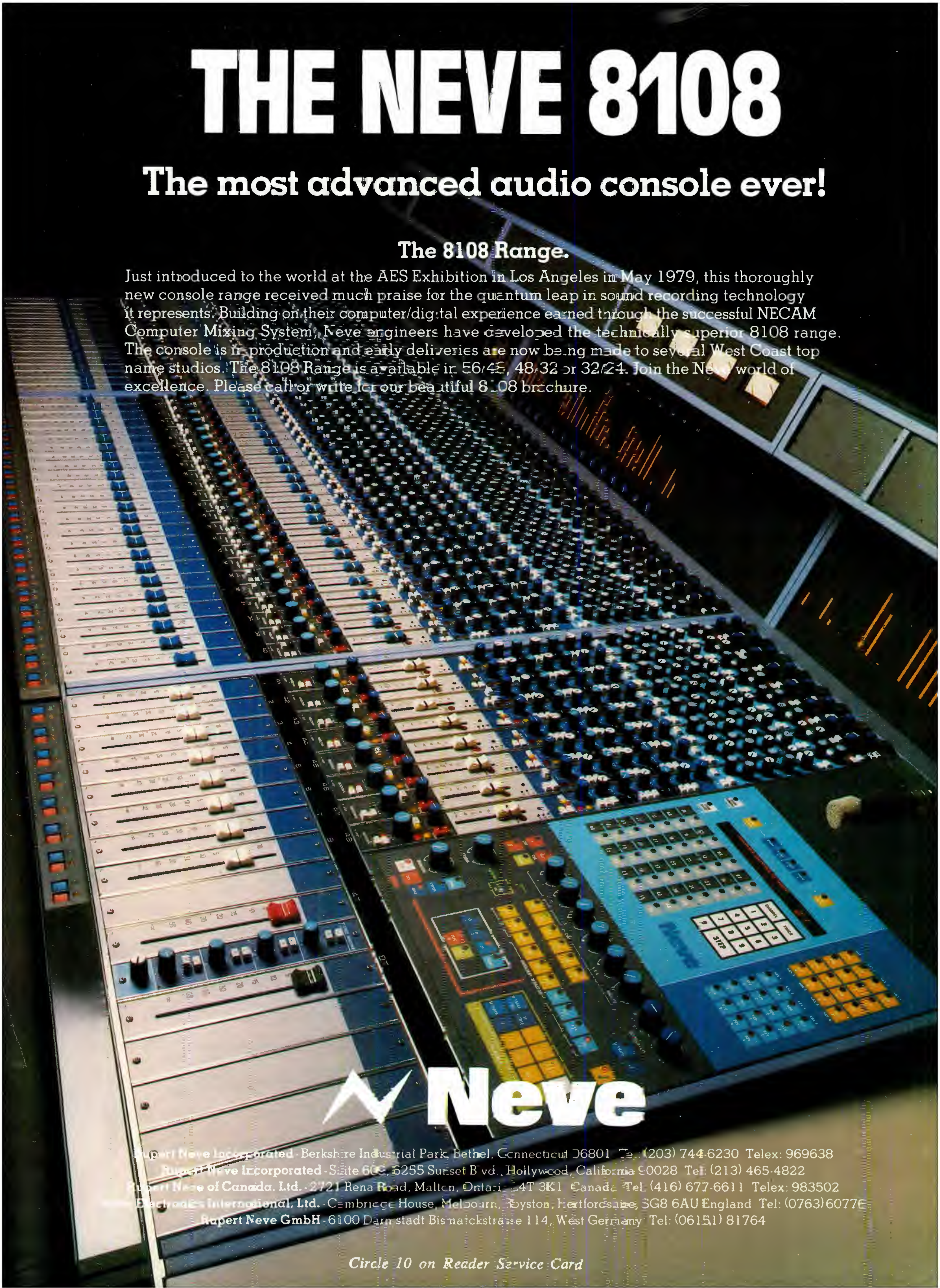

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Coming Next Month

• Next month, **db** goes to the movies, to see and hear what's been happening to soundtracks since the days of the "Jazz Singer." We'll offer a general overview on sound for the cinema, plus some details about Comtrak's four-track discrete stereo soundtrack system. And, there will be a review of speaker system requirements for the motion picture theater, as well as a look at what it takes to create the sound effects for "Star Trek (—The Motion Picture)." What else? Find out next month, in the March issue of **db—The Sound Engineering Magazine.**

About The Cover

• Digital Recording promises to have a major impact on sound reproduction in the 80s. Electronic digital editing is the latest advancement, extending the creative as well as the practical potential of this recording technology. In the foreground, the remote control boxes for a 32- and 4-track system flank the editing console that assumes control of the digital recorders during precision editing.

db

THE SOUND ENGINEERING MAGAZINE

FEBRUARY 1980 VOLUME 14, NUMBER 2

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

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Letters

To The Editor:

We have read with interest the article "Build a Heater for a Condenser Microphone" by Bob Katz in view of the fact that we have represented the NEUMANN line of microphones since 1958. It is our wish, and indeed the Neumann Company's wish, that the many Neumann microphones in the world be operated and maintained properly. I am sure that Mr. Katz was trying to be helpful and likely, in view of the illustrations which are in the article, meant his advice to owners of microphones other than Neumann's.

We have the following comment from the Neumann Company: (translation)

1. It is a question of the capsule construction and materials used, whether a condenser microphone is sensitive to temperature and humidity or not. This applies both to the capsule and the amplifier input etc. We can claim with a clear conscience that our microphones are extremely insensitive (to these factors), even if we do not make any claims in our technical brochures. Such data would provide little information anyway since there is no agreement existing about what determines these parameters in the climatic boundary areas.

If one allows a temperature range of -20° to +70° C (which we do), then the basic capacitance of the capsule will change slightly, as will the membrane stiffness (ΔC) slightly. It is important, however, that these changes are compensated by the construction which has been selected by us, and therefore does not allow the microphone's properties to deteriorate.

It is therefore unnecessary and it serves no purpose to install such a heating element on NEUMANN microphones.

2. While it is theoretically possible to clean membranes with alcohol, it is recommended that such cleaning *not* be performed by anyone but an experienced lab technician, and then only using distilled water.

A) Alcohol, because of its rapid evaporation, cools the membrane too fast. This may cause moisture to form on the back of the membrane and this would be difficult to remove.

B) Alcohol is so volatile that it will creep under the membrane clamp ring and will carry dirt into those crevices with it.

C) Alcohol evaporates too quickly. It leaves no time for careful cleaning procedures and stains always remain.

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Low-Frequency Noise	A15HP	**High Pass Filter** provides a low-frequency microphone cutoff to reduce unwanted low-frequency noises and proximity effect.
High-Frequency Noise	A15LP	**Low Pass Filter** provides high-frequency cutoff to reduce objectionable high-frequency noises.
Lack of Presence	A15PA	**Presence Adapter** adds voice-range intelligibility and extra brilliance.
Sibilance	A15RS	**Response Shaper** provides excellent sibilance filtering; flattens microphone response.
Line Level to Mic Input	A15LA	**Line Input Adapter** converts balanced low-impedance microphone input to line level input.
Matching/ Bridging/Isolating	A15BT	**Bridging Transformer**, a balanced unit, matches balanced or unbalanced devices of different impedances.
Troubleshooting	A15TG	**Tone Generator** produces a continuous 700 Hz low-impedance microphone level signal — extremely useful in setting-up and troubleshooting lines. Helps check levels, connections, mixer inputs, and cables. Allows one man to do the work of two!
Microphone Impedance Matching	A95 and A97	**Series Line Transformers** make it possible to connect low-impedance lines to mid- and high-impedance inputs (or vice-versa). Completely reversible. Solves problems of excessive high-frequency loss and objectionable hum.

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Gotham has a great deal of experience with the cleaning of NEUMANN condenser capsules. We avoid this procedure unless absolutely necessary and most certainly do not recommend it as a "cosmetic" procedure. It serves only to prolong the life of a capsule on the way to failing.

It may be interesting to note that the new model U 89 recently released by NEUMANN has a unique construction in which both membranes are at 0 volt potential, making it far less likely that dirt will lead to a breakdown of the capsule.

STEPHEN F. TEMMER
President,
Gotham Audio Corporation
New York

db Replies:
Since Mr. Temmer's letter was a timely one, we decided to include it in this month's **db**. *Next month, we will have a reply from Bob Katz, the author of the article cited in Mr. Temmer's letter.*

TO THE EDITOR:

Further to the letter in your November issue regarding standards for "XLR" type microphone connectors, I have checked with the EIA and found the following:

1. There is no EIA standard for this type of connector yet.
2. There *is* a standards proposal, number S.P. 1290, which is available from the Engineering Office of the EIA.
3. The standards proposal will probably be circulated in January 1980. If manufacturers agree on its content, it will become a standard probably in March-June, 1980—and will probably be assigned number RS-297B. It probably will include standards for pin configurations other than the 3 normally used for microphones.
4. This will be the second circulation of the proposal. The first circulation, several years ago, resulted in disagreement over certain dimensional requirements. Efforts have been made this time to assure compatibility with connectors in foreign countries also.
5. For further information regarding the status of the standard, contact the chairman of EIA committee P5.1, Mr. Robert Pontone, TRW Cinch, Elk Grove Village, IL (312) 439-8800.

Such a standard will indeed be welcome. As designers of many sound systems overseas, we have been dismayed to discover the lack of compatibility the hard way.

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db Calendar

FEBRUARY

25-28 **AES 65th Convention** (London). London Hilton and Park Lane Hotels. For more information contact: Audio Engineering Society, Inc., 60 East 42nd St., New York, NY 10017.

25-29 **The 13th International Instruments, Electronics and Automation Exhibition (IEA).** National Exhibition Centre, Birmingham, England. For more information contact: Industrial and Trade Fairs Ltd., Radcliffe House, Blenheim Court, Solihull, West Midlands B91, 2BG England, Telephone 021 705 6707.

25-29 **B&K Measurement Seminar—Industrial Noise Control I.** B&K Instruments, Inc.,5111 W. 164th St., Cleveland, Ohio 44142, Telephone (216) 267-4800.

26-28 **"Sound 80".** Cunard Hotel, Hammersmith, London.

MARCH

Syn-Aud-Con Sound Engineering Seminar

11 Day of Basics

12-14 Three-day Seminar. Dana Point Marina Inn, Dana Point, CA. For more information on the "Day of Basics" and the three-day seminar contact: Syn-Aud-Con, P.O. Box 1134, Tustin, CA 92680, (714) 838-2288.

18-21 **Measurement Seminar—Quiet Product Design.** B&K Instruments, Inc., 5111 W. 164th St., Cleveland, Ohio 44142. Telephone: (216) 267-4800.

APRIL

13-16 **National Association of Broadcasters (NAB) 58th Annual Convention and International Exposition.** Las Vegas Convention Center, Las Vegas, Nevada. For more information contact: National Association of Broadcasters, 1771 N St., N.W., Washington, D.C. 20036. (202) 293-3500

Syn-Aud-Con Sound Engineering Seminar

15 Day of Basics

16-18 Three-day Seminar. Dana Point Marina Inn, Dana Point, CA. For more information on the "Day of Basics" and the three-day seminar contact: Syn-Aud-Con, P.O. Box 1134, Tustin, CA 92680, (714) 838-2288.

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Export: Gotham Export Corporation, New York, NY

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calendar (cont.)

28-5/1 **NOISEXPO '80.** The National Noise and Vibraton Control Conference and Exhibition. Hyatt Regency O'Hare, Chicago, IL. Registration information is available from: Noisexpo, 27101 East Oviatt Road, Bay Village, OH 44140, (216) 835-0101.

15-18 **Communications '80.** Communications Equipment and Systems Exhibition. National Exhibition Centre, Brighton, England. For more information contact: British Information Services, 845 Third Avenue, New York, NY 10022, (212) 752-8400.

21-25 **B&K Measurement Seminar—Industrial Noise Control I.** B&K Instruments, Inc., 5111 W. 164th St., Cleveland, Ohio 44142. Telephone: (216) 267-4800.

28-5/1 **Audio-Visual '80 Exhibition & Conference.** Wembley Conference Centre, London, England. For more information contact: British Information Services, 845 Third Avenue, New York, NY 10022, (212) 752-8400.

MAY

6-7 **B&K Measurement Seminar—Audiometer Calibration.** B&K Instruments, Inc., 5111 W. 164th St., Cleveland, Ohio 44142. Telephone: (216) 267-4800.

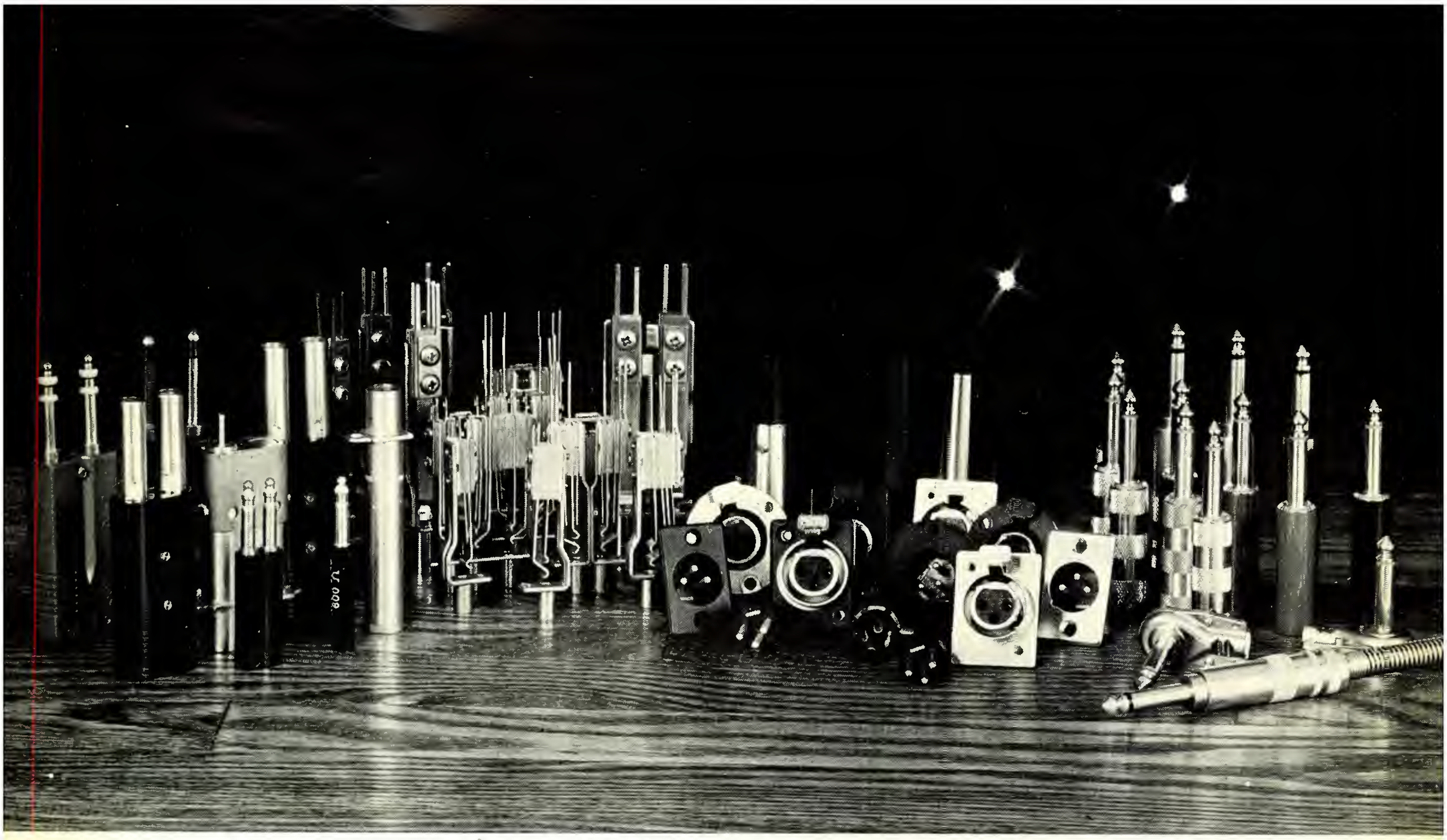
27-30 **B&K Measurement Seminar—Quiet Product design.** B&K Instruments, Inc., 5111 W. 164th St., Cleveland, Ohio 44142. Telephone: (216) 267-4800.

JUNE

15-18 **1980 International Summer Consumer Electronics Show (CES),** Chicago, IL. McCormick Place, McCormick Inn, and Pick-Congress Hotel. For more information contact: William T. Glasgow, Vice President, Consumer Electronics Shows, Two Illinois Center—Suite 1607, 233 N. Michigan Avenue, Chicago, Illinois 60601 (312) 861-1040.

19-20 **APRS '80** International Exhibition of Professional Recording Equipment. Connaught Rooms, London, England. For more information contact: British Information Services, 845 Third Avenue, New York, NY 10022, (212) 752-8400.

23-27 **B&K Measurement Seminar—Industrial Noise Control I.** B&K Instruments, Inc., 5111 W. 164th St:, Cleveland, Ohio 44142. Telephone: (216) 267-4800.



and amplifier, and have metal chasses. They have an internal ground connection, which should be the only one. But now, if those chasses touch one another, making another ground connection, you will have a loop: one connection made electrically and purposely by the designer, in the input connection; the other made accidentally where the two chasses touch. And that can cause hum.

This is one reason why metal chasses are usually anodized or painted, so that accidental electrical contact is unlikely, if not impossible. But if the insulating surface is worn off, the same thing can happen.

Another kind of loop can cause virtually the same kind of hum; but this was something our early engineers found out and now is usually eliminated in the design. That is when the ground return of an input circuit follows a widely different route from the input circuit itself. Use of concentric cable, or at least closely laid twin, preferably twisted, from the input device—such as a microphone or pickup—to the input stage of the amplifier, prevents this.

Having found that out, designers always take care to use a connection that does it. But should you develop a ground fault—a break, and use an external wire to overcome it temporarily—you may introduce the same kind of loop that the designer was careful to avoid and presto, you have a hum. The remedy is to carefully run the external connection very close to the input connection, all the way. Even a little loop, where you neglect to do this, can cause the trouble.

The best thing to do, of course, is to get a replacement connector, without the fault. But sometimes, if you know what you are doing, you can improvise, taking these precautions.

INPUT HUMS

That about covers "loop hums." Then there are a variety of input hums. These vary, according to the impedance of the input circuit. In tube days, all input stage circuits were high impedance. Low impedance input circuits required an input transformer to step the impedance up to match the input stage. Nowadays, a current-amplifying transistor, or its equivalent in an amplifying chip, provides a low impedance input stage, so such an input transformer is no longer necessary. This avoids injection of hum into the input transformer core, which was the kind we referred to first in this article.

In those days, a device that had a naturally high impedance, such as a condenser microphone, needed a high impedance input, which could be engineered to work directly into a tube grid (with appropriate circuitry, which introduced other problems we will not go into here). A low impedance device, such as a dynamic microphone or pickup, needed an input transformer.

Nowadays, the electronic part of the input circuit is designed to match the device: a current-amplifying transistor matches low impedance devices; a field-effect transistor matches high impedance devices. And the different impedance connecting circuits have the same problems they always have had, with the difference that the engineer has usually taken care of the problems in the physical design of the system.

STATIC HUM

A high impedance device is usually single-ended, or unbalanced, meaning that one connection is ground and the other "live" and at high impedance. The kind of hum to which such a circuit is susceptible is generally called "static" hum, because it is caused by electric, rather than magnetic fields. It is also characterized by a "ticky" sound. Although the ticks occur at a frequency of 60 or 120 hertz—or maybe even a higher harmonic of 60—they are noticeable as a succession of ticks, rather than a 60-hertz tone.

The remedy is complete shielding of the "live" input connection, to keep out the electric field. In such circuits, a break in the ground connection that

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is part of the shielding, or even a poor connection, can bring back such a ticky hum. So if you hear that, you know what to look for: a poor ground connection somewhere.

A low impedance input may be single-ended (also called unbalanced), or balanced. In either event, the most likely cause of hum is an input loop, because hum signals are induced magnetically. Where a very high gain is used, connection with a flat twin pair, instead of a twisted pair, can make all the difference. This is because the magnetic hum field can even get down between the two conductors of the flat twin, while by being twisted, each twist has the field going the opposite way between them, cancelling out the effect.

Shielding does not do much good for low impedance induction of hum. If the conductors inside are not twisted, the shielding may keep out any static (ticky) hum, but the magnetic hum, which has a lower, definite 60-hertz (or harmonic thereof) tone, will still get in. The input connecting cable must use twisted conductors, as well as being shielded. In fact, having it shielded may not be critical at all.

One more thing we should cover, in a discussion such as this, is that sometimes one encounters a hum that seems to be controlled by strange circumstances. There are many variations. One such situation had the hum go up and down, whenever a thermostatically controlled unit, such as a refrigerator cut in or out. In one phase of the phenomenon, the hum would go up when the thing cut out, and go down when it cut in again.

Then perhaps someone switched a light on in some other part of the building, and the action of the refrigerator was reversed: now the hum came up when the refrigerator cut in, and went down when it cut out. What could explain such a phenomenon?

GROUND POTENTIAL

This could be due to one of two causes: either a change in ground potential due to change in ground currents to other appliances or circuits; or a floating ground, which could be affected by changes in electric field in the building. A well-designed, correctly operating piece of equipment should be completely unaffected by any such external happenings, so you should look for something wrong in the equipment itself.

How does it get its ground? Is the power supply 2-wire or 3-wire? Only 3-wire is legal, most places today. But quite a bit of 2-wire supply is still in service. If it is 2-wire, maybe the equipment has no true ground, and needs an extra connection to some stable ground, such as a water-pipe. If it is 3-wire a possibility is that the ground connection has become faulty. Three wire power supplies have a live connection—a "common" connection—which should be close to ground potential, and a true ground, which should not carry any current unless there is a fault.

The supply people, and fire marshalls, are concerned with safety measures, rather than hum. The ground wire is there to conduct electric current to ground, in the event something becomes defective, without causing a shock hazard to people, and without causing fire. This means the ground connection must be husky enough to do that, so some protecting breaker or fuse will "come out," before any damage occurs.

But for audio purposes, a ground need not be that husky, so long as it conveys the ground potential to points in the system where it ought to be. At the same time, a husky ground is no disadvantage, because it makes sure that very little potential "above" ground can develop, if there are any ground currents present.

In conclusion, a personal note. You are reading this. But when I talk about it, the remnant of my British accent invariably causes someone to ask whether I am talking about "ground" or "earth?" English publications invariably use the word "earth" wherever Americans use "ground," a fact that I have forgotten (until someone reminds me) because I have been in America for over 26 years now. ■

Circle 16 on Reader Service Card

MARTIN DICKSTEIN

db Sound With Images

Training '79

• Speaking of seminars, which we did in this corner last month, there was a big one toward the end of last year at the Sheraton Center in NY. Sponsored by *Training Magazine,* the seminar ran 4 days, and included both workshops and a large exhibit. Since the prime interest of the magazine, and those that attended the sessions, was in the field of training, much of the conference was aimed toward management, software, and the methods and techniques used by trainers of personnel. However, there was a good deal there to interest audio visual specialists as well.

No matter who does the training or in what type of work the trainee is involved, there is tremendous use of audio and visual devices. The field of training is one of the largest users of video and projection—as these media lend themselves to both large and small groups. Video is used for "role playing" in which the trainees in a bank, for example, play the parts of a teller and a customer. The "transaction" is video taped and played back for the trainer and the students to critique. The same thing is done for salesmen, no matter what they sell. Executives also have themselves video taped, rehearsing to make a speech or a presentation.

Projection is used to set a mood for a meeting and also to provide information to a whole group at the same time—emphasizing specific points and details which the presenter wants the audience to remember most. So some of the sessions were of interest to the people involved with audio visuals, whether specific interests were in hardware or software.

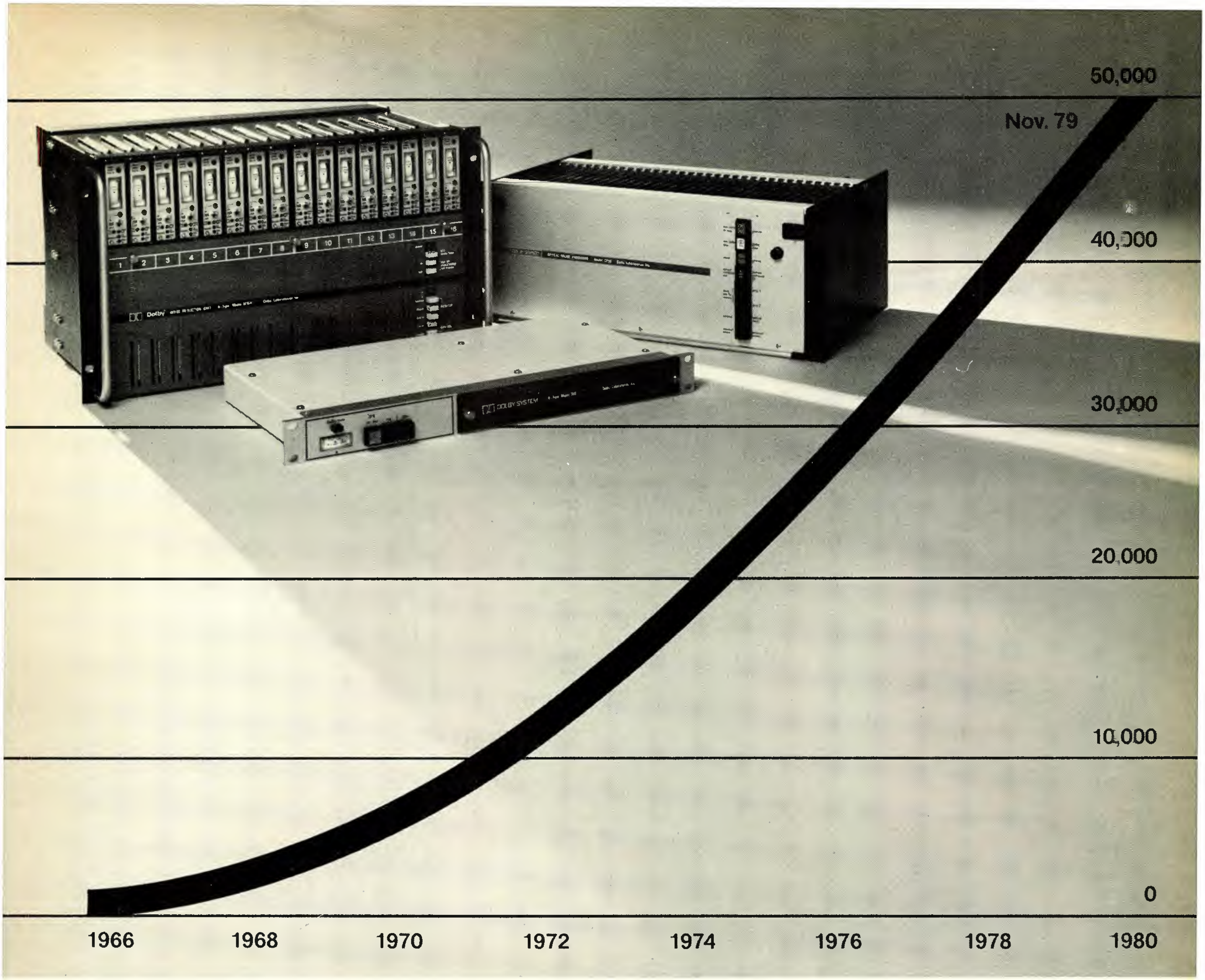
Just to give you an idea of the number of sessions there were, one or two seminars were given each morning and afternoon. On some days, there were as many as three seminars given simultaneously, but these were repeated at a different time to allow most of the attendees to take part in as many sessions as possible. Over 150 sessions were scheduled for the 4 days. Some were cancelled for one reason or another, while others were repeats from another day or time, but there were well over 100 individual, different seminars.

A SAMPLING OF SEMINARS

The sessions related to the audio/video/visual media included such titles as: Video Scripting Techniques, Developing Effective AV Training Programs, Graphic Design Basic For the Trainer, The Videodisc and How It Can Help Trainers, Putting Together a Slide/Sound Show, Basic Portable Video and Electronic Field Production, How To Edit Videotape and Transfer Other Media To Video, Editing Portable Video and Electronic Field Production Program Materials, and Video Studio Design.

TRAINING TECHNIQUES

Some of the sessions dealt with subjects connected with audio visuals



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In November 1979, the number of audio tracks throughout the world equipped with Dolby A-type noise reduction passed the 50,000 mark. No other single form of signal processing has ever been so widely accepted by professional sound engineers.

The reason is simple. Every practical method for storing and transmitting sound adds noise to the original signal. The Dolby system diminishes the noise by 10 dB without audible side effects on any kind of program material. This performance is maintained with any type and amount of noise encountered in normal professional applications. Add proven dependability and world-wide compatibility, and that is why each year more and more professionals continue to choose Dolby noise reduction.

The original Dolby noise reduction unit was the two-channel A301, nearly all of which are still in use. Today there is a range of models for every application, from the MH series for multi-track recording to the CP series for cinema sound reproduction. Together they account for the more than 50,000 equipped tracks now fulfilling the Dolby system's original promise: effective noise reduction combined with complete signal integrity.

DOLBY LABORATORIES, 731 Sansome Street, San Francisco CA 94111, Telephone (415) 392-0300, Telex 34409 • 346 Clapham Road, London SW9, Telephone 01-720 1111, Telex 919109. Dolby and the double-D symbol are trademarks of Dolby Laboratories S79/2086

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from the software and presentation end. For example, there were such topics as: Developing Your Own Platform Skills and Techniques, Writing For the Ear and Eye, How To Write Training Materials That Turn People On, and Communications In Focus: Verbal Presentation Workshop. All the other sessions dealt with training techniques, methods and materials; management and improvement of training programs; the use of computers in training; and assessment and encouragement of personnel achievement. Each of the sessions was run by an expert in the field, and many of them were designed for attendee participation.

"Writing For the Ear and Eye" was conducted by Judson Smith, media editor for *Training Magazine* and audiovisual producer. Smith discussed writing in all its forms: journalism, radio writing (for the ear), speed writing (for the ear of the audience and the eye of the speaker), tv script writing (for the eye), writing for training films and filmstrips (for the ear and eye), and for audio visual presentations (ear and eye). In his very casual manner, he talked about his past experiences in these fields, and the methods he uses to write anything. The rules were simple: write first draft quickly, save some ideas and write second draft (possibly at some time later, apart from the time of the first draft), edit and shorten, read into tape recorder, play and listen, then rewrite again, etc.

"Developing Your Own Platform Skills and Techniques" was conducted by Robert L. Montgomery, president of his own consulting company specializing in communications. Author of "A Master Guide to Public Speaking," he himself is a very commanding presenter. His dynamic vitality kept the attention of the audience riveted on him. He spoke very rapidly, moving freely without tying himself to the lectern (he refuses to use the word "podium" for the thing behind which speakers stand since the word podium, beginning with "pod," means something for the feet, like the platform on which speakers stand), and using his arms expressively to emphasize every point he made. He discussed how to overcome tension, how to speak confidently and convincingly, and how to use visual aids effectively. He involved the audience by asking for show of hands, asking individuals questions, getting all in the audience to turn and greet the people next to them, and telling them specifically when note-taking would be beneficial. With stories and humor, he kept the audience awake, aware, and interested.

Debbie Shapiro, an audiovisual specialist, ran the meeting "Graphic Design For the Trainer." She brought numerous items to help the attendees learn the subject, and demonstrated each of them. Terminology, techniques, and procedures were all discussed thoroughly to help trainers work with their own as well as outside AV material designers. Subjects covered ran from tools of the trade to size of letters, and included literature for ready reference.

SLIDE/TAPE BASICS

"Putting Together a Sound/Slide Show ended with a demonstration of a simple slide show synchronized with an audio track. The entire meeting, conducted by Bebe McClain of B. F. McClain Productions, moved through the scripting, slide shooting, sound recording, editing and synchronizing of all the elements. Different slides, including close-ups and medium shots of the same items (at different angles), were presented for the audience to see how certain slides were selected and others rejected. Participants were free to discuss their own scripts and problems after the session.

EXHIBITORS

The other sessions were also interesting and it was difficult, sometimes, to decide which sessions to attend and which to miss when a limited time was available. Then, of course, there was the exhibit. About 150 exhibitors presented their wares. These related mostly to the software of the training profession. Books, magazines, slides, consultants were all present and shown.

There were also hardware people. Bell & Howell was there. So was DuKane, as well as Optisonics, LaBelle, NEC, Oravisual, and Variable Speech Control.

All in all, there sure is a close association between training and the use of audio visuals, and the more the AV specialist knows of the techniques and methods of the training field, the more chance that specialist has to serve the trainer, both with service and equipment. ■

db New Products & Services

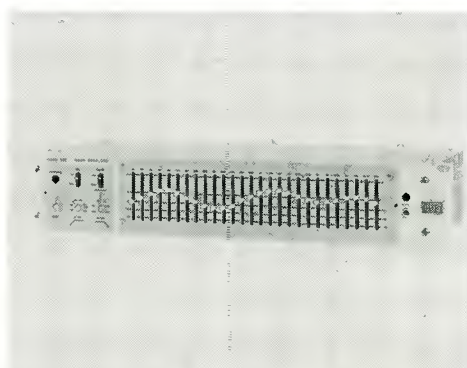
REVERBERATION UNIT

• A single transmission line system with stereo inputs and outputs, the BX-5 reverberation unit features balanced inputs and outputs, built-in reverb/dry signal mixing, adjustable input sensistivity, built-in shelving-type low-frequency equalization (±10 dB at 100 Hz) and parametric midrange equalization (±15 dB, center frequency adjustable 500 to 5,000 Hz). The BX-5 offers three selectable decay times: 1½, 2½, or 3½ seconds (approximate); and VU meter indication of the input level (single meter automatically displays the higher level of the two input signals). The unit is rack mountable, and weighs 12 lbs.
Mfr: AKG Acoustics
Price: $1,195.00
Circle 50 on Reader Service Card

MICROPHONE

• Specifically designed to be mounted on acoustic stringed instruments and other acoustic musical instruments, the SM17 miniature dynamic microphone comes with two mounting options: an expansion mount for string hole mounting on violins, violas, and cellos; and a clip mounting that fits on the sound hole of acoustic guitars, and edges of other instruments. The SM17 is supplied wired for low-impedance microphone inputs, and features an omnidirectional pickup pattern.
Mfr: Shure Brothers Inc.
Price: $76.80
Circle 51 on Reader Service Card

EQUALIZER



• An active filter set for room equalization, the model 539 provides 0 to 15 dB of attentuation at each of 27 ISO one-third-octave frequencies from 40 Hz to 16 kHz. Band-end tuning is also offered via a high-pass filter which is continuously tunable from 20 to 250 Hz, and a low-pass filter tunable from 3.5 Hz to 20 kHz. An adjustable front panel control provides up to 20 dB gain.
Mfr: UREI
Price: $824.00
Circle 52 on Reader Service Card

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The MCI 8-Track:

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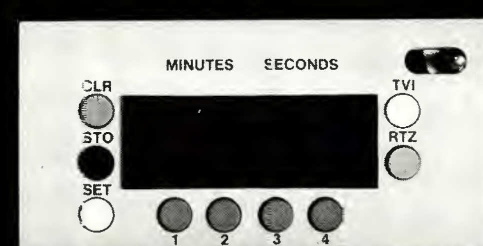
... 2 edit modes, for splicing or dumping tape
... 3-speed, separate equalizers and bias adjustments for each speed
... plug-in ready for use with MCI JH-45 AutoLock (SMPTE synchronizer)



Integral RTZ III Tape Locator provides:

... presettable up/down real time tape counter/locator
... 4 memories
... built in TVI (Tape Velocity Indicator) reading in ips or quarter semitones

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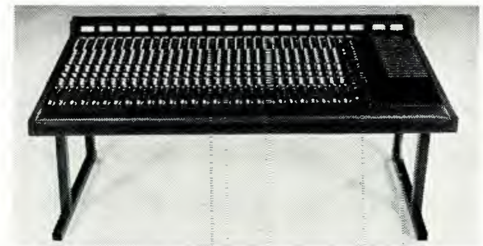
TURNTABLE

• A quartz synthesizer direct drive turntable, the model SL-1015 features an anti-resonant and acoustically deadened turntable base, and a new tonearm system design. The quartz synthesizer pitch control provides precise variation in 0.1 per cent increments above or below any of the three standard speeds (33, 45, 78 rpm), up to a maximum of +9.9 per cent. A pulsed power supply prevents hum induction, and high torque, combined with advanced circuitry, provides instant starting and stopping of 0.4 seconds. The tonearms can be interchanged for precise matching with different cartridges.
Mfr: Technics
Circle 53 on Reader Service Card

POWER AMP

• Offering totally modular construction, teflon interconnecting wires, an all-steel chassis and cover, metal-cased output transistors, and large aluminum heat-sink extrusions, the model 50A delivers 25 watts per channel (into an 8 ohm load) with a maximum thd of no more than 0.02 per cent. The hum and noise level (unweighted, 20 Hz to 20 kHz) is better than 102 dB below 25 watts, and the intermodulation distortion is less than 0.01 per cent at rated power. The unit is switch selectable for either 120 or 240 volts a.c. Dimensions for the model 50A are: 1¾ inches wide, 19 inches long, and 11½ inches deep.
Mfr: BGW Systems
Circle 54 on Reader Service Card

MIXING CONSOLE



• A 28 input, 16/28 output studio mixing console, the 800-D is totally modular. The console consists of 28 input modules (each with two discrete line input circuits), a master module, a complete communications module, and 16 illuminated VU meters all housed in a sturdy mainframe. Each input has eight panable assigns, 3 band parametric equalizers, 3 sends, pan, stereo solo, a long-throw slide fader, and a second line input with an independent slide fader, 2 band equalizer and pan. Standard on all "D" models is a 384 point patch bay wired to accept two 16-track tape recorders, or one 24-track, or 32-track recorder.
Mfr: Speck Electronics
Price: $22,900.00
Circle 55 on Reader Service Card

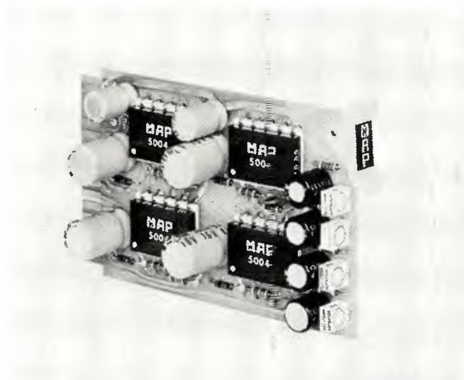
Circle 15 on Reader Service Card


AMPEX
AMPEX

P. A. SYSTEM

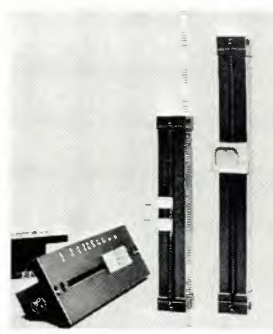
• Entitled the Dominator, this horn-loaded P. A. system provides a frequency response of 60 to 16,000 Hz. An EVM™ 15L woofer is used in a folded-horn-type enclosure for improved efficiency and smooth response in the lower octaves. The midrange horn is designed to maintain a 100-degree horizontal dispersion angle at any midrange frequency; and the high-end is handled by an ST350A tweeter. The SPL is 123 dB, at 4 ft. with 100 watts input. The Dominator is constructed of black vinyl-covered ⅝-inch plywood, with protective aluminum edging.
Mfr: Electro-Voice, Inc.
Price: $600.00
Circle 56 on Reader Service Card

CARD AMPLIFIER



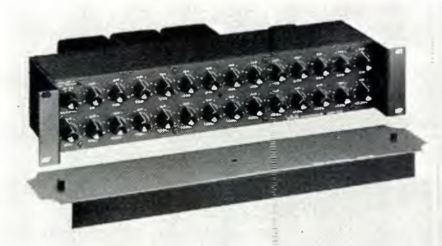
• Capable of delivering up to +20 dBm into 75 ohm loads on four independent channels, the model 4020 is the most recent addition to the IMPAC series of modular plug-in card amplifiers. Each channel of the 4020 has a self-contained gain control, and a differential type input, which can accept up to a 47K resistive source impedance. Frequency response checks in at ±0.25 dB from 20 Hz to 20 kHz, with a maximum distortion of 0.1 per cent. The model 4020 is powered by conventional ±15 V d.c. power supplies.
Mfr: Modular Audio Products
Circle 57 on Reader Service Card

SLIDE CONTROL



• Internal SPST or DPST "cue" switches have been added to the Series 400 Audio/Broadcast Slideline Controls. The switches utilize gold-plated contacts and require no additional space in package size. The 400 Series controls are ⅝-inch wide, and are available in linear and audio taper styles. Center taps are available in linear taper units.
Mfr: Duncan Electronics
Circle 58 on Reader Service Card

EQUALIZER

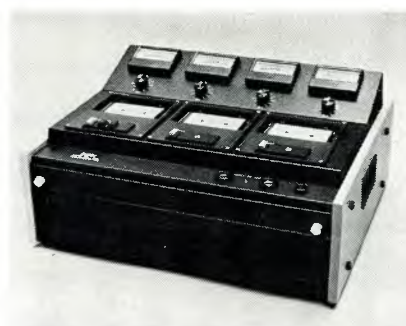


• The model 4240 is a one-sixth octave active equalizer which concentrates its double resolution in the speech intelligibility region between 200 Hz and 2 kHz. The 4240 provides 10 dB boost/cut, and is designed specifically for the equalization of sound systems employing speech as the primary program material.
Mfr: White Instruments, Inc.
Circle 59 on Reader Service Card

RIBBON MICROPHONE

• Utilizing a high-energy transducer, the M 260 ribbon microphone offers a flat frequency response from 50 Hz to 18 kHz. The ribbon element is made of pure aluminum, weighs 0.00034 grams, and is 0.85 inches in length. The hypercardioid directional pattern offers a 20 dB off-axis attentuation at 120 degrees. In addition, the M 260 provides a 12 dB bass cut filter at 50 Hz, and has an output of -60 dBm into a 200 ohm electrical impedance. Housed in either a steel or brass case, the M 260 is extremely compact, having an overall length of 7⅛ inches and a diameter of 1¾ inches at the bulb—1 inch at the shaft.
Mfr: Beyer
Burns Audiotronics, Inc. (Distributor)
Price: $189.00
Circle 60 on Reader Service Card

CASSETTE DUPLICATOR



• Offering an 8:1 duplicating speed ratio, long-life ferrite heads, flip-down panel for easy access to alignment controls, modular transport units, and duplication of all four tracks simultaneously, the DP-4050-C2 in-cassette duplicator system consists of a cassette master and two slaves. An add-on slave unit, the DP-4050-Z3, contains three cassette slave decks, and is attached to the master via a multi-pin connector cable. Up to nine additional slaves (in groups of three) can be added to the DP-4050-C2, for a total of eleven slaves all driven from one master. Each unit in the system is equipped with a servo-controlled modular transport; therefore, the failure of one unit does not disable the entire machine. The master transport, in the system, features an automatic rewind and stop.
Mfr: Otari Corporation
Price: DP-4050-C2, $2,950.00;
DP-4050-Z3, $2,750.00
Circle 61 on Reader Service Card

db Editorial

Previewing the Digital Decade

By now, surely everyone has read at least one story in which digital audio is solemnly proclaimed to be the cure-all for the various and sundry afflictions which beset our old friend analog. It seems that everything from tape hiss to lousy pressings is to become a thing of the past, once the "digital decade" of the 80s gets up-to-speed.

If half the prophecies come true, we shall be in for some impressive sonic revelations indeed. With tape and pressing noise banished, perhaps we shall, at last, be able to hear air-conditioner rumblings, and maybe even the traffic outside the studio!

In other words, the hardware isn't the only thing that will change—even studio design will be influenced by digital technology. Of course, acoustical consultants have spent years getting the noise level down, and many of them are more than ready to confront the demands of the DTR.

As the recording hardware gets more sophisticated, and studio designs more exacting, we can expect a new generation of computerized test and measurement devices will be required, to verify the performance of studios and systems. We suspect that the Badap audio micro-computer, described in our first story, may be typical of much future design work—and in areas beyond tests and measurement as well.

But, how can a system that professes so much versatility get by with so few controls? Why, there's nothing but a small collection of pushbuttons on the face plate, and no sign of a potentiometer anywhere! In fact, most of the pushbutton functions aren't even labelled. What then do they do? Well, that depends. What would you like them to do? It turns out that their function depends on the type of measurements you wish to make. The particular function at any moment is indicated on the CRT display. As your requirements change, so does the display. One button takes the place of many. There are no irrelevant controls to get in the way. If you need something, it's there—if you don't, it goes away. In other words, sophistication and simplicity at the same time.

The system also lets the user pretty much determine his own display format. For example, amplitude may be represented by a vertical line, a horizontal bar, a changing color—you decide what's needed, and it's there.

Will this sort of technology eventually find its way into other hardware? For example, what about a console fader that controls level, equalization or reverberation, depending on the needs of the moment? On a tape recorder, who needs a Play button when the transport is already moving (or a Stop button when it's not)? On a 32-track machine, wouldn't it be nice to have just one set of controls, to handle record and playback equalization? On second thought, make that a single knob that will do everything. Well, why not?

Speaking of tape recorders brings us to our next feature, on the Mitsubishi Digital Audio System. At the heart of the system is of course the digital tape recorder, but eventually the system should expand to include everything, from just after the microphone, to just before the monitor system.

Just now, attention seems to be drawn to the tape recorder itself, but it seems to us that eventually, it may be the digital everything-else that finally eclipses analog audio. Elimination of analog tape hiss, wow and flutter and such is certainly impressive. But, what about the signal-crunching capabilities that can be realized once the audio is transformed into a series of 1s and Øs? This is where digital can offer possibilities that are completely beyond analog capabilities. Once the DTR becomes a bit more accessible, we may indeed see and hear some exciting new sounds, made possible through digital signal processing.

Of course, there must be a convenient way to edit digital recordings, before we can expect to see a large-scale swing away from analog. Here, digital technology promises to become more than just a replacement for analog's razor blade and splicing tape. Our feature on electronic digital editing outlines some of the similarities, differences and future possibilities. Not the least of the advantages is that edits may be previewed and changed, again and again, without the necessity of making endless cuts and splices.

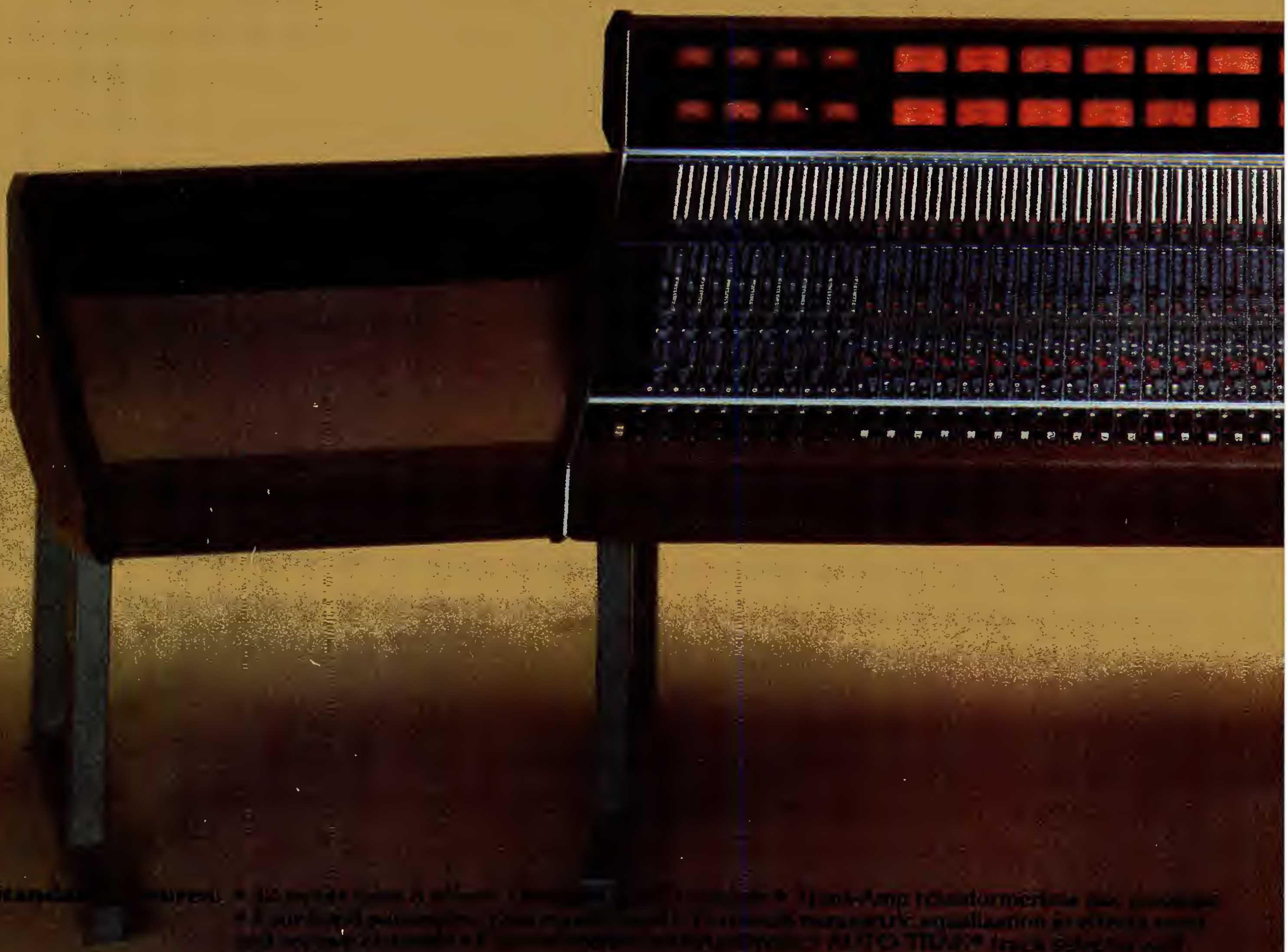
And of course, the DTR and its editing system will certainly be used in conjunction with a computerized recording console, of the type described in our next feature. It's one more example of what we can expect to see, as the digital decade gets moving.

It's tempting to speculate on what we may expect to see as this new decade draws to a close. Perhaps by 1990, the tape recorder, its editing system, and the studio console will have lost their separate identities, and be merged into a single recording "system." We have already seen tape recorders that include console functions, and consoles with tape recorder functions (See our August, 1979 and January, 1980 issues). As the technology evolves, surely there will be very much more of this, and eventually, perhaps a recorder/editor/console in an all-in-one package, since there's really little need for redundant control functions. Besides, it's expensive!

As the Badap micro-computer illustrates so well, when systems become more complex, it's also possible—and indeed, practical—for the human interface to become simpler, without sacrificing flexibility. In fact, flexibility may be greatly enhanced, since system design is no longer "frozen" when the manufacturing process begins. As any computer maven can assure us, once the basic hardware is on board, the user can keep on updating by writing new software. Therefore, the system can be continually improved, to keep up with the demands placed on it. Just try that with analog!

Of course, the complete digital recording/broadcast/home entertainment system won't happen overnight. There will surely be a lot of "cutting-and-trying" as we try to tailor the emerging digital technology to meet the demands of this digital decade. It should be an interesting period, so stick around, and we'll try to learn about it together. ■

Before you buy any automation console, test drive Auditronics 532 Memphis Machine...



Standard features: [illegible]

... it's the best you can buy for under 100 grand

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WILLIAM SOMMERWERCK

An Audio Micro-Computer For Tests and Measurements

With the appropriate programming, an audio micro computer can run frequency response, noise, distortion and flutter tests on a tape deck without human intervention. And that's only the tip of the automated-testing "iceberg."

CONSIDER THE TEST equipment you need in your studio or shop. If you're going to be well-equipped, you'll need an audio sweep generator, a.c. voltmeter, distortion analyzer, and maybe a wow/flutter meter and digital voltmeter with true rms readings. With the purchase of these items, you're all set to make measurements, and maintain and repair your equipment.

Well, not quite. You still have to find a place to mount or store all this equipment, as well as working out a convenient switching scheme to access the inputs and outputs without getting caught up in a tangle of cables. If you want to use the equipment in the field, you probably don't, because it seems like too much trouble to disassemble and reassemble the system.

Consider, too, that your test equipment is *dumb,* in every sense of the word. The instruments cannot talk to each other (exchange data) and they can't think for themselves (control their own operation and process the data they acquire). Like children, they have to be held by the hand and guided. For example, if you wish to measure the frequency, distortion, and flutter of a tape deck, you will have to set up several pieces of test equipment, attach and disconnect them from the recorder, and make and record the measurements by hand. Two tracks are bad enough, but imagine the time required for a 24-track machine!

Here's something else to consider. Traditionally, if you want to make a new type of measurement, you have to buy a new piece of test equipment. That can get expensive! If you are called on to do room reverb measurements, you have to buy (or rent, or lease) a reverberation timer, and maybe a storage oscilloscope to study the data in detail. If you can't get or afford this equipment, you might lose out on profitable contracts.

All these things point up the problems that exist with the kind of test equipment we've always used:

1. Lack of versatility: A given piece of equipment usually does only one or two things. New types of measurements require new kinds of test equipment. Multi-function test sets can be hard to master and it's easy to lose competency if not frequently used.
2. Lack of flexibility and convenience: The user must configure the equipment for a given test, and manually note the readings. There is no synergism, only an all-pervasive tangle of cables.
3. Lack of computational power: "Smart" instruments are usually quite expensive. For what most users can afford, the pocket calculator is the data-processing instrument of necessity.
4. Lack of sophisticated displays: You have meter needles and leds, but that's about it. If you want to store a display, you usually need a storage scope. And the test equipment industry acts as if the three-color CRT just doesn't exist!
5. Lack of automation: There is no way to have the equipment run a series of tests, in sequence, without attention. Think how much time you could have for more important things, if your studio equipment could "test itself."

The Badap 1™ Audio Micro Computer seen in FIGURE 1 represents a three-year design effort to resolve these and other problems. It wasn't just enough to achieve these design goals. It had to be done at a price that most studios, labs, service shops and acousticians could afford. The size and weight, too, had to be reasonable; carrying the unit should not require a forklift!

William Sommerwerck is an engineer at Barclay Analytical Ltd., Wynnewood, PA. Mr. Sommerwerck is a member of IEEE and AES.

Figure 1. Badap 1 Audio Micro Computer.

A COMPUTER-BASED AUDIO MEASUREMENT SYSTEM

The solution to these problems was achieved by designing a computer-based system specifically designed to perform audio measurements. Technically, Badap 1 isn't capable of doing anything by itself. It is the programming in it which tells it what to do, and changes it from one type of measurement to another.

Some readers are going to object that programmable test systems aren't new. True. But remember that these systems require specialized test equipment with interface buses and a small computer or calculator which often costs as much as Badap. The user has to create the programming to provide the desired testing. Badap 1 is *self-contained.* It combines the flexibility of being programmable with the economy offered by specializing in audio measurements.

The one problem a computer can't solve is "lack of sophisticated displays." We handled that by giving Badap its own color monitor. A good-quality color tv is not extremely expensive, which makes us wonder why no one has ever used one before.

Besides the color monitor and a hefty power supply, Badap 1 contains a full computer system, complete with central processing unit and memory. Program memory is on UV-erasable PROMs simplifying program changes or additions. There is also a high-speed video generator capable of creating color graphics, as well as alpha-numerics.

The remaining section contains specially-designed hybrid analog-to-digital interface circuitry, as well as the microphone and line preamplifiers. It may come as a surprise, but we use conventional analog filters and high-speed, wide dynamic range analog detectors. At the present, analog circuitry is far more cost-effective for the types of measurements we would like to do. Even more important, there is little point in going through A/D and D/A conversions, solely for the sake of digital filtering. Once a signal has been digitized, there are a myriad of other things that can be done with it, such as FFT (Fast Fourier Transform), narrow-band swept analysis, user-specifiable filters, cross-convolution of spectra, and the like. Any digital processing system should include these and other features to justify its higher cost. But since most of our users would rarely use these things, we will stay with our hybrid system. However, we are working on a digital system and when it becomes practical at a reasonable price, it will be a simple matter for those users who need it to interchange boards.

Now, what can we do with what we have? A great deal! Let's take a fairly detailed look at the Real-Time programs to see the kind of functions full computer operation makes possible. Then we'll look at some of the current and future programs to get a better grasp of its versatility.

In most real-time analyzers, a filter bank divides the input signal into octaves or, more commonly, one-third-octaves. The signals are detected, and their level shown on some kind of display, usually CRT or an led matrix. The visual effect is that of a constantly-changing bar graph.

FULL-COLOR DISPLAY

As far as that bare outline goes, the Badap 1 Real-Time Analysis System is no different from anyone else's. But Badap 1 *is* different. First off, a full-color display permits different sets of data to be distinguished by color. (Black-and-white systems can show only one display at a time, or must rely on cross-hatching on a monochrome screen for differentiation.) Blue, red, green and yellow are provided, plus four other colors for special situations. Besides the traditional "bar-graph" type display, small dots may also be displayed, again in all four colors. This gives eight possible displays, and all eight can run at the same time, showing different parameters of the signal. Or, we could show the response of a speaker from eight different positions, or the response of eight tracks of a tape deck, or—use your imagination!

Another unusual feature is an 80 dB dynamic range. The signal level can change over a 100,000,000:1 range, and still be visible, *without* range switching. Special high-speed, high-linearity detectors had to be created to make this practical.

Unlike other analyzers, the detector outputs do not feed the display directly. Instead, the outputs are sampled and stored as digital words. These words can be manipulated by the computer to provide displays that would not be practical with analog circuitry. For example, we can have an "accumulate" system where the maximum level of the signal is held indefinitely on the screen. With analog circuitry, it would be necessary to charge a capacitor to the maximum value, and read off that value with a high-impedance amplifier. But no matter how good the capacitor or amplifier is, the charge will slowly leak away,

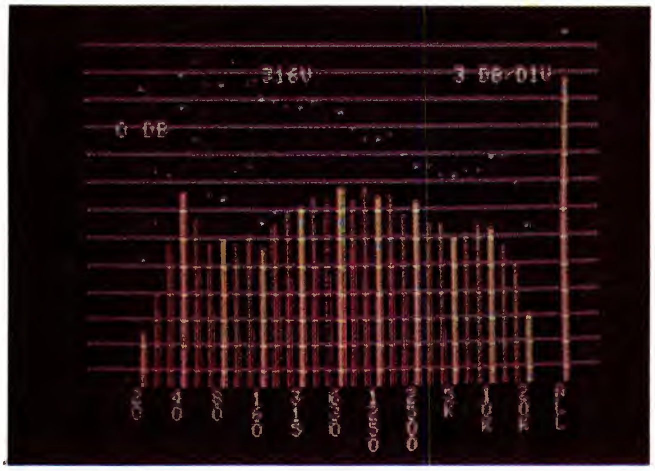


Figure 2. Full set-up. Red bars indicate average levels; yellow, peak levels; blue dots, accumulate maximum average levels; green dots, accumulate maximum peak levels.

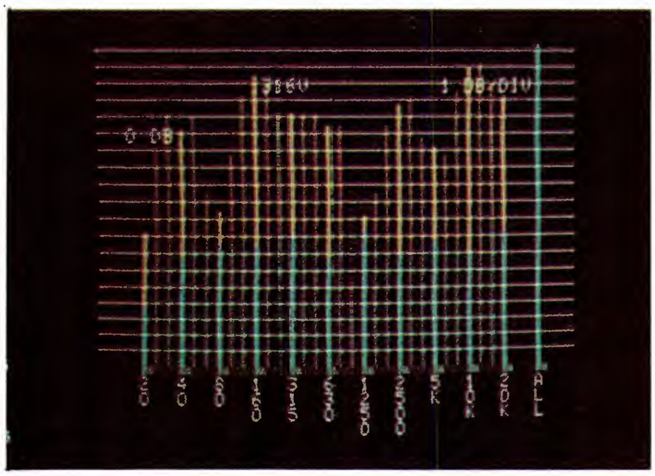
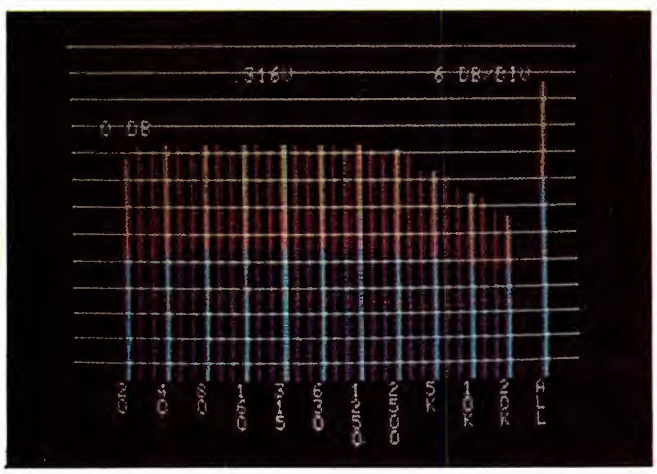


Figure 3. Response of a speaker before room equalization (yellow) and after (green).

Figure 4. Cassette frequency response at different recording levels.



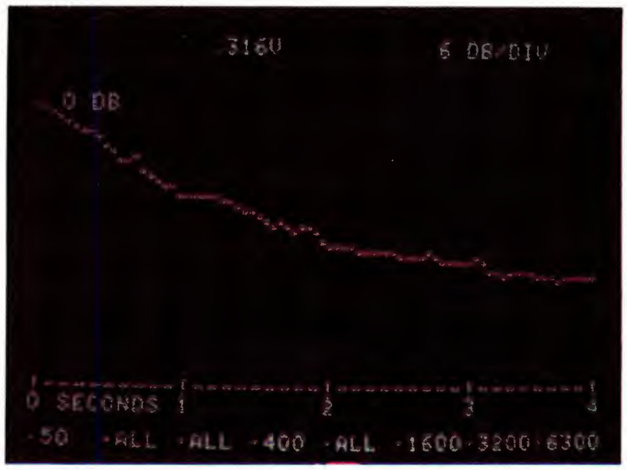


Figure 5. RT60 measurement.

changing the measured value. In Badap 1, the peak value is a word in memory, and it will remain there, unchanged, until the memory is erased or the machine unplugged.

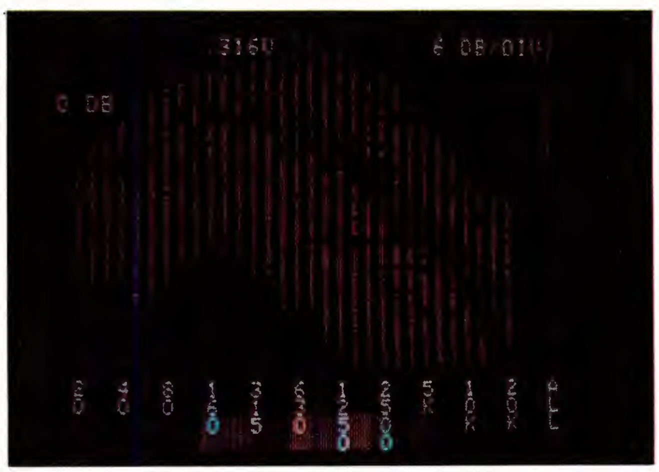
ABSOLUTE MEMORY

Another unusual feature of this system is the "absolute" nature of the memory. On some analyzers with a memory, the stored values are the relative position of the screen display. This makes it impossible to re-scale the display without modifying or destroying some data, and data which runs off the top or bottom is not recorded. Badap, on the other hand, stores the *absolute* value of the data. It can be rescaled or manipulated at any time without alteration or destruction.

As you may have deduced by now, the actual screen display is generated by special display circuitry that reads directly from the memory. The memory is divided into eight sections, one for each combination of color and shape (dot/bar). Once the user has selected a display and the type of measurement desired, the computer feeds the appropriate data to that memory, constantly updating it.

This is an extremely-flexible system. If the computer tells a particular memory to hold what it has, and stop updating, then the corresponding screen display will freeze at that point. Or, we can do the opposite—tell the memory to stop feeding the display circuits. That part of the display will disappear, but its memory will continue to be updated, until we are ready to view it again. There is also an instruction to completely clear a memory of all data.

Figure 6. Chromatic Spectral Decay.



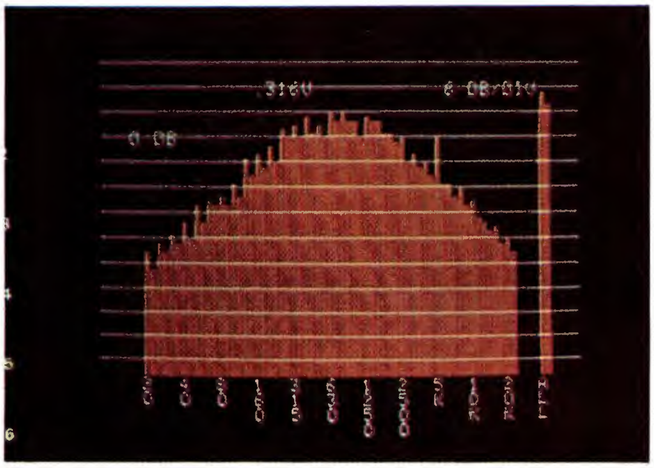


Figure 7. L + R components of a coincident microphone recording.

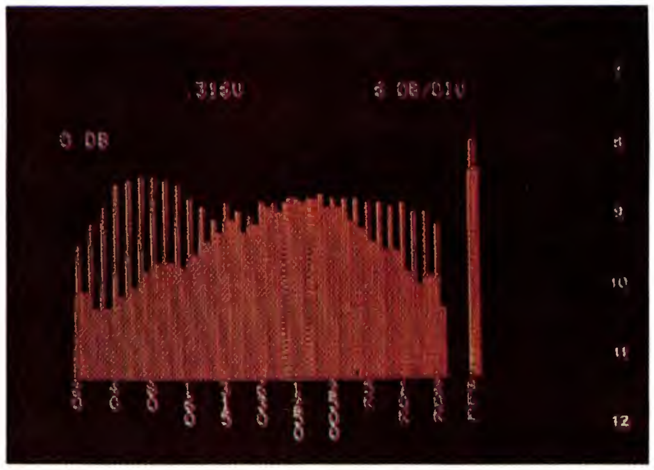


Figure 9. In-phase (orange) and anti-phase (yellow) components of a commercial recording.

MEASUREMENT DISPLAYS

Six types of measurements may be displayed. Naturally, average and peak are provided. These may also be set to accumulate, retaining the maximum levels reached. Finally, there is a special long-term average for both peak and average, called Float, which is used in setting up room equalizers. (We'll have more to say about Float later on.) This special averaging is done by the computer, manipulating the digital words that represent the sampled input. It is *not* done by switching in a bank of larger capacitors, as has to be done on analyzers using analog circuitry.

FIGURE 2 shows a typical set-up. The red bars are average, the yellow are peak. Blue dots are used to accumulate maximum average levels, and the same is done with green dots for peak levels. The user chooses the colors; almost any other combination could be used.

In FIGURE 3, Float mode is used, with average readings to show the response of a speaker before equalization (yellow) and after equalization (green).

DIGITALLY-CONTROLLED TEST SOURCE

You're probably wondering what we use as a test signal for equalization. We *don't* use pink noise; it has certain limitations that we'll discuss later. Instead, Badap 1 has its own digitally-controlled source, which can generate a wide variety of test signals. The most commonly-used output is the DGFS (Digitally-Generated Flat Spectral) signal. It is not noise, but a controlled, deterministic sound which has some important advantages over noise. Unlike pink noise, the DGFS has a crest factor (peak-to-average ratio) which is constant over the audio band. Pink noise has a higher crest factor at low frequencies than at high. This requires an analyzer using pink noise to have a much longer detector time constant at low frequencies, if the display is to be stable. (This isn't just theoretical. If you've ever equalized with a conventional analyzer, you *know* just how slow the lowest octaves are when the integration time is long enough for a steady display.) The constant crest factor of the DGFS eliminates this problem; all one-third octaves respond at the same rate. Since integration is performed by the computer, not by a large capacitor hanging across the detector, it is possible to rationalize the conflicting demands of rapid response and a stable display. Badap doesn't respond instantly, but it is rather quicker than what you are used to.

The other advantage of the DGFS is that its crest factor is several times that of pink noise, more-closely approximating that of music. It stresses the unit under test more the way music does. FIGURE 4 shows the frequency response of a tape recorder with peak-reading meters, using the DGFS. The meter is set to read 0. For this peak level, the *average* level of the DGFS is lower than the average level of what pink noise would be, so the response of the machine is appropriately wider. This more-clearly indicates the subjective response of the deck when recording music, as opposed to test tones.

REVERBERATION MEASUREMENT

Now that we've gotten a good overview of the operation of

Figure 8. Sum (red) and difference (blue) components of the same coincident microphone recording in figure 7.

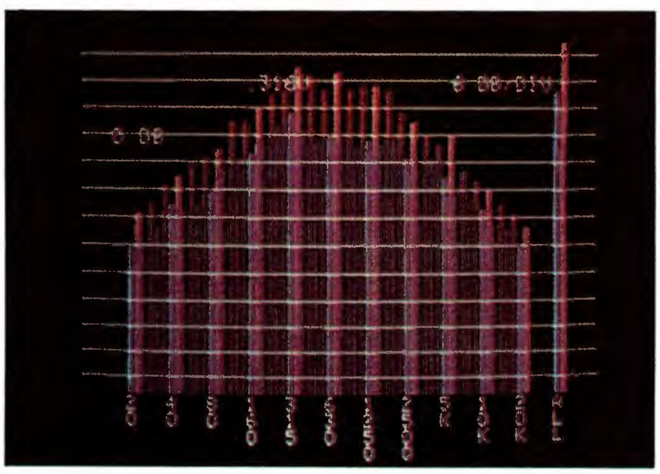
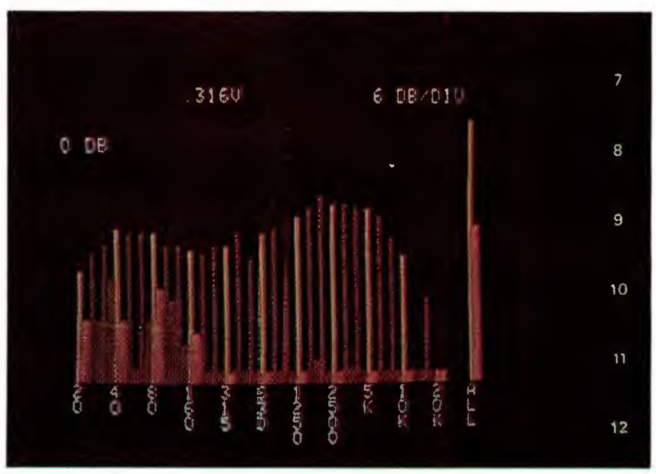


Figure 10. Sum (orange) and difference (yellow) spectrum of record clicks and pops.



the real-time analysis program, let's return to that rhetorical question of: What would happen if you suddenly found it necessary to do room reverberation measurements? The traditional stopwatch and starting pistol approach just doesn't give enough information. The acoustician would like to look at sections of the audible spectrum, in at least one-octave-wide bands, and preferably have the ability to analyze any one-third-octave in detail. This kind of measurement has traditionally required one-third-octave filters, a noise source, and a storage scope. (You do all the computations yourself. Don't forget to buy one of those little clear plastic rulers to lay across the screen when you're interpolating the slope.) This complexity has encouraged some companies to develop reverberation timers which provide bandpass filtering and compute RT60. But you still need a storage oscilloscope if you want to analyze the decay in detail. And are you really willing to trust a number when you can't see the data from which it's derived?

Badap 1 can run RT60 measurements, without using any external equipment, simply by installing a set of PROMs which includes the RT60 program. (There is substantial programming space in Badap, and giving up an existing program to get a new one is rarely required. Memory expansion modules are available.)

To run an RT60 measurement, the Source output of the Badap is connected to a speaker/amplifier in the room to be tested. The desired one-third-octave is selected, the level to the speaker adjusted, and a RUN button touched. The test signal will shut off automatically, and a graph of the decay will be displayed. (There are eight memories for this program, so up to eight runs may be stored at one timc.) The user can now inspect the decay in detail, to look for irregularities, or to make his own RT60 determination. Or, the computer can calculate RT60, if requested. If a determination is possible, the machine will indicate whether or not it was made to ISO standards. If no determination can be made, or the data looks bad, the display will flash "INVALID." (FIGURE 5)

It might be worth pointing out that the basic Badap costs more than what one would pay for a good real-time analyzer (although Badap is considerably more powerful and versatile). But Badap 1 is not *just* a real-time analyzer, and when RT60 is added, the cost is below what one would pay for separate instruments to do things. The separate instruments would be harder to store and carry, and not perform as well. The bottom line is that RTA and RT60 are just the beginning of what Badap 1 can do.

ACOUSTIC DISTANCE MEASUREMENT

Take, for example, a recently-introduced program: Acoustic Distance Measurement. The Source outputs a pulse to the hall's speaker system, and a timer determines how long it takes for the pulse to reach Badap 1's microphone. This is converted to distance, and displayed. The applications in determining critical distance (D_c) and in setting the delay lines of multi-speaker sound reinforcement systems should be obvious.

CHROMATIC SPECTRAL DECAY

FIGURE 6 shows the screen display of a program which is beyond the capabilities of all but the most elaborate and expensive audio analyzers. (Since no one else has color, we could imply that we're *really* unique, but there are ways other than color to display this data.) The program is called "Chromatic Spectral Decay," and it enables the user to view the reverberant characteristics of the room in just a few seconds. The vertical bars are spaced at the usual one-third-octave intervals, the height shows the amplitude of the decaying field, and color changes denote the passage of time. (It is also possible to use color as amplitude and height as time.) Not only can we see how quickly the field decays at any frequency, but by looking at a line of constant color (i.e., constant time), we can see the *frequency response* of the reverberant field. Just think how long it would take to acquire this data, one band at a time, and then hand-plot the response. Neat, huh?

STEREO ANALYZER

Traditionally, when an engineer reviews a master tape cutting, or sets up spaced microphone pairs, he relies on an oscilloscope to view the in- and out-of-phase information. A Lissajous pattern is used, with deflection at 45 degrees to the horizontal showing in-phase, and 135 degrees showing out-of-phase information. The usefulness of the display lies in the fact that in-phase signals indicate lateral groove motion of the disc-cutting stylus, and out-of-phase displays vertical motion. The problem with this display system is that it gives only a very general idea of what's happening. There is no exact measurement of the level of either component, let alone how the components vary with frequency. The Stereo Analyzer system consists of the regular Badap 1, with an extra set of analog cards to accommodate another channel. The two inputs may be shown in separate RTA displays, identical to the one seen in the regular RTA program, with all the same functions. Both inputs may be shown at the same time, either as two compressed RTA displays next to each other, or with the frequency bands "interleaved."

Clearly, this is convenient, but hardly justifies the product when you could sit two cheaper RTA's next to each other on the shelf. But there's something we can get from this display that is simply not available on conventional analyzers. Instead of left and right signals, we can look at the *sum* and *difference* spectra, which, as we remember, were the lateral and vertical groove motions. (Or baseband and subcarrier modulation for stereo f.m.) FIGURE 7 shows how useful this is. The sum component is shown as a red display, the difference is in yellow. The two spectra have been interleaved. Where the displays are the same level, the colors blend to produce a uniform orange. Components higher than this show up in their own color.

FIGURE 7 shows the left and right components of a coincident microphone recording of solo piano. Not surprisingly, the two channels are almost identical. In FIGURE 8, we see the sum (red) and difference (blue) components of the same recording. Again, there is no signficant anti-phase components. But look at FIGURE 9. This is a commercial rock recording in which the performers deliberately introduced variable—and out-of-phase effects. Over much of the band, the anti-phase components (yellow) are much stronger than the in-phase. The producer now has a way of *visually*, as well as audibly, monitoring these effects.

One commercial "click and pop" remover only monitors the difference of the two channels (vertical motion). Ever wonder why? FIGURE 10 *shows* why. This is the spectrum of several pops and clicks. Note how much energy there is in the difference (yellow) component, and how little in the sum (orange).

On a more-mundane level, the Stereo Analyzer may be used with two calibrated microphones to quickly establish a compromise equalization in sound systems. With the overlapping displays, it becomes very easy to see how changing the EQ affects both room positions.

This is only the beginning of the Badap 1 System. We now have an Input Multiplexer which shows the peak and average levels of 32 signals, simultaneously. The display colors may be changed in blocks of four to correspond with particular instruments or groupings. If your board is automated, a second multiplexer will allow the superimposition of the level sets on the main display. And, it should be obvious how useful this unit is in stage monitoring. (Inputs are Hi-Z, balanced, transformerless.)

Since Badap 1 is a computer, automated testing is easy. The appropriate programming lets a user run frequency response, noise, distortion, and flutter tests on a tape deck without human intervention. Add a modified multiplexer, and up to 32 tracks can be measured automatically. An RS232 interface is available to drive a plotter. If a plotter is beyond your means, an inexpensive electrographic printer may be attached to the video output. It makes a permanent copy of anything on the screen in just a few seconds. ■

KUNIMARO TANAKA

The Mitsubishi Digital Audio System

Hand-in-hand with the current surge in digital technology is the speculation that PCM will become audio's wave of the future.

AN EXAMINATION OF THE complete audio recording/reproduction chain, from microphone to speaker system, immediately reveals that the quality of the reproduced sound is subject to a number of limitations, including noise, distortion, cross-talk, and so forth.

The last few years have seen numerous attempts to achieve significant improvements in equipment performance, by the introduction of pulse-code modulation, or PCM, techniques. The introduction of PCM technology has been supported by contemporary developments in semiconductor technology, and has been making real progress which indicates that it will become the mainstream of audio engineering in the 1980s.

The potential of PCM for improving the performance of tape recording equipment was recognized by Mitsubishi Electric, and an R & D program resulted, in September 1976, in the world's first PCM tape deck operating on 6.3mm (¼-inch) tape at a tape speed of 38 cm/sec. (15 in/sec.). By October of the following year we had developed a PCM recorder for the amateur, based on use of a VCR unit, and another world first in the form of a digital audio disc (DAD) system. Since then our on-going R & D program has been directed at; a self-contained PCM recorder and DAD player for home use, featuring a semiconductor laser, and, a two-channel fixed-head professional-type tape recorder, featuring tape-cut editing. The latter received a most enthusiastic reception when it was unveiled at the New York AES Convention in November, 1979. Mitsubishi Electric is actively pursuing the adoption of PCM in audio equipment.

HOW DOES PCM RECORDING EQUIPMENT ACHIEVE HIGH QUALITY?

Conventional analog recording equipment records the original audio input signal waveform in the form of remanent magnetization of the recording tape. This means, of course, that the non-linearities of the magnetic tape distort the waveform, with noise also arising from inhomogeneities in the distribution of the magnetic domains in the tape. In addition, wow and flutter, etc. caused by inadequate mechanical precision in the tape-transport mechanism, also distorts the waveform. The result is a degradation of the original signal at playback, as shown in FIGURE 1.

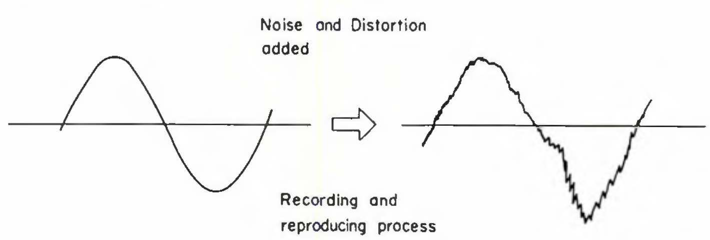
With a PCM recorder, the input signal is converted into numbers and groups of pulses, which are then recorded on the tape. Of course, the waveform of these pulses is also subject to the above-mentioned influences of noise and distortion, but these do not affect the numbers and groupings of the pulses, and so the recorded information is preserved without degradation. The original waveform is faithfully reproduced, as shown in FIGURE 2. This is the most important and basic principle of PCM recording.

The coding of the signal is performed as follows. First, the input signal is sampled at a frequency of some 45 to 50 kHz. According to Shannon's sampling theorem, a given sampling frequency permits the recording of audio frequencies up to one half of the sampling frequency. The choice of 45 to 50 kHz therefore enables recording up to 20 kHz.

Next, each sample is measured, and expressed as a binary number, consisting of a series of "0" or "1" digits. "1" is represented by a pulse and "0" by the absence of a pulse. Since errors in the measurement of the sample will appear as noise, it should be measured as accurately as possible, and expressed by a binary number with as many digits as possible. In general, the signal-to-noise ratio for a signal coded with N bits is given by:

$$S/N = 6N + 1.8 \text{ (dB)}$$

Figure 1. Degradation of analog signal.



Kunimaro Tanaka is the senior design engineer in the Product Development Laboratory—PCM Product Development for Mitsubishi Electric Corporation, Osaka, Japan.

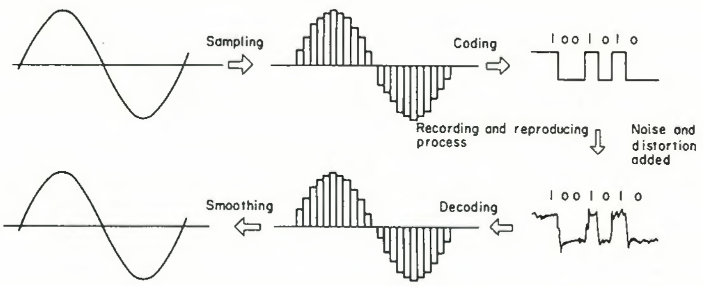


Figure 2. Principle of PCM recording.

Thus, whereas the performance of conventional analog recorders is determined by the characteristics of the magnetic recording tape and the performance of the heads, the performance of the PCM recorder is determined by the electrical circuit design.

The density of the information in PCM recording and playback equipment is generally extremely high, so that dust, etc., adhering to the surface of the magnetic recording medium (disc or tape) very readily generates error signals. Error-correction codes are therefore normally added to the signal so that, should an error arise, it will be automatically detected and corrected. If the errors are too numerous to be corrected in this way, error-concealment techniques are brought into use. These measure the error-free samples immediately before and after the error sample and substitute their average value in place of the error sample. Obviously, the effectiveness—or otherwise—of these error-correction and error-concealment functions is critically important for the stable and reliable operation of the equipment.

JUST HOW GOOD IS PCM EQUIPMENT PERFORMANCE?

As an example of the level of performance that can be obtained by the introduction of PCM techniques, we turn to a professional fixed-head recording deck, using a sampling frequency of 50.35 kHz and 16-bit coding.

FIGURE 3 gives the performance specifications, FIGURE 4(A) the frequency response, FIGURE 4(B) the distortion characteristics, and FIGURE 5 shows the changes in the amplitude of a 20 kHz sine wave. In an analog recorder, the high-frequency playback output is subject to variations in level due to varying degrees of contact between the tape surface and the heads. This is completely absent in the PCM recorder.

By incorporating a buffer memory in the playback circuit, the effects of wow and flutter are eliminated. In a PCM recorder, wow and flutter is no more than the "inaccuracy" of the quartz-crystal oscillator clock. This means that a one-hour recording program will play back over precisely the same length of time, to within the order of one milli-second.

Figure 3. Performance specifications of fixed-head PCM recording deck.

Dynamic range	Better than 90 dB
Frequency response	10 Hz-20 kHz (±0.5dB)
Total distortion	0.02% (at peak level)
Crosstalk	–85dB (1 kHz)
Wow and flutter	Only limited by quartz crystal oscillator
Playback signal level variation	None
Print through	None
Residual level after erasure	None

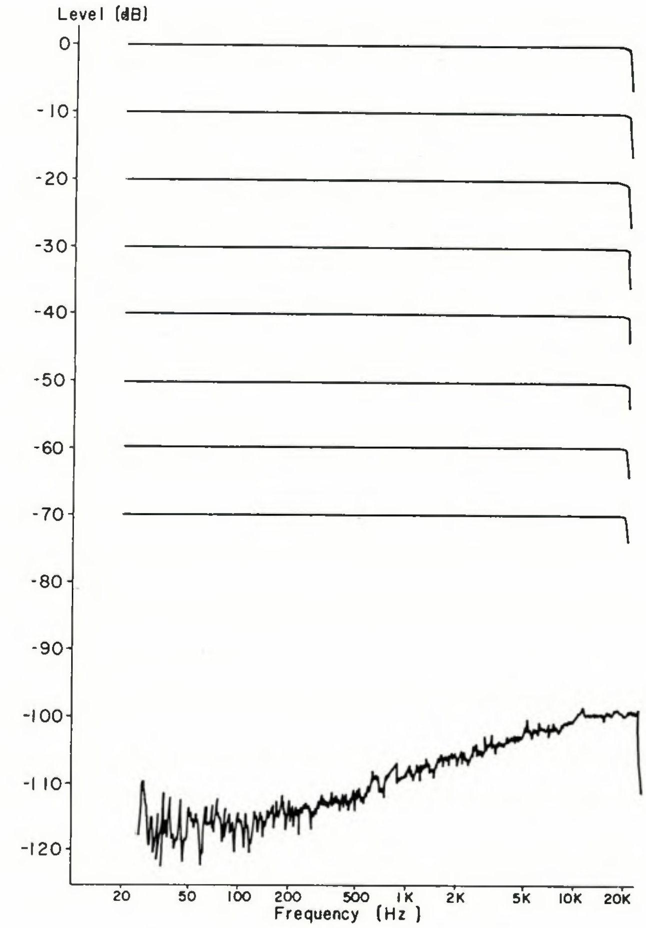


Figure 4(A). Frequency response.

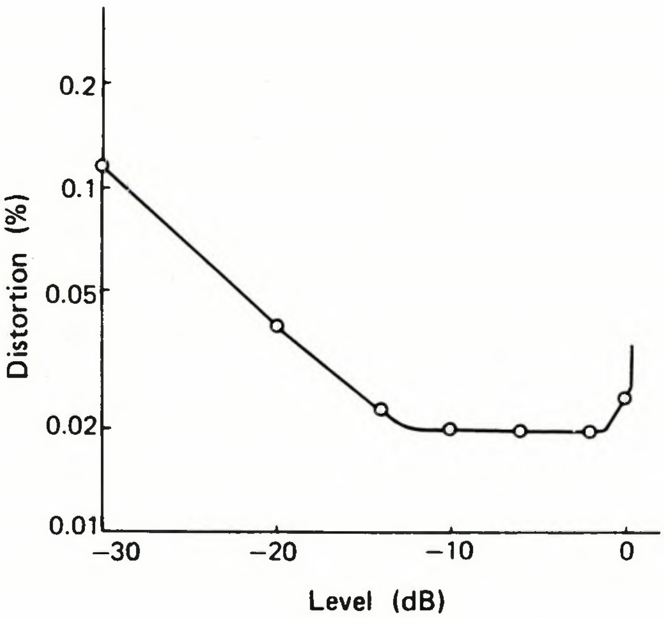
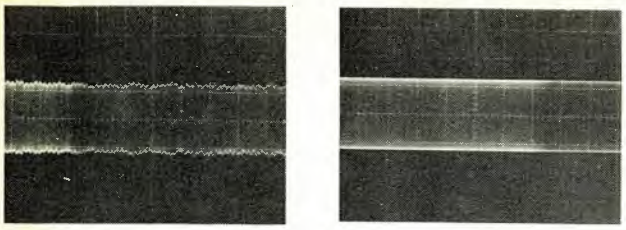


Figure 4(B). Output level distortion characteristics.

Dubbing is performed with the digital signal itself, so that there is very little degradation of the signal. Even after 40 or 50 repeated dubbings, there is no detectable change. The reproduced sound is also free of print-through, and no ghosts are left after erasure. There is negligible crosstalk, and no difference between the characteristics of the left- and right-hand channels. These are advantages of a very high order indeed.

(For more on PCM, see **Digital Modulation For High-Quality Audio,** in the June, 1978 **db**—Ed.)



(A) Analog recorder (B) PCM recorder

Figure 5. Amplitude fluctuation.

WHAT KINDS OF PCM RECORDING AND PLAYBACK EQUIPMENT ARE CURRENTLY AVAILABLE?

PCM recording and playback equipment may be divided into PCM decks for recording, and digital audio disc equipment (the latter corresponding to conventional analog audio discs). PCM recording decks may be further subdivided into fixed-head and rotating-head equipment. The features of the fixed-head and rotating-head equipment are given in FIGURE 6. The fixed-head types are suitable for professional use, and the rotating-head types, because of the comparative simplicity of the circuitry used, are more suitable for domestic use.

FIXED-HEAD PCM RECORDING DECK

FIGURE 7 shows the external appearance of a two-channel professional recording deck. Specifications are listed in FIGURE 8.[1] As is clear from the photograph, it is similar in appearance to current analog decks. It offers simultaneous monitoring and tape-cut editing functions, and can be operated in virtually the same way as a conventional analog deck. The choice of a tape speed of 38 cm/sec. (15 in/sec.) means that, for audio channels, the PCM signal must be distributed between eight tracks. This keeps the recording density along each track at some 20 kbit/inch.

(For more on PCM track formats, see **A Proposed Digital Audio Format,** in the November, 1978 **db**—Ed.) Errors originating in the tape will generally affect some 100 to 200 bits of data along a signal track. However, the use of multi-track recording means such errors are distributed more-or-less randomly along each track, so that the addition of an error-correction code across the width of the tape is highly effective for correcting errors. This deck uses a Reed-Solomon error correction code across the width of the tape, with a cyclic-redundancy check along the length of the tape. The extremely powerful error-correction utilized in this deck ensures uniformly high performance as follows:

1. Even if one track fails, almost all error signals are corrected with no resultant deterioration in the quality of the reproduced sound.

Figure 6. A comparison of fixed- and rotating-head decks.

Rotating - head type	Fixed - head type
+ High recording densities possible	+ Splice editing possible
+ Can be compact	+ Error correction using multi-track feature
+ Few circuit elements	+ Little operating noise
- High operating noise level	+ Fast duplication possible
- Fast duplication difficult	+ SIM.SYNC possible
- SIM.SYNC difficult	- Current models have many circuit elements
- Some equipment has slow start-up	- Currently high price

Note: a "+" indicates an advantage, a "–", a disadvantage.

Figure 7. Two channel fixed-head PCM recording deck.

Sampling frequency	50.349645714 kHz
Bit number	16 bits (linear)
Recording tracks	8 PCM (= 2 audio) 1 Auxiliary analog 1 SMPTE
Recording bit density	797 bit/mm (20240 bpi)
Tape speed	38.1 cm/sec (15 in/sec)
Magnetic tape	6.3 mm high-density tape

Figure 8. Specifications for a two channel fixed-head PCM recording deck.

2. Even when errors are too numerous for correction, error concealment by interpolation is performed. Computer simulation studies indicate that with the usual type of tape in normal use, the need for error concealment should only arise about once a year, and that it would be about 10,000 years before this could be expected to give rise to audible noise!
3. Even with tape-cut editing and splicing, the errors associated with the splice will be corrected.

There are two methods of editing recordings made with PCM recording decks: the tape-cut editing that is used with conventional analog decks, and the electronic editing that has become possible for the first time with the introduction of PCM equipment. The comparative advantages and disadvantages of these two methods are listed in FIGURE 9.

The method of tape-cut editing employed in this PCM deck is almost identical with that used for current analog equipment. There is no need to "develop" the recorded pattern as with, for example, the Quadraplex video tape recorder. FIGURE 10A shows an example of zeroth order extrapolation at the splice, and FIGURE 10(B) shows the join achieved with cross-fading. Our deck naturally uses the latter.

FIGURE 11 shows an electronic editor with automatic editing facilities. It controls one recording deck and three playback decks, so as to store instructions for 99 different editing points that are edited automatically in succession.

Figure 9. Comparison between tape-cut editing and electronic editing.

Item	Electronic Editing	Tape-Cut Editing
Method	To copy original tape selectively to slave tape.	To cut tapes physically and splice them with splicing tape.
Equipment needed	Two or three PCM recorders and editing adapter.	One PCM recorder and splicing block.
Editing time required for one hour program	One hour + time required for editing procedure times number of edits.	Time required for editing procedure, times number of edit.
Readjustment of sound quality	Possible	Impossible
Technique needed	Editing is performed by pushing button.	Identical to the analog type.

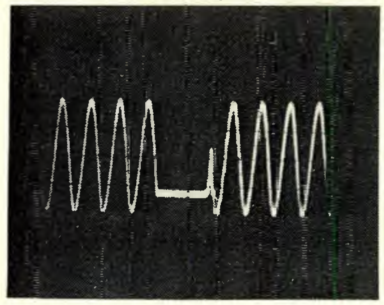


Figure 10(A). Zeroth order extrapolation.

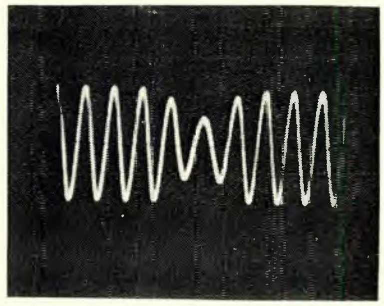


Figure 10(B). Cross fading.

ROTATING-HEAD PCM RECORDING DECK

The Electronic Industries Association of Japan, in order to stimulate the distribution of PCM recorders for home use, produced standards designed to ensure the interchangeability of PCM recording equipment based on the use of VCRs, in June of 1979. The time of home PCM recording equipment is rapidly approaching. FIGURE 12 shows a PCM adaptor[3] which conforms to the EIAJ standards, and FIGURE 13 lists the specifications.

For this adaptor to become popular, prerecorded cassettes must become available. If the master recordings for these cassettes are prepared on the professional PCM deck introduced above, the difference in the sampling frequencies means

Figure 11. Electronic editor.

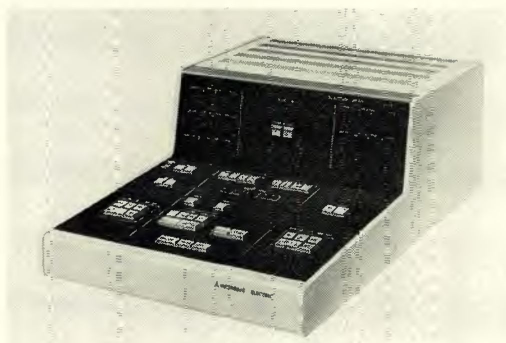


Figure 12. PCM adapter.

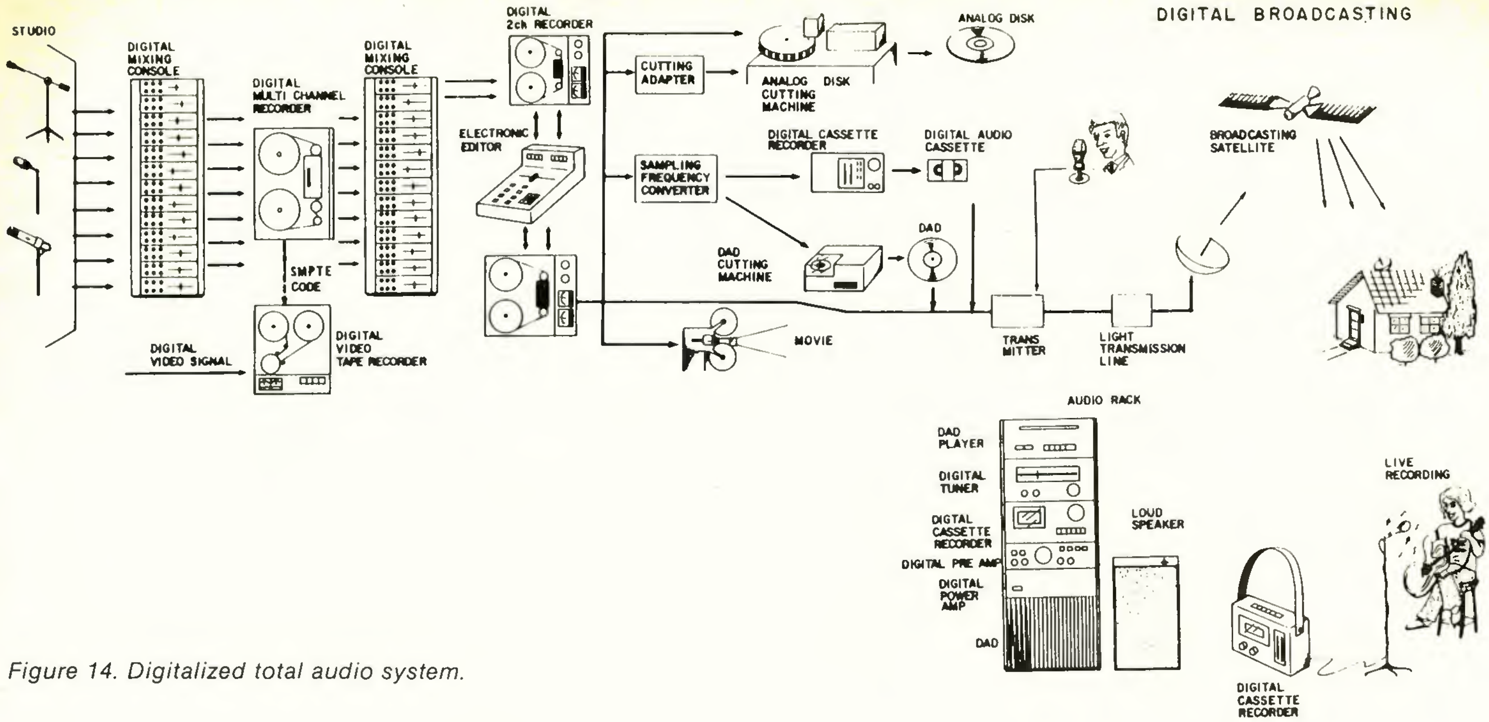


Figure 14. Digitalized total audio system.

Sampling Frequency	44.056 kHz
Transmission Rate	2.643 Mbps
Quantization (Both Encoder and Decoder)	D-1: 3 segment 12 bit companding D-1L: 14 bit linear
Wow and Flutter	Undetectable
Dropout Compensation	Correction by interleaved b-Adjacent error correction code with CRC
Signal Format	Based on the EIAJ Technical File STC-007
Conforming VTRs	VHS, β, U etc.

Figure 13. Specifications of PCM adapter.

that the digital signal cannot be dubbed as is, but must be converted back to an analog signal before dubbing, with an unavoidable slight loss of quality. However, if a sampling frequency converter is connected, the output supplies a signal in the video format, with the 44.056 kHz sampling frequency, and can be recorded as it is for VCRs.

FUTURE PROSPECTS FOR PCM IN AUDIO ENGINEERING

In addition to the above equipment, Mitsubishi Electric is also developing a multichannel fixed-head PCM recording deck, and tentative specifications include the use of 25.4mm (1-inch) tape moving at 76 cm/sec. (30 in/sec.), and giving 32 audio channels. We are directing our efforts at the development of a total audio system based on PCM technology, and are pursuing an active research and development program. The underlying concept of the total audio system is as shown in FIGURE 14, involving conversion of the signal from the microphone into a PCM signal so that mixing, recording, transmission, etc., will eventually result in reproduction of the very highest quality from the speaker system. ■

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1. Kunimaro Tanaka et al, "Improved Two Channel PCM Tape Recorder for Professional Use," Preprint of AES 1533 (G-3), 1979.
2. T. Inoue, "Comparison of Performances Between IPC Cord and RSC Code When Applied to PCM Tape Recorders," Preprint of AES 1541 (H-5), 1979.
3. Yoshinobu Ishida et al, "A PCM Digital Audio Processor for Home Use VTRs," Preprint of AES 1528 (G-6), 1979.

ROBERT J. YOUNGQUIST

Electronic Digital Editing For Multi-track Comes of Age

With digital recording comes the requirement for a new method of editing. Electronic editing now available for this purpose offers something more than just a replacement for the cut-and-splice technique of analog.

IT WAS JUST A YEAR ago this month that recording studios began installing 3M's 32-track digital mastering system. But even before those historic first deliveries, it had become obvious that an electronic editing system was needed, not only to advance the state-of-the-art, but to protect the data recorded on the digital tapes. As artists, producers and engineers began recording digitally, it soon became clear that studio acceptance and day-to-day use of the new digital recording technology could not be expected, until a viable editing system was made available. As converts to the multiple benefits of digital recording, studios wanted to preserve the full quality of their tapes by avoiding the transfer to analog for conventional editing.

Today, an electronic digital editing system is available which offers—according to those who used and critically evaluated the preproduction prototype—exceptional precision, risk-free preview capability, splice-free masters, digital's lack of degradation throughout the process, plus a new vista for editing creativity.

DEVELOPMENT OF THE SYSTEM

Initially, 3M contracted ITX to help it develop a system. At the Winter '78 Audio Engineering Society convention, a preliminary prototype of a deluxe automated system was shown which incorporates a graphic video representation of the music to verify those points that, in theory, are most suitable for editing. Work on this system, however, was later set aside in favor of a much simpler, easier-to-use system developed in 3M's St. Paul laboratory. A working prototype of this second system was completed late last summer.

This latter unit took into account studio preference for an editing system whose operation would be relatively simple and easily learned. The system also needed to be of compact size, feature semi-automatic edit-point cueing and fine tuning for smooth transitions, be of reasonable cost and adaptable to possible future options.

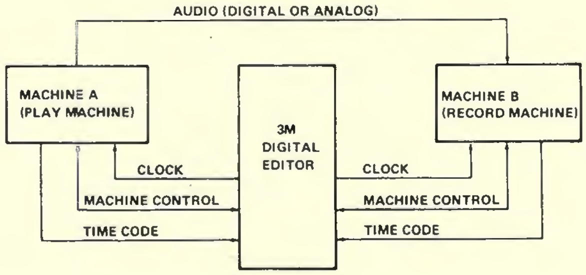
The prototype was employed in several major projects during the fall, to permit some critical field evaluation and constructive user-feedback. As a result, a number of refinements were made, primarily involving the human interface. Control functions were combined, repositioned and relabeled to make identification of the contents and their operation more convenient and natural. This streamlining also had the side benefits of making size reduction of the console possible and reducing its basic cost.

DESCRIPTION OF THE SYSTEM

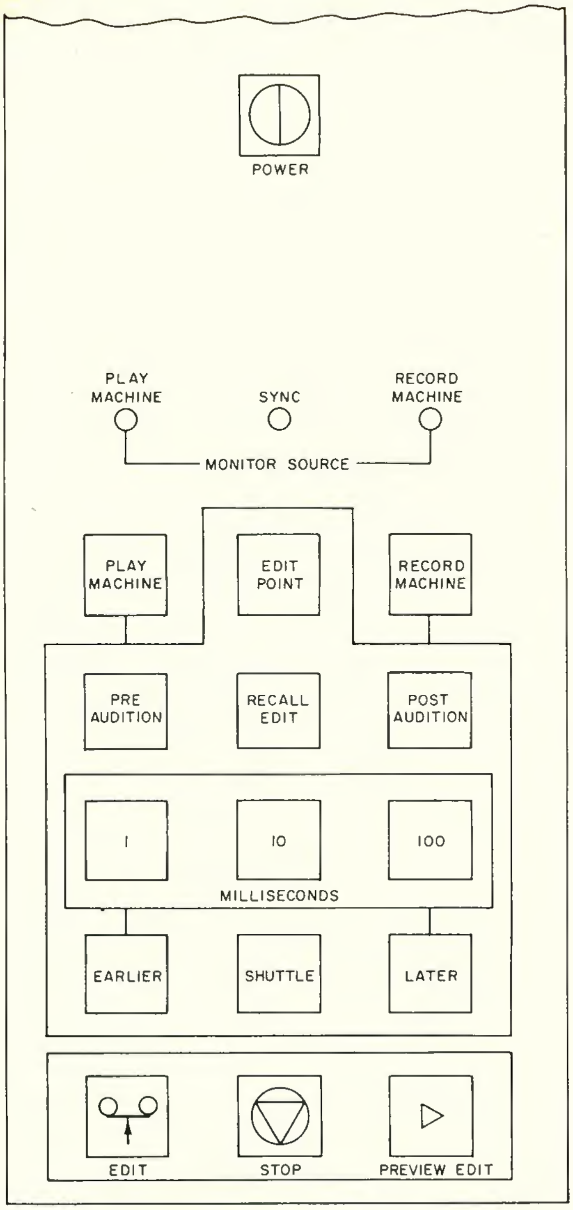
Digital editing has a number of advantages over the traditional razor blade-and-splicing-tape editing necessary with analog masters. The most obvious one is that, with electronic editing, the original tapes remain physically unaltered, so their structural integrity is never compromised by potential physical or magnetic damage.

But, perhaps more importantly in the view of the producer is the ability to select tentative edit points, and preview them repeatedly, separately or as the proposed edit. Any edit point may be refined by moving it in either direction as little as one-thousandth of a second. Then, the final aesthetic choice of an edit-point may be executed upon command of a single button.

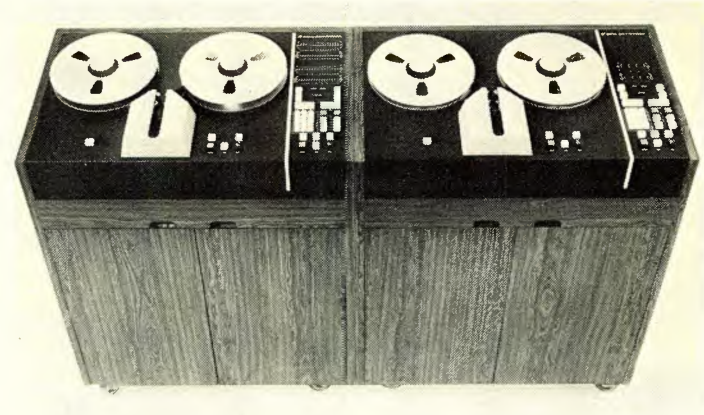
Digital editor block diagram.



Robert J. Youngquist is the research manager responsible for the development of digital audio equipment in 3M's Mincom Division, St. Paul, MN.



3M's digital editor control panel.



3M 32- and 4-track digital mastering system.

This precise and easy control is in part achieved by the microprocessor, the miniature dedicated-computer system or the brain, that performs the logic and a number of housekeeping chores on behalf of the operator.

The key to electronic editing is a time code—the mileposts, if you will—recorded on one track of both recorders. This code, laid down automatically during recording, writes a 24-digit binary number every 320 microseconds; thus, every three-ten thousandths of a second, there's an electronically identifiable location on the tape.

As the tapes are reviewed for potential edit points, the time code is constantly monitored by the editor's circuitry. When the human editor punches the Edit Point button for the machine he's monitoring, the time code number is captured and stored, ready for reference.

The tape adjacent to each edit point can be auditioned as many times as desired, before finalizing the edit. Use of the Shuttle button with either Pre-Audition or Post-Audition continually shuttles the selected segment through a two second sequence, muting the sound either before or after the edit point, as appropriate. Activation of the Edit Recall button alone recalls the last edit made on the play or record machine.

At this stage, it is quite likely that the producer will want to move one or both edit points slightly; it's easily done, forward or backward, in increments of 1, 3, 10, 30 or 100 milliseconds. For this operation, the producer commands the direction with either the Earlier or Later button, then the time through one, or a combination, of the time commands. For example, pressing the one-millisecond button three times moves the tape a distance equivalent to three milliseconds in the designated direction. However, this may be shortcut, because pushing the 1 and 10 buttons simultaneously will also move the edit point three milliseconds. Likewise, pushing the 10 and 100 millisecond buttons simultaneously results in a 30-millisecond move.

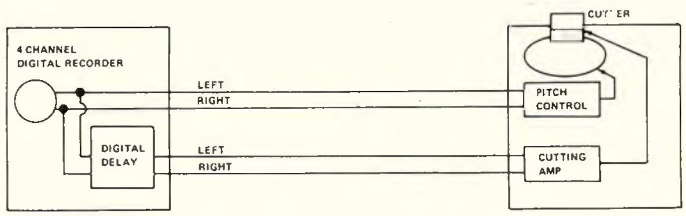
When both edit points are thought to be final, the edit can be previewed in full by punching the Preview Edit button, which causes both tape machines to rewind to a point ten seconds before the selected edit point. Then, the record machine plays back the first take to the edit point, where it mutes its sound, while the music is heard starting from the second edit point on the tape of the play machine. These segments of the two tapes will shuttle back and forth across the edit point in sync without the actual edit being recorded. Each time, the producer hears how the final edit will sound.

If satisfactory, then all that remains is to push the Edit button. This also starts the recorders 10 seconds before the edit point; where the point is reached, the record machine is automatically put into the recording mode and the dubbing begins. When the end of the second take is reached, the Stop button halts the tape motion of both machines and the next source can be cued up.

Obviously, there is a transition from the old to new program data at the edit point. The error-correcting method which permits the 3M mastering system to record 32 tracks on a one-inch tape has been designed so that the signal transition can be smoothly accomplished. If at a later time, an error (drop-out) were to develop at this location, however, it is conceivable that incorrect reconstruction would occur, since some of the non-adjacent error-correcting data has been removed by the edit.

To avoid this remote possibility, it is recommended that the completed master tape be digitally dubbed. This will recreate new error-correction data for all program material.

Block diagram of 3M's digital preview system.



EXPERIENCE

At the time of this writing, studio experience has involved the prototype unit, with the exception of continuous demonstrations of the first production unit during the most recent Winter AES. The applications were varied, involving several major popular and classical music projects. The editor was used to edit prior to mix-down—16 track to 16 track—and during mix-down of a Columbia Masterworks album. It was also used during a 32 to 2-track mix-down of a tune at Sound 80, Minneapolis, and for editing after mix-down of a single and an album at Record Plant, Los Angeles.

Reactions of producers and engineers who used the system were positive. They collectively concurred that, while proficiency naturally increased with familiarity and practice, electronic editing didn't require learning a new technology. As with analog, the editing job still consists primarily of artistic organization of takes, selection of edit points and decisions as to the other creative criteria by the professional's ear.

Rather than paraphrase user comments, it is perhaps more informative to discuss specific applications and include their exact comments.

The first user was Andrew Kazdin, producer of a Columbia Masterworks recording of the New York Philharmonic Orchestra playing "Petrouchka," conducted by Zubin Mehta. After recording the work at New York's Avery Fisher Hall, on 16 of the digital pre-mix machine's 32 tracks (with the other 16 tracks used as backup), Kazdin awaited completion of the editing equipment.

The prototype editor was used for fine-editing work before and during mix-down. What was his reaction? "Electronic editing offers additional creative flexibility and potential. It wouldn't be honest to say it was all easy, however. As a new technique involving unfamiliar equipment, there were obviously certain things to learn and explore.

Seated before a portable mixing console and a prototype of 3M's electronic digital editing system (foreground) is Andrew Kazdin, producer of a New York Philharmonic Orchestra recording. The editor, since refined as production equipment, was used during a multi-track editing session at 3M's St. Paul research facility. Bob Youngquist observes.

"For instance, I was a bit surprised to discover that some edits, that would have been relatively easy conventionally, were a bit more difficult to achieve in the digital domain. But, conversely, many that would have been difficult or impossible in analog were made with relative ease. This all points out that this is a new technique with some similar and different capabilities," Kazdin said.

Kazdin went on to comment upon digital in general, as he had not heard the "Petrouchka" tapes since a brief playback at the recording sessions. He indicated that reviewing the tapes reconfirmed digital's unexcelled sound purity, and the absence of noise and distortion. The absence of wow and flutter was a particularly notable benefit on stringed passages.

Herb Pilhofer, composer/musician/producer and principal of Sound 80, Minneapolis, referred to electronic digital editing after his brief experience with it as "more than just a substitute for cutting and splicing." In fact, he noted with surprise, "It becomes a very-useful musical tool. We wanted to overlap certain sections and to make some fairly-precise edits of a musical nature where I would usually hesitate to do that with a blade. The ability to rehearse edits and sample the results of an edit in several locations is a new luxury."

Pilhofer said, "I see a trend; electronic editing opens up some new avenues for music production not before feasible, and I predict that a lot of things are made possible for which we haven't yet found any practical application."

His viewpoint was echoed by Scott Rivard, producer and chief engineer of Sound 80, who used the editing system during mix-down. "We were assembling some very complicated mixes on the 4-track and the editor allowed us to do this very effectively." He also indicated that he was able to use the system after very little practice, although real proficiency came later.

At Record Plant, engineer Mike Stone began his sequence for the Bonnie Pointer editing sessions following mix-down, using the 4-track master recorder as the playback source. Assembly then took place on two tracks of the 32-track pre-mix machine. After the assembly was complete, the recorder's roles were reversed; the tape was played back by the 32-track recorder for one continuous dub onto the 4-track unit. As the 4-track recorded the program material, it also automatically generated new error-correcting parity code blocks to protect against possible future dropouts at edit locations.

Stone used a metronome to sync the performance on individual tracks. Because of that perfect tempo, he could feel when very slight timing adjustments were necessary. "Since we could adjust in milliseconds, we could get exactly what we wanted—and you can't get that kind of accuracy with analog editing." Stone also felt it was relatively easy to learn digital editing. "If you have a good feel for punching in, your proficiency can develop fairly readily," he said. He added that he feels a practiced engineer can soon learn to edit digitally as rapidly as he can in analog—and, of course, with more precision.

The two tracks were then used to feed the new, optional 3M Digital Preview Unit, which digitally delays program material to the cutting lathe by a selected amount (from 5 milliseconds up to 1.96 seconds). The undelayed signal is fed to the automatic pitch-control mechanism and the delayed signal to the lathe cutting head. Since the signal is digitally delayed, no loss of quality occurs.

SUMMARY

As a result of the field experience and refinements thus incorporated, production editing equipment is now installed in studios for use on a day-to-day basis. This fills in the missing link critical to the ultimate viability of digital recording for many producer and artist groups.

In the process of creating this editing system, we have learned of additional studio needs and have begun development work in some of these areas. One of these may be a cross-fade option; a prototype of this is just about ready for evaluation. And, with the enthusiasm toward digital production techniques increasing and 3M's commitment to make conversion to digital complete, there is even more to come. ■

ROBERT HALSALL & DR. HUBERT S. HOWE, JR.

A Computerized Recording Studio Console

With complex mixes exceeding the level of human dexterity, relief is in sight as the microcomputer takes charge of functions previously relegated only to human intervention.

AS WE ALL KNOW, the number of channels on multi-track tape has continued to grow over the last several years. Not too long ago, 16 channels were standard. Now, 24 channels are the norm, and many studios are investigating the possibility of upgrading to 32 channels, or to 48, by combining two 24-channel machines.

The main reason for expanding the number of channels is to increase flexibility in mixing. More channels enables you to postpone more mixing decisions until the musicians have left the scene and you can try out different combinations. We mention these obvious facts here, simply to point out that studios continue to search for new ways to increase the flexibility of their systems, and generally they will expand to whatever level of technology is available.

The system we describe in this article is also geared to increase flexibility in mixing, but in a much more dramatic way than simply increasing the number of available channels. It uses a microcomputer to take over certain functions that were previously relegated only to human intervention. We expect that ultimately the computer will be able to control *all* functions that are presently regulated by humans, but at the moment only the most important functions—level and equalization—will be implemented. In the near future, these functions will be expanded to include at least switching (channel assignment), panning (level control across a stereo field), and control of outboard equipment (special effects, delay lines, reverberation, etc.).

The system that we describe here has been designed, and will be marketed, by one of the authors (Halsall), so we have a more-than-passing interest in it. When comparing it to others, we will refer to it as "our" system.

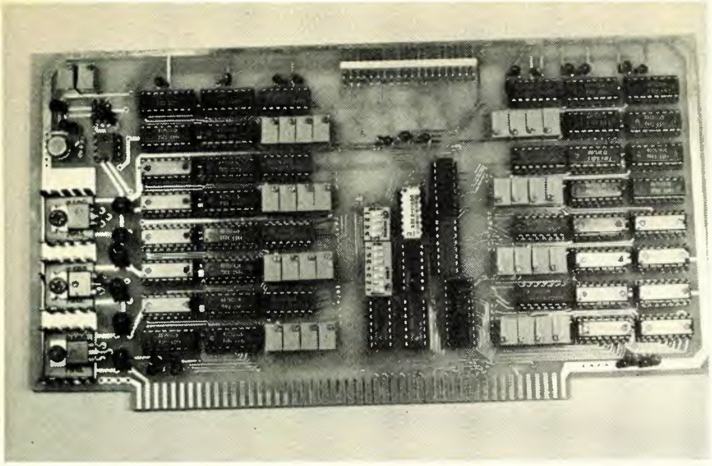
Robert Halsall is president of the audio consulting firm of Robert R. Halsall Associates, Spring Valley, NY. Dr. Hubert S. Howe, Jr. is a composer—teaching music at both Julliard and Queens College, NY. Currently, Mr. Halsall and Dr. Howe, Jr. are working, jointly, on the synthesis of computer-generated music.

The way in which any mixing system works is of crucial importance to the recording engineer. Presently, the engineer is expected to control a larger and larger number of channels, and the task becomes more difficult as the number gets larger. With a manually-controlled system, all fader adjustments must be made dynamically, during the course of a mix. But, there is no way a human operator can accurately control 48 faders at once.

COMPUTER-CONTROLLED FADERS

However, when the faders are controlled by a computer, the operator has only to perform a level adjustment once. As he does so, the computer stores the information that he transmits to the faders and it will repeat his exact movements on each subsequent mix, while he adjusts other channels. In every other respect, performing at the computer-controlled mixing console is pretty much the same task as the recording engineer is already accustomed to. Therefore, he may approach the mix in several possible ways. For example, he can preset all channels to an approximate position, and then let the computer play the music

Shown on the workbench, in Bob Halsall's laboratory, is the cabinet that houses the Supermix System Computer. Extremely compact, the cabinet's dimensions are 19" x 7".



The Digital-to-Analog Converter (DAC) board contains 32 8-bit DACs used to control the faders on the mixing console.

back again while he continues to make refinements until that "perfect mix" is achieved. Another method would involve working with different groups of instruments in isolation from the others and then combining them. When a recording studio is controlled by a computer, numerous operating systems can be implemented by merely changing the software.

In addition to this increased power in the mixing system, we do not want to fail to mention one of the most important side benefits of the computer-controlled studio; the computer remains usable as a business machine, and can handle all the accounting, payroll, inventory, etc. of the studio. Theoretically, all of this could even be implemented in a multi-programming task loop, so that these operations might be carried on *simultaneously* with a mix. Practically, however, this would probably place too much strain on the system, so normally these functions would be run separately.

Several console automation systems are presently being manufactured, by Harrison, MCI, Allison, Neve and others, that lend themselves to computer control. Most larger recording studios now have equipment to which a computer system can be hooked up.

TWO SYSTEMS

Presently, there are two generic systems of computerized mixing for recording studios. One method (not ours) takes data from the faders and writes it on one track of the multitrack tape recorder. This method has at least two major disadvantages: (1) It works nicely when you have only one fader to worry about, but things may get slightly out of synchronization when several channels are controlled simultaneously. By the twentieth fader, the error can be 50 to 100 milliseconds—not a long time, but enough to be audible. (2) Each separate mix requires another track on the multi-track tape recorder. Two mixes require two tracks, three require three tracks, etc. This factor inhibits experimentation with different possibilities and ultimately restricts the flexibility of the mix.

Our system employs a floppy disc as an integral part of the recording process. All dynamically-changing (analog) information is stored on the disc in parametric digital form. No matter how many mixes there are, only one audio track is used for recording of the timing information that synchronizes the computer with the tape recorder. The computer generates and reads SMPTE time code, which resolves system timing to within one television frame (i.e., 33.33 milliseconds). The computer thus makes an assumption that changes which occur during a frame happen at the frame boundary, resulting in a resolution of one frame, and a mean error of 16.67 milliseconds. This resolution never changes, no matter how many changes occur within the frame. Since the computer generates and reads SMPTE code, it can also run synchronously with video tape recorders.

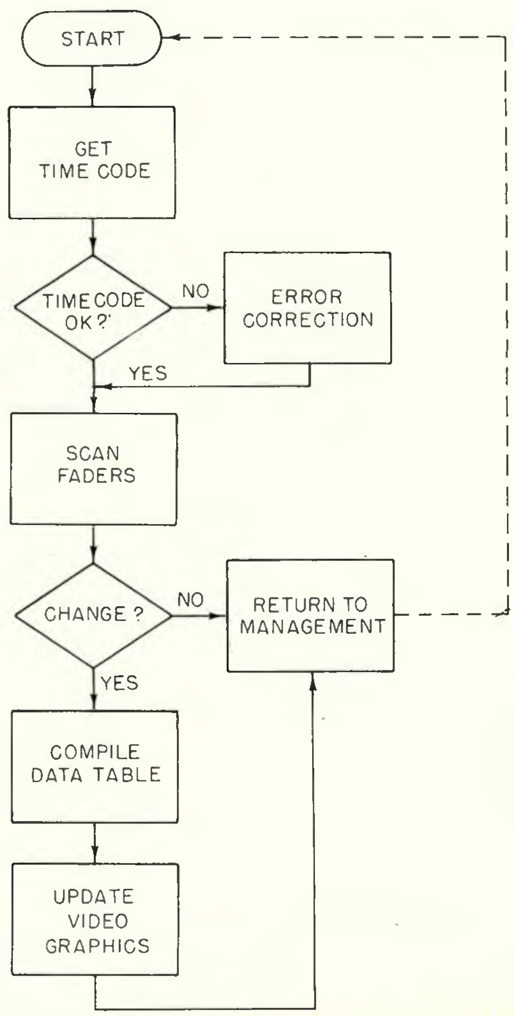
The main feature of our system is that it uses a floppy disc (as opposed to the master tape itself) to store mix information. By sorting data on the disc, it is possible to go back and play different sections from different mixes, thus creating hybrid mixes. You can identify the instrument group, mix number, and time. The concept gets a bit difficult to visualize because it is so flexible, but with a little experience the value of this flexibility becomes clear.

THE COMPUTER SUB-SYSTEMS

All of the main computer hardware used by this system is standard, so that the same computer can run normal accounting and business software as well. The special hardware required for controlling the console—each item on a standard S-100 bus board—is as follows:

1. A time-code generator and reader. It writes and reads synchronization code on the audio tape recorder.
2. Analog-to-digital converters. These do data acquisition for the faders and any other equipment that has analog voltage outputs. The conversion is 8-bit, providing 256 (2^8) discrete levels, to be specified. The analog-to-digital sub-system incorporates a Z-80 micro-processor to do the data acquisition, while the main CPU takes care of other business.
3. Digital-to-analog converters (DACs). These do all the work of outputting data to control the faders or other equipment. The controls for one device can be set in about 2 microseconds.
4. Video display controller. This unit presents an interactive graphic display of the level information in each channel of the

Figure 1. Flowchart of the program section for recording data.



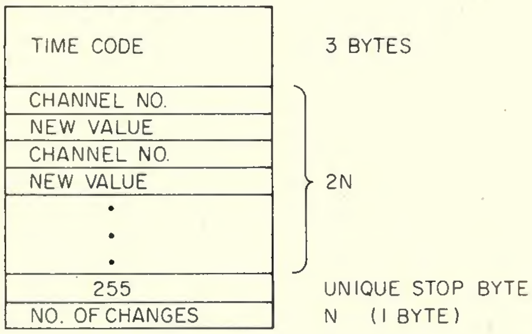


Figure 2. Table format for a single frame. N changes occur within the frame, requiring 2N + 5 bytes. Several tables are packed in one 256-byte record on the disc.

tape recorder. It is also a "smart" controller, with an 8085 processor on board, making it very fast from the main CPU's point of view. The CPU (Central Processing Unit) has to output only the color, start of line, and end of line; the rest is controlled by the 8085. The controller hooks up to any standard color television, and the graphics are software-controllable to a 192 by 256 density—the highest possible resolution with a standard color monitor. The video displays the fader position (height) and updates all channels in real time. It looks like animation. (Also, the computer can play the game "starwars," using the faders on the console to aim guns!)

We have been able to realize the computing power of mini-computers for this application through the use of distributed processing. Each of the sub-systems described above possesses an entire microcomputer dedicated to a particular task, freeing the main CPU for data management. (The price of systems based on micros is about half of those based on minis.)

FIGURE 1 shows a flowchart of the portion of the program that handles the recording of data. This routine is entered on an interrupt basis whenever the time code generator/reader receives a new frame from the tape recorder. An "interrupt" is a feature of the computer whereby it can respond to an external condition by suspending whatever it is doing, to go to a routine which services the device that generated the interrupt. At the conclusion of the interrupt service routine, the computer returns to the "management" portion of the program, which resumes whatever the computer was doing at the time of the interrupt.

Each symbol in the flowchart describes a separate function or decision point in the routine. After the computer receives the time code, it checks to see whether it is valid. If not, there are several ways in which the code might be corrected, so these are attempted first. (Errors occur very infrequently, and are usually caused by oxide dropouts on the audio tape recorder.) If the error correction routine is unsuccessful, an error message is printed and the operator is requested to regenerate the code. If the code is good, the faders are scanned to see if anything is different from the previous frame. If not, the routine is complete, and control is returned to management. If there are changes, a table is compiled (see FIGURE 2), and the video graphics are updated to reflect the changes in fader positions.

FIGURE 2 shows the format of the table that is stored on the floppy disc. Several tables are created and stored in an active memory buffer until there are enough tables to fill one record (256 bytes) on the disc. When the buffer is full, the computer starts filling an alternate buffer, while it writes the full buffer to the disc. Each table describes the dynamic conditions at the recording console for a particular frame. At the head of the table is the time code, packed in a three-byte format that is more efficient than the SMPTE time code that comes off the audio tape. The time code is followed by pairs of bytes indicating the console channel number and new value. After all changes have been entered into the table, a "stop byte" is inserted, followed by a byte indicating the number of changes during the frame. This scheme limits the system to controlling 255 different channels—probably enough to accommodate consoles for the indefinite future!

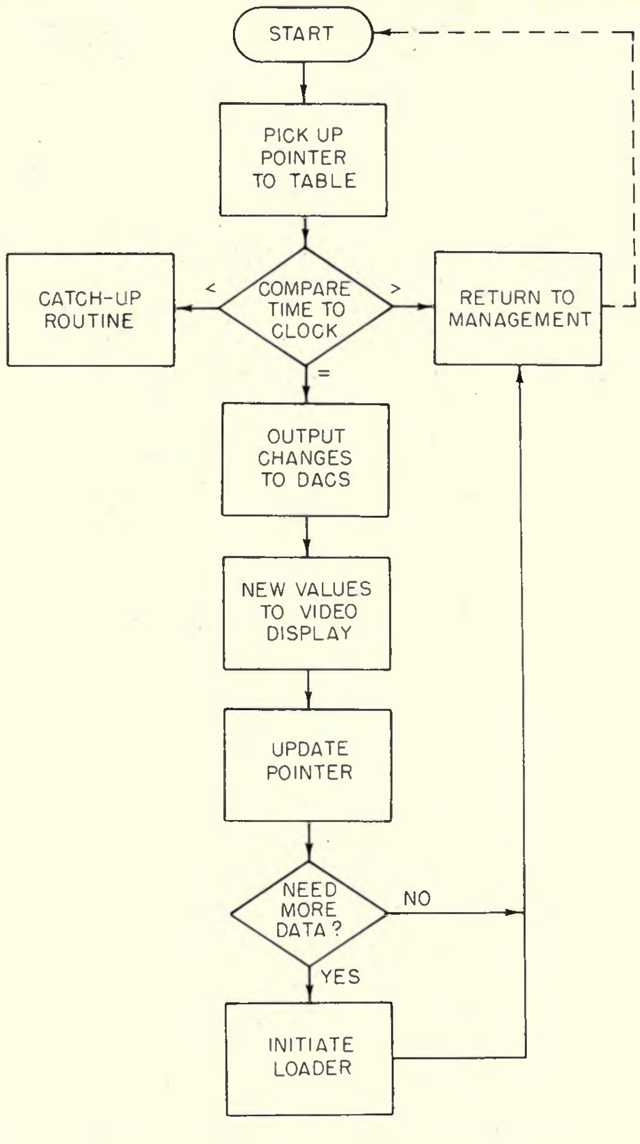


Figure 3. Flowchart of program section for playback.

FIGURE 3 shows the routine that handles playback. It is entered on an interrupt basis, just like the recording routine shown in FIGURE 1. At the beginning, it picks up a pointer that indicates the memory location of the first table. If the three-byte time value in this table is later than (greater than) the current time (which is read from a real-time clock), no action is taken, and the computer returns to the management tasks. If the time is equal to the current time, new levels are output to the faders and the video display, and the pointer is incremented to the next table. Also, if the record in memory is then exhausted, a read request is initiated, and the pointer is set to use the alternate buffer while the old buffer is being refilled. If the time indicated by the table is earlier than (less than) the current time, this task is abandoned and control is given to a special "catch up" routine. In practice, this only happens if you start playing back the tape recorder in the middle of the mix, as after a fast-forward operation. Typical catch-up times for a minute of program material are on the order of 300 milliseconds.

In the process of developing this system, many software techniques were developed that increase the power and flexibility of the system without using any additional hardware. The main trick was to get the floppy disc to work in a real-time data acquisition application. While many of these facts are of crucial importance to the operation of the system, they would be the subject of another article. ■

Digital Tape Recorder Directory

MCI, Inc.
4007 N.E. 6th Avenue
Fort Lauderdale, Florida 33334
(305) 566-2853
JH-220 Digital Recorder,
7 tracks, ½-inch tape

Mitsubishi Electric Corporation
Melco Sales, Inc. (Distributor)
3010 East Victoria Street
Compton, California 90221
(213) 537-7132
and
7045 North Ridgeway Avenue
Lincolnwood, Illinois 60645
(312) 973-2000
X-80 Recorder
XE-1 Electronic Editor
10 tracks (8 digital, 1 mono analog, 1 SMPTE Code)
¼-inch tape

Sony Corporation
9 West 57th Street
New York, New York 10019
(212) 371-5800
PCM Digital Recorders
(See db Special Report for Details)

Soundstream Digital Recording System
34 South 600 East
Salt Lake City, Utah 84102
(801) 355-9610
Soundstream Digital Tape Recorder and Editor
2 or 4 track, 1-inch tape

Technics
Panasonic Company
One Panasonic Way
Secaucus, New Jersey 07094
(201) 348-7000
4-channel, ¼-inch Digital Recorder/Reproducer and Digital Editor

PCM Recording Processor SH-P1, Panasonic Omnivision Recorder, VHS cassette tape

3M Company
3M Center, Mincom Division
Saint Paul, Minnesota 55101
(612) 733-0301
3M Digital Mastering System
32-track, 1-inch Tape Recorder
4-track, ½-inch Tape Recorder
Digital Editing System

JOHN M. WORAM

The Digital Format Dilemma

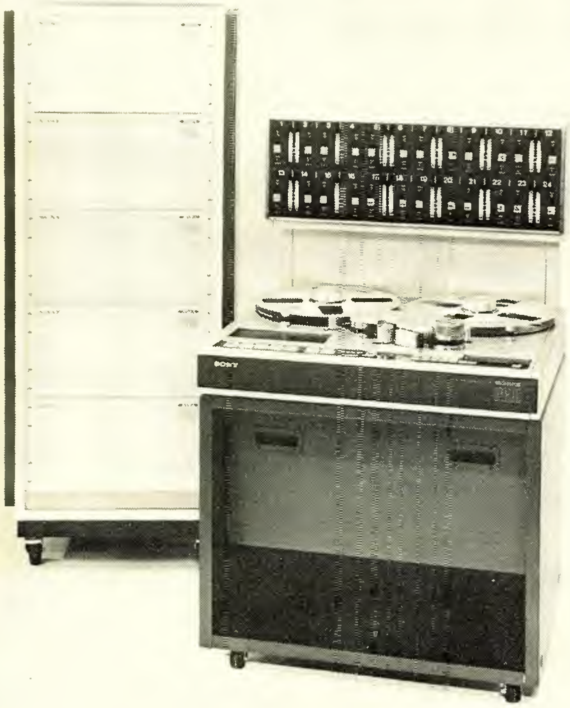
ALTHOUGH THERE CAN be no question that the digital tape recorder is here to stay, there certainly can be questions about some of its operating characteristics. Obviously, there is no question that recording engineers will want to be able to take digital tapes from one studio to another, just as they do now with analog masters. Yet, as we begin this digital decade, a single DTR standard is still somewhere off in the future, and compatibility is nowhere in sight.

Of course, life would be a lot simpler if everyone could agree on standards right away. However, premature standardization carries with it the risk of locking the industry into a system that quickly passes into obsolescence, as the technology advances. So, despite the short-term incompatibility problems, it's probably to our long-term advantage to endure a "shake down" period, during which we learn a little something about what everyone has to offer. Then, based on practical experience, we shall eventually be better able to make decisions about DTR/VCR compatibility, sampling frequency (frequencies?), and encoding formats.

One need look no further than the Sony catalog for a "crash course" on the various systems now competing for our attention. As one of the largest producers of digital audio equipment, the company's Digital Audio Division currently manufactures systems in a variety of formats. Here, we will quickly put together a tape duplicating system, to see what it takes to go from one format to another.

MULTI-TRACK OPEN-REEL FORMAT

We'll begin with a 24-track master tape, recorded on a fixed-head, 30 in/sec. open-reel recorder, using half-inch video tape. The model PCM-3324 employs a 16-bit linear quantization format, and requires one digital track for each audio channel. In other words, the infinitely-variable analog input level is reduced to a finite series of discrete levels; 65,536 (= 2^{16}) of them. The machine's sampling frequency determines the rate at which these quantized levels are, well, "sampled." The PCM-3324 has seven switchable sampling frequencies (32, 44.056, 44.1, 48, 50,



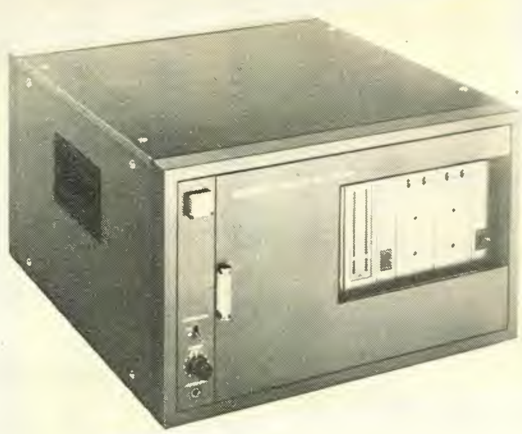
The PCM-3324 offers 24 tracks on half-inch open-reel video tape. Sampling frequency is switchable, between 32 and 50.7 kHz.

50.35 and 50.7 kHz). Let's assume the sampling frequency is set at 50.35 kHz, which is the same frequency used by the Mitsubishi Digital Audio System described elsewhere in this month's **db.**

FOUR-TRACK OPEN-REEL FORMAT

The PCM-3204 is a 15 in/sec. four-track open-reel recorder which also uses a 16-bit linear quantization format and a switchable sampling frequency. Since the tape speed is half that of the multi-track system, the PCM-3204 requires two digital tracks per audio channel.

For digital remotes, the PCM-3204 uses a four-track, quarter-inch format. At 15 in/sec., the system requres two digital tracks per audio channel.

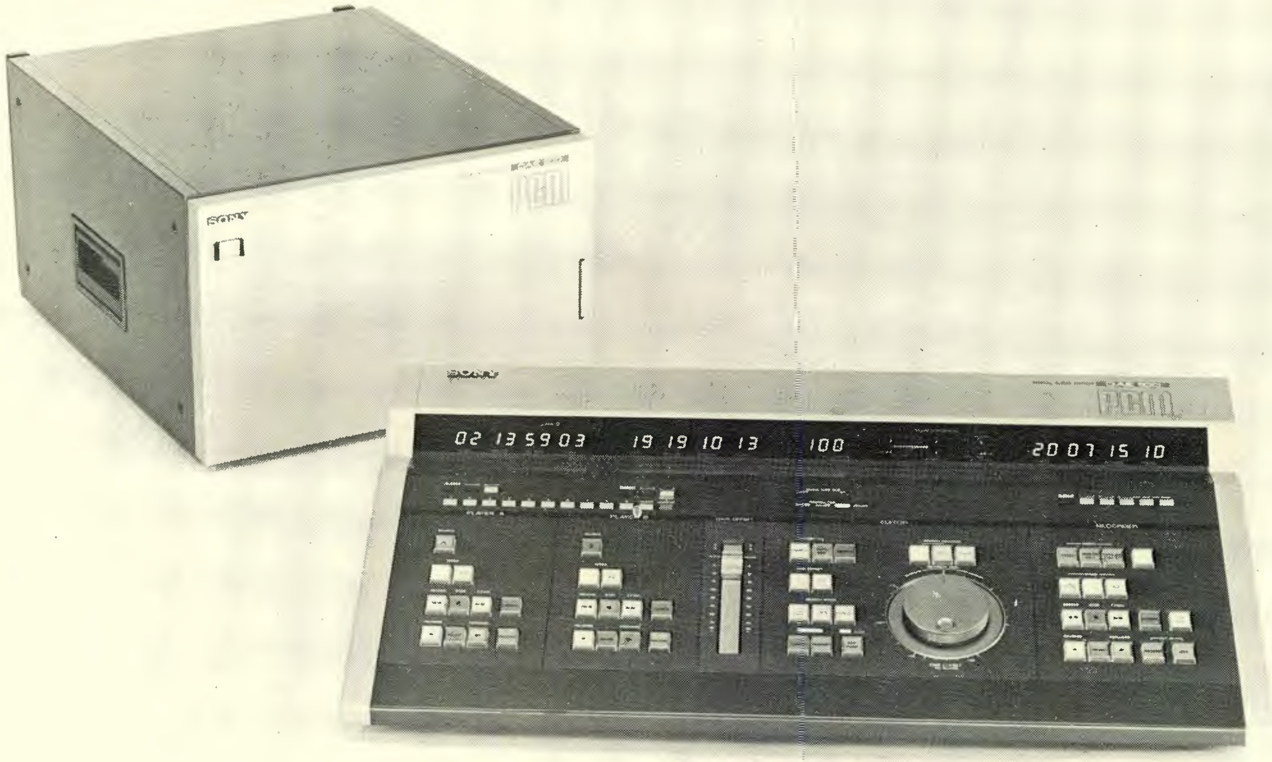


The PCM-1600 Digital Audio Processor is used to record digital audio onto a U-matic video cassette recorder.

VIDEOCASSETTE FORMAT

Professional studios have become so accustomed to using open-reel tape recorders that the thought to going to a cassette format may take a little getting-used-to. However, as electronic editing becomes more popular, there may be less and less point to insisting on open-reel. For the moment though, digital multi-trackers are all open-reel, while cassettes—as well as open-reel decks—are being used for stereo masters.

The PCM-1600 is a two-channel PCM processor, which allows digital audio programs to be recorded on a U-matic videocassette recorder. Once again, a 16-bit linear quantization format is used. However, the sampling frequency is 44.056 kHz, chosen for the sake of compatibility with the video horizontal scanning rate.



The DAE-1100 Digital Editor. Note the large edit-search dial on the front panel. The system memory stores six seconds of program material, which may then be "rocked" back-and-forth, as in analog tape-cut editing. Once the edit point is found, the digital editor makes the "splice" electronically.

DUBBING, DTR-TO-VCR

Assuming we wanted to make a digital-to-digital transfer between the PCM-1600 and one of the PCM-3300 series open-reel machines (or vice-versa), we would find this impossible to do, if the sampling rates did not correspond. And so, we would need a DSX-87 Sampling Rate Converter. This permits real-time transfer between any two machines whose sampling-rate ratios are 8:7 or 7:8 (44.056:50.35 = 7:8).

VCR-to-VCR

The PCM-100 is a lower-priced digital audio processor, similar in function to the PCM-1600, but using a 14-bit quantization format, which conforms to the digital recording standards adopted by the Electronic Industries Association of Japan (EIAJ).

To make transfers from a 16-bit to a 14-bit format, the DQP-6040 Digital Quantization Processor should be used. This allows the higher performance 16-bit tapes to be transferred in real time, without an intermediate D/A, A/D process. The system also permits 14-to-16-bit conversions.

EDITING

The DAE-1100 is a second-generation digital audio editor, designed for use with the PCM-1600 processor. When a tape is stopped in the vicinity of an edit point, six seconds of program are automatically stored within the system's memory. A large search dial on the front panel may then be slowly rotated, as though it was the capstan on an analog transport. The operator hears the program, just as if an analog tape was being slowly "rocked" back-and-forth across the head. However, the digital tape is not actually in motion; the operator is merely listening to a "playback" from memory. Once the precise edit point is established, the edit may be previewed and then executed, with an accuracy of 362 microseconds.

SUMMARY

Summarizing the formats described above, we find;

model	tape	Tracks audio	Tracks digital	tape speed	quanti-zation	sampling frequency, in kHz
PCM-3324	½"	24	24	30	16-bit	32-50.7†
PCM-3204	¼"	4	8	15	16-bit	32-50.7†
PCM-1600	*	2	1 (TV)	*	16-bit	44.056
PCM-100	*	2	1 (TV)	*	14-bit	44.056

*PCM-1600 and PCM-100 require an external U-matic VCR.
†Switchable.

Which format will become the standard? At this point, it's still too early to say. But hopefully by the end of the decade the question will have been long-since answered in the pro audio marketplace. ■

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People/Places/Happenings

• Industry veteran **John J. Bubbers** has been appointed president of **Dynaco, Inc.,** Canton, MA. Mr. Bubbers comes to Dynaco from his post as president of **Celestion Industries,** Holliston, MA. Mr. Bubbers also has served as president of BSR-owned **Audio Dynamics Corp.,** as well as vice president of field engineering for **Stanton Magnetics** and vice president and professional products manager of **Pickering and Company.**

• **PCI Recording,** Rochester, NY, has announced the opening of its newly renovated 24-track facility. The studio includes a Neotek 32 x 32 Series III Console, an MCI JH-114-24 recorder and an MCI JH-110-2-14 mastering deck. PCI Recording is located at 907 Culver Rd., Rochester, NY 14609.

• Named professional products field sales manager for **Bose Corporation,** Framingham, MA, **Jeffrey J. Pallin** will assume the responsibility for the U.S. sales of Bose concert and public address sound systems. A graduate of Boston University, Mr. Pallin joined Bose in 1977 as midwestern field sales representative.

• Utilizing satellite technology, the **RKO Radio Network** now feeds live satellite transmissions to the twelve RKO Radio owned and operated stations, located in eight major U.S. markets—via **Western Union's Westar Satellite.** The live feeds consist of three-minute newscasts and 90 second lifesound features. In addition, the RKO Radio Network will provide exclusive stereo-produced music specials to all affiliates.

• **Burns Audiotronics,** Hicksville, NY, exclusive U.S. distributor for **Beyer** products, has recently appointed **Bob Lowig** head of the audio transformer sales department.

• Formerly field engineering manager for **BASF Systems,** Bedford, MA, **Anthony Saratora** has been promoted to manager, quality assurance. In this position, Mr. Saratora will oversee field engineering; systems and planning; and quality control operations. Since joining the company in 1968, Mr. Saratora has held varied positions within the quality assurance department.

• **Panasonic Company,** Secaucus, NJ has announced the appointment of three new Technics assistant national sales managers: **Richard Del Guidice,** electronics and speakers; **Paul Foschino,** tape recorders; and **Kenneth R. Wipfler,** turntables. In addition, **Sid Silver** has been promoted to the position of public relations and show manager, Technics Department. Mr. Silver joined Panasonic in 1972 as sales promotion coordinator, and most recently served as merchandising coordinator.

• **Jeffrey Marks** has been named advertising manager of the Magnetic Tape Division of **Sony Industries,** New York. Mr. Marks' responsibilities will include advertising, sales promotion, collateral material and product publicity. Joining Sony a year ago, Mr. Marks previously held the position of advertising manager for Sony Industries' Business Products Division.

• Undertaking a major expansion and reorganization of its international marketing program, **McMartin Industries, Inc.,** Omaha, Nebraska, has recently named **Thomas S. Butler** director of international sales, after serving three years as Eastern U.S. sales manager. In addition, assisting Mr. Butler will be **Ernest Credgington,** consultant on international projects and **Ron Briggs,** international contract and sales administrator.

• **Howard P. Ladd,** chairman of the Audio Division of the **Electronic Industries Association/Consumer Electronics Group,** and **Jerry Kalov,** chairman of the **Institute of High Fidelity,** jointly announced that negotiations to merge the IHF into the EIA/CEG have been successfully concluded. The merged organization of the IHF will become an operating subdivision of the Audio Division of the Consumer Electronics Group.

• **Otari Corporation** has moved to a new, expanded facility which contains 50 per cent more office space and 100 per cent more lab space. The new address is: 1559 Industrial Road, San Carlos, California 94070. Telephone: (415) 592-8311.

• **Clyde Moore,** formerly vice president of marketing for **Crown International, Inc.,** Elkhart, Ind., has been appointed to the newly-created post of vice president/planning at Crown.

• The **Fidelipac Division** of **Harvel Industries Corporation** has begun operation as a totally independent organization, under established management. Retaining their current positions are: **Daniel McCloskey,** general manager; **Arthur Constantine,** sales manager; and **Robert J. Gosciak** and **Frank A. DiLeo,** development engineers. Now known as **Fidelipac Corporation,** the company is headquartered at 109 Gaither Drive, Mount Laurel, New Jersey 08057.

• Former Chief of the Broadcast Bureau of the **Federal Communications Commission, Wallace E. Johnson** has become executive director of the **Association For Broadcast Engineering Standards, Inc.,** (ABES), Washington, D.C. In his new position at ABES, Mr. Johnson will assume primary responsibility for directing the Association's response to aural broadcast allocations and technical standards. Mr. Johnson retired from the FCC after 37 years of employment with that agency.

• Two of the co-founders of **Switchcraft, Inc.,** Chicago, **Wilfred L. Larson,** Chairman, and **Fred O. Dumke,** Secretary-Treasurer, have announced their retirement from the company. Both Mr. Larson and Mr. Dumke will continue indefinitely as consultants.

• **Belden Corporation** has appointed **Ronald L. Stier** to the newly created position of long-range marketing manager for its Electronic Division, based in Richmond, Ind. Joining Belden in 1964, Mr. Stier has been marketing manager of the Electronic Division since 1974.

• **Aspy Tantra** has been promoted to the position of quality assurance manager for **Shure Brothers Inc.,** Evanston, Ill. Mr. Tantra joined Shure in 1977, and most recently was manager, central automation and tool engineering.

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