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THE SOUND ENGINEERING MAGAZINE

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NOVEMBER 1979
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Coming Next Month

• In December, we take another look at microphones. Although new design concepts may not come along at the same pace as in console design, still, there is always one more applications concept for us to try out. New ideas are developed, and old ones are rediscovered, and you can keep up with it all by reading the December issue of **db—The Sound Engineering Magazine.**

About The Cover

• Spanning several generations, a family portrait of recorded discs. In the foreground, the new Philips Compact Digital Disc is shown in perspective with an antique gramophone.

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THE SOUND ENGINEERING MAGAZINE

NOVEMBER 1979 VOLUME 13, NUMBER 11

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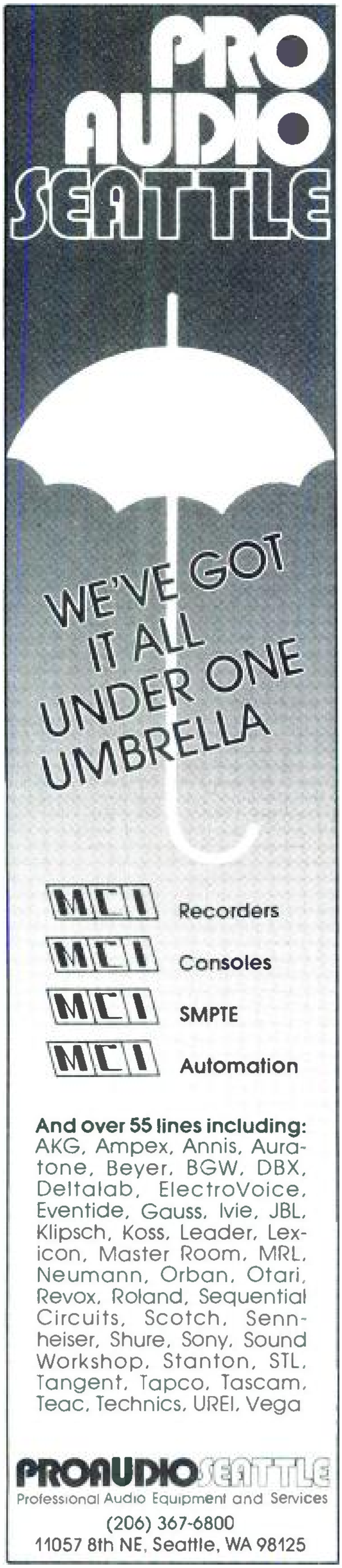

Circle 35 on Reader Service Card

db Letters

TO THE EDITOR:

I would appreciate your sending any information which you have concerning reputable programs in Acoustical Engineering, Audio Engineering, or other similar disciplines. This information would be of much benefit in counseling students who wish to transfer to such programs.

TO THE EDITOR:

I was wondering if you have available a list of recording institutes or schools that you could send me? I'm currently employed as a program director and announcer, but am interested in becoming a recording engineer, or something related.

Your help would be greatly appreciated.

TO THE EDITOR:

Could you please send me any information that you have concerning sound engineering as a career. My daughter is interested in schools where courses are offered preparing one for such a career.

Any information that you can give me would be greatly appreciated.

db Replies:

The preceding letters represent a small sampling of typical letters we receive, from time to time, asking for information on careers in audio, and audio engineering schools. For more on the subject, see this month's Special Report on Education and Audio.

TO THE EDITOR:

It is perhaps understandable and forgivable when A. Stewart Hegeman says, as he recently did, that modulation once occurred can't be undone. I had expected, however, that an audio expert such as Mr. Crowhurst wouldn't make the same error: db (July 1979)—"In fact, however distortion gets in, you cannot take it out again."

Yes, Virginia, there is a Santa Claus undistorter! Essentially, if one has non-linear distortion arising from any number of non-linear processes (or elements) one can cancel it completely (under a wide range of conditions) by passing the distorted signal through processes having just the "opposite" type of distortion. This matter is discussed in detail in I.R.E. Trans. on Audio AU-7, 128-133, Sept.-Oct., 1959; AU-8, 104-105, May-June, 1960; and AU-9, 103-105, July-August,

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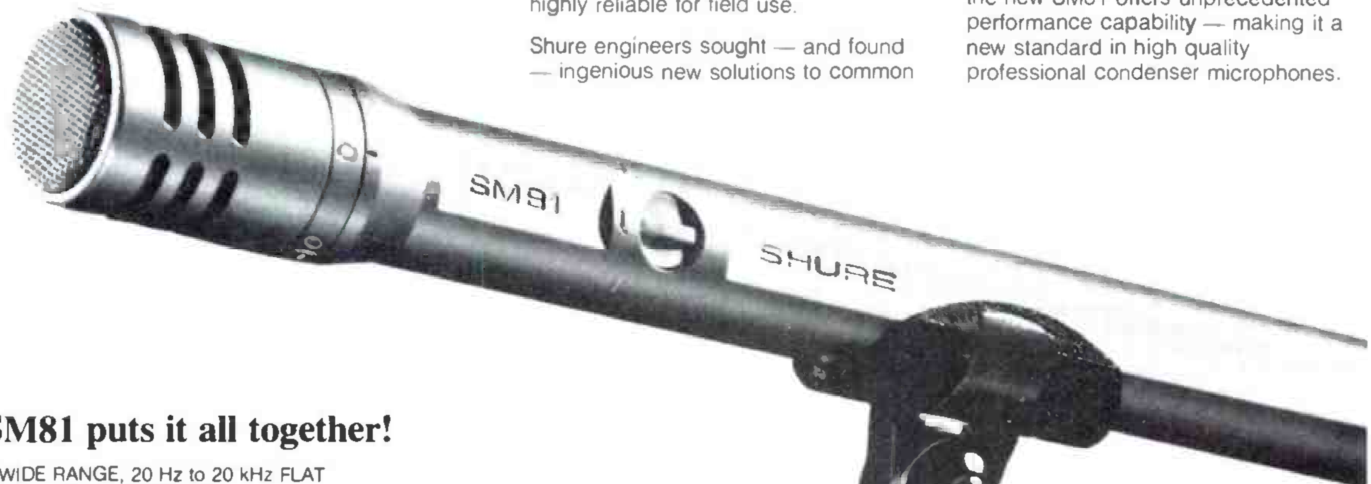

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1961. Perhaps Mr. Crowhurst missed these at the time. Paul Klipsch didn't and fell into the same trap now also occupied by Mr. Hegeman and Mr. Crowhurst. His unbelief was rectified (single-wave?) in the AU-8 article above.

I'd be interested in Mr. Crowhurst's response to these matters.

J. Ross Macdonald
(William R. Kenan, Jr.
Professor of Physics)
University of North Carolina
at Chapel Hill

Mr. Crowhurst replies:

This letter repeats a misconception that has come up before. Of course, a lot of processes are possible, particularly with today's technology. But no electronic equipment has the capability of knowing what a "signal" was supposed to be—only what it is. Any simple feedback circuit can *reduce* distortion, to any desired degree, *provided* it has the undistorted input signal to use as a reference. But that was not the situation my column was talking about.

When a signal comes into a processor, of whatever kind, that is all the processor has to work with, unless there is some extraneous "information" from some other source. All the processor "knows" is what it is told. To do what Dr. Macdonald suggests, the processor must be told what kind of distortion is present, that needs correcting, otherwise it will also remove part of the signal, when it does not happen to be distortion.

That is without getting into the mathematics of how it does whatever it does. For example, to eliminate 2nd harmonic that we know is there, the processor can produce a 2nd harmonic of the input signal, of equal magnitude and opposite phase. Suppose it has 10 per cent second: a little math shows that such a process can be made to cancel the 2nd harmonic already there (regardless of whether it is supposed to be or not) at the same time producing 1 per cent of 4th harmonic, that was not there before.

A more sophisticated processor approach could employ recognition analysis. For example, a certain combination of harmonics might be recognized as a sound from a wooden clarinet—or as closely approximating it. And if it departed from the programmed analysis, "distortion" could be removed to make it conform to the programmed content. But suppose it was a silver clarinet, what then?

Another basis uses waveform, rather than frequency content. Computer technology uses this all the time, to "shape up" pulses that lose their shape. But such a system is programmed according to the shapes it is designed to handle. In simplest terms, whatever form of analysis is used, waveform or frequency content, the composition can be "standardized:" the wave can either have a standard shape, or a standard frequency spectrum.

But this kind of correction will make all sounds alike, using whichever basis, regardless of whether they are supposed to be alike or not. There are other possibilities, such as programming the electronics to detect non-musical "signals," and to use the kind of subtractive signal Dr. Macdonald suggests to remove what the system identifies as "unwanted" components.

It all comes back to basics. Sure, a black box can be designed to do anything—but think about what it's doing. It just does as it's told.

To The Editor:

Your article on "Nuts 'N' Bolts" and a subsequent letter in the February 1979 issue of **db** pointed out that there is more confusion about connectors than meets the eye.

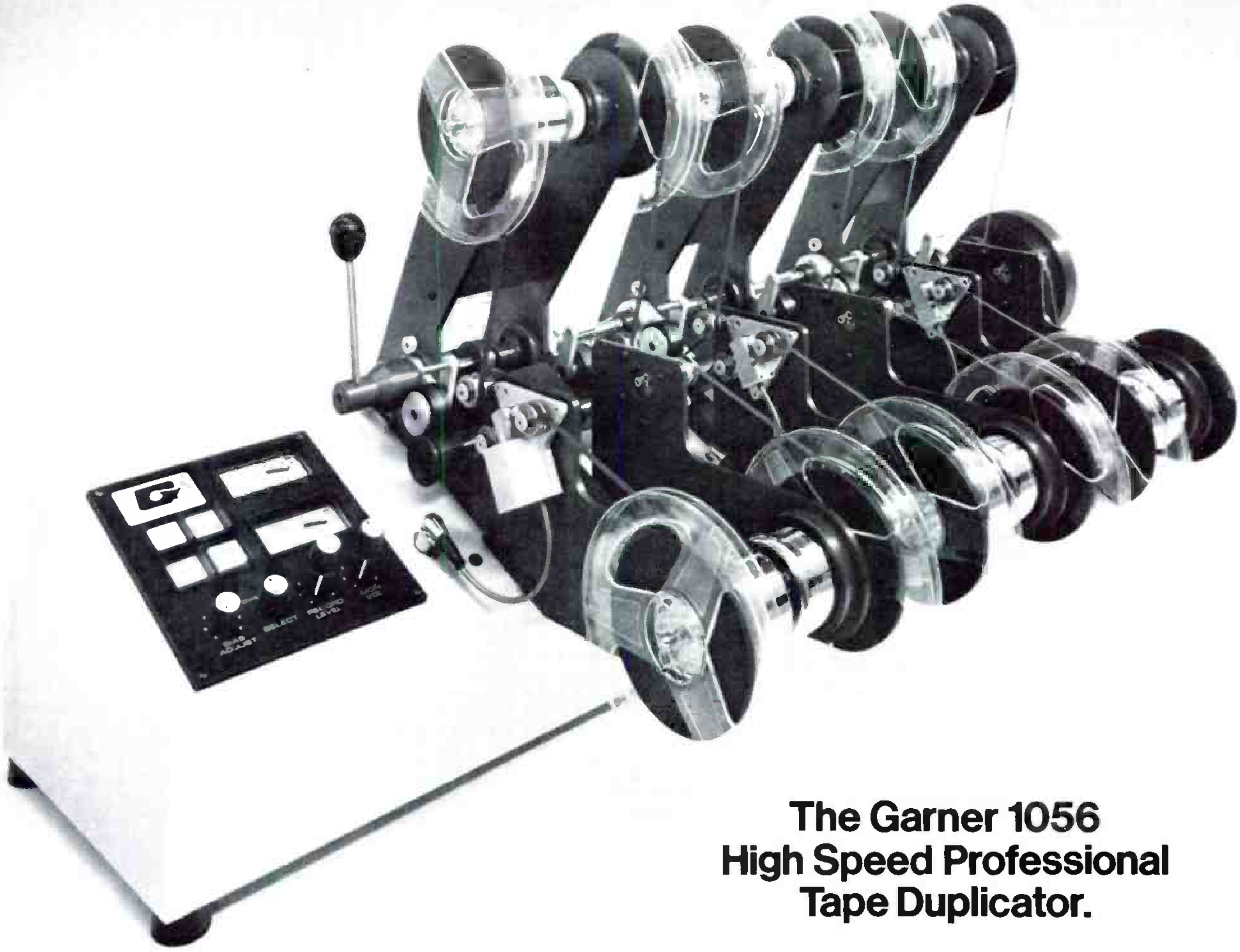
The E.I.A. standard RS-297-A describes *not* the "XLR" connector, but what Cannon calls the "UA" connector. This well designed but expensive and short lived (remember the EV 666?) connector is not the same as the "XLR" at all and perhaps you should let your readers know this.

I am in the process of researching standards, whether published or just "in house," for connector wiring (which pin is what) and microphone phase (or polarity). It seems so far that about 45 per cent of the world wires their connectors one way; another 45 per cent wires their connectors out of phase with the first group, and the remainder do something totally different!

Any help from you and/or your readers would be appreciated.

William F. Ruck, Jr.
Broadcast Engineer
San Francisco, CA

Dub a dub, dub a dub, dub.



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db Calendar

NOVEMBER

2-5 **N.Y. AES Convention.** Waldorf-Astoria, N.Y.C.

Synergetic Audio Concepts Sound Engineering Seminars:

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14-16 Orlando

For information on these three-day seminars contact: Synergetic Audio Concepts, P.O. Box 1134, Tustin, CA 92680 (714) 838-2288.

15- **Billboard's First International Video Music Conference.** Sheraton-Universal Hotel, Los Angeles, CA. More information is available from the Billboard International Video Music Conference, 9000 Sunset Blvd., Los Angeles, CA 90069. (213) 273-7040.

DECEMBER

5-7 **Synergetic Audio Concepts Sound Engineering Seminar.** San Antonio, Texas. For more information contact: Synergetic Audio Concepts, P.O. Box 1134, Tustin, CA 92680. (714) 838-2288.

11-14 **International Entertainment Exposition.** Las Vegas Convention Center, Las Vegas, Nevada. For more information contact: American Expositions, Inc., One Lincoln Plaza, New York, NY 10023. (212) 691-5454.

JANUARY

4-9 **1980 CES (Consumer Electronics Show).** Las Vegas, Nevada.

FEBRUARY

25-28 **65th AES Convention (London).** London Hilton and Park Lane Hotels.

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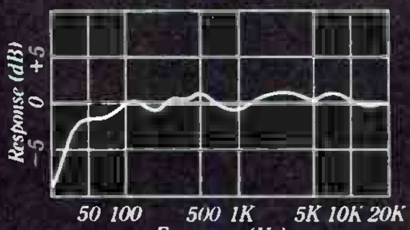

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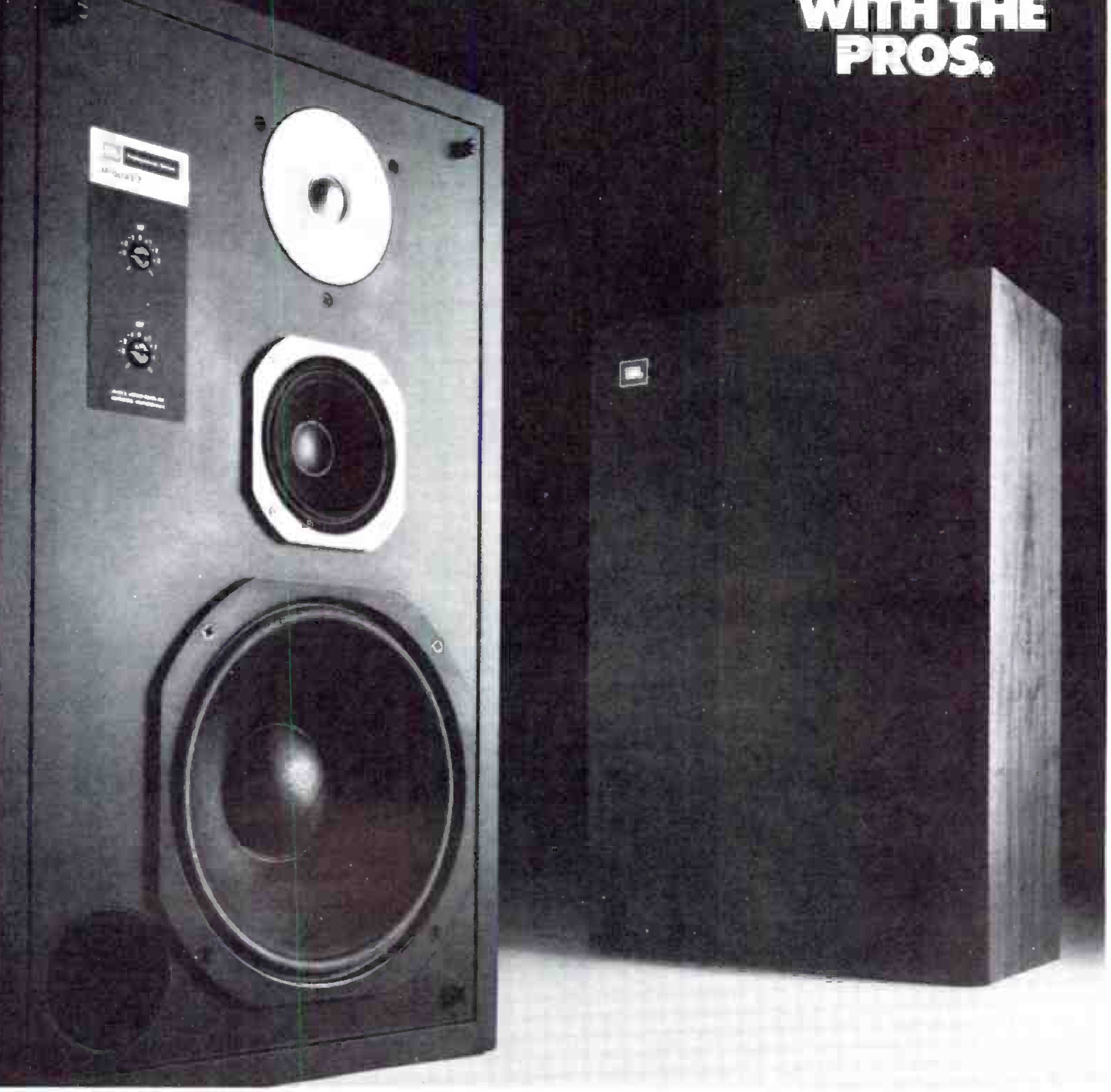

Circle 17 on Reader Service Card

Home-built Audio Pads

• Control of signal levels throughout the station's audio system is important to audio quality, as well as operational convenience. All the major units of the system will have variable on-board gain controls for operational purposes, and the system will also contain a number of fixed pads for isolation, matching, and general fixed level control. In some of these locations conditions are more critical and require a precision pad. And there will be many locations or situations in which control is far less critical so that a home-built pad will do the job adequately. This month we will discuss some aspects of building resistor pads for these non-critical areas of audio control in the system.

BASIC CONCERNS

The most basic reason we need a fixed-loss pad in a particular location in the system is that the normal signal levels are too high for the amplifier or other unit which follows that point. When these normal levels are too high this can create either of two problems: the first is overload of the following amplifier and attendant poor audio quality and distortions; the second is operational inconvenience in that the on-board variable gain control of the load amplifier may be just barely cracked open. Since resistor pads are always loss devices, they can be used to advantage in these high level situations. By inserting a fixed-loss pad between the high level source and its load, the signal is brought into an area where the load amplifier can operate more comfortably in its mid-range linear condition, and the on-board gain control can be placed in its mid-range position for more operational range to handle temporary level changes.

When the pad is inserted in the signal path, it must reduce only the signal amplitude, and it must not disturb the normal circuit impedance or cause an impedance mis-match. An impedance mis-match can create a poor response curve across the audio band-pass. To

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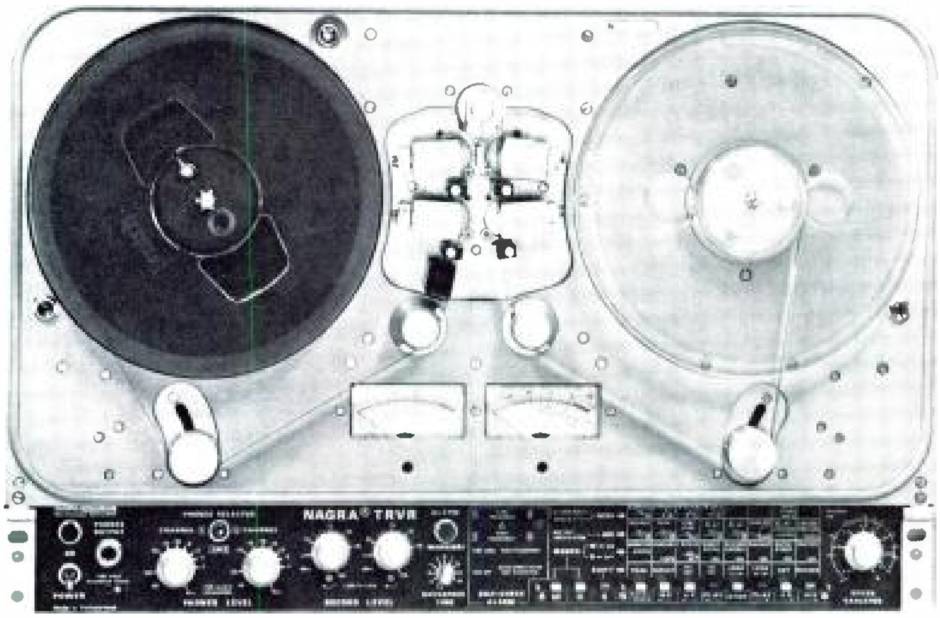

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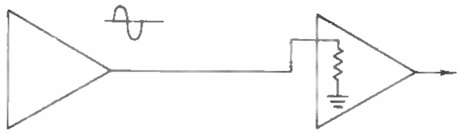
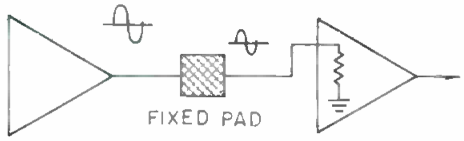
avoid this, the pad must be designed to match the impedances of the particular circuit. Besides matching the impedances of a given circuit, the pad can be designed as an impedance matching device between two dissimilar circuit impedances. The impedance requirement, as well as the loss factor, gives the pad more versatility. That is, a pad may be designed to provide a fixed-loss between two like impedances, or it can provide a fixed-loss-plus-impedance transformation, or it can be designed primarily as an impedance transformation device with minimum loss.



(A) Without Pad



(B) With Pad

Figure 1.
(A) When the signal level is too high, the amplifier can be easily overloaded, and its gain control must run barely open.
(B) A pad reduces the level, so that the gain control and amplifier can operate in midrange.

BASIC INGREDIENTS

Since a pad will be designed to fit a given situation, certain basic facts about that situation must be known beforehand. The first important factor is the impedance of the source and that of the load device in the path the pad will be inserted. These two impedances become the input and output impedances of the pad respectively, and they effect the value of the resistors used in the pad. A pad of a given loss-value will require different value resistors when it is inserted between two like impedances than one inserted between two different impedances.

The normal signal amplitude from the source is the next important factor. The design will work from this, and will naturally effect the next important factor of required loss. The loss required in most cases will be somewhat approximate since what we may want to do is move the load unit's operation out into mid-range. In this situation a couple of dBs one way or the other is really immaterial. Should the situation call for an exact loss, then a precision pad is needed. You may be able to hit it with a home-built pad, but the odds are against it.

Yet another important factor is whether the circuit at that point is balanced or unbalanced. Design the pad to suit the circuit conditions and there will be less chance for problems. Mixture of balanced and unbalanced will work sometimes, but problems may develop either at the outset or at some later time.

For the majority of cases, power will be of little concern since most of the signal paths will be very low in power. But, when designing a pad for a high-level audio circuit and speaker runs, considerably more power is present in the signal. The resistors in the pad for these applications must be of sufficient power rating to handle that audio power or they will soon burn-up.

Some situations may dictate a particular style of pad to use, but in most cases the style is one of choice. There are many styles and types of pads which can be used. Text books and similar reference manuals will provide information on many styles of pads and the formulas to compute the resistor values. For the most

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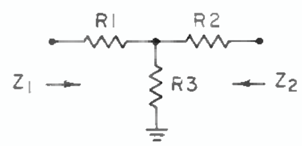
common type pads, broadcast equipment catalogs often have already computed the resistor values for various losses and have these listed in chart form.

Whether you compute the resistor values or select them from a chart, there is one thing to keep in mind—the pad will not be a precision pad. You will use standard stock value resistors, and these seldom come out to the exact values that were calculated for the pad you need. Because of this, the pad will be somewhat less than precise, yet entirely satisfactory for many situations.

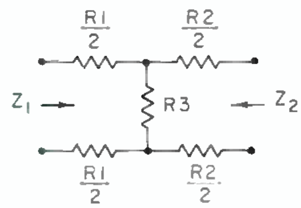
MOST COMMON PADS

The two most common pads you will use for these less than precision cases, are the "T" and the "H" pads. These pads get their names from the appearance of the schematic diagram which appears as a letter T, or an H on its side. These two are essentially the same pad, however, the "T" is for unbalanced circuits and the "H" for balanced circuits. Since either pad contains the same *total* resistance value, it is necessary to compute the values only for the "T" pad. That is the way the formula is designed. In the situation where an "H" pad is needed, simply divide the R1 value into two equal parts and place one on each side of the

Figure 2. The "T" and the "H" pads. The "H" is the same as the "T", but arranged for balanced circuits.



"T" Pad



"H" Pad

Formulas:

$$R_3 = \frac{2\sqrt{N Z_1 Z_2}}{N - 1}$$

$$R_1 = Z_1 \frac{(N + 1)}{(N - 1)} - R_3$$

$$R_2 = Z_2 \frac{(N + 1)}{(N - 1)} - R_3$$

Z_1 = Source Impedance
Z_2 = Load Impedance
N = Loss in dB

pad input to balance the circuit. Do the same with the R2 value on the output side of the pad. R3 (shunt value) remains the same in either case. Either of these pads can be used to provide a fixed-loss between two similar impedances, or as a fixed-loss-plus-impedance transformation between two dissimiiar impedances.

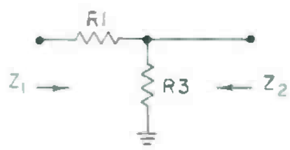
When impedance transformation (matching) is all that is desired, then use the "L" or the "U" pad. These provide the impedance transformation with minimum loss. The lesser loss results from the fact that fewer resistors are used. Again, the "L" pad is for unbalanced circuits, and the "U" pad is for balanced circuits. Calculate for the "L" pad and then divide the series arm into two equal values for the "U" pad.

OTHERS

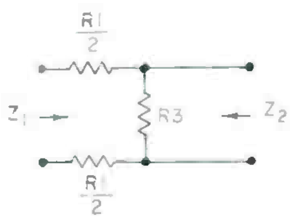
Besides the common pads discussed, a variety of other configurations can be used for situations of bridging, matching, and other loss uses. Reference books will show many of these, as well as the necessary formulas. Once you have found one or two styles that work well for you, stick with them.

One such pad I came across many years ago and have used extensively is for

Figure 3. When matching-only is desired, use the "L" or the "U" pad. The "U" pad is for balanced circuits. There will be loss, but less than that of the "T" or the "H" pad.



"L" Pad



"U" Pad

Formulas:

$$R_1 = Z_1 \sqrt{1 - \frac{Z_2}{Z_1}}$$

$$R_3 = \frac{Z_2}{\sqrt{1 - \frac{Z_2}{Z_1}}}$$

Z_1 = Source Impedance
Z_2 = Load Impedance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IMAGINATION!

situations requiring the division of a signal source into two equal loads, or the mixing of two equal sources into a common load. For example, the two outputs of a multi-tray cartridge tape player into the single console fader, or combining Left and Right stereo audio to form a mono signal. The pad can be worked either way—one into two, or two into one. The pad is made up of (3) 600 ohm resistors and is for 600 ohm circuits. There is a 6 dB loss across the pad.

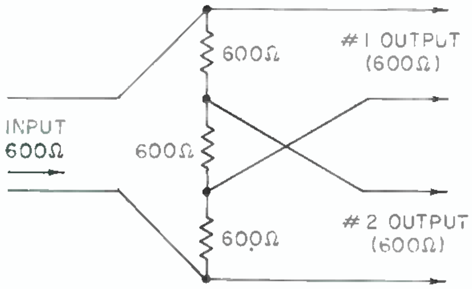


Figure 4. A simple pad for mixing or dividing the signal in 600 ohm circuits. This pad can be worked in either direction.

BUILDING THE PAD

Once the required loss, and impedance configuration have been determined, and the resistor values computed, you are ready to build the pad. The first thing you will discover is that there may not be many stock value resistors that equal those computed values. Select resistors from the stock values which are the closest to the computed values. The closer you come in this regard, the closer the pad will produce the predicted results. Although the values will not be exact, they will generally be close enough so that the pad will produce generally-acceptable results. Should the stock values you have be far from the computed values, you can add series resistors to bring the individual values near the computed value. In any case, when building an "H" pad or other balanced pad, use an ohmmeter and actually measure the individual resistors that will be used on each side of the circuit, selecting those values that will maintain the circuit balance.

However you construct the pad, physically, may depend upon its physical location in the system, as well as your own ingenuity. The important thing is to identify which are the input terminals and which are the output so that it can be connected into the circuit properly. The second important thing is to insulate the internal connections of the resistors in the pad so that they do not accidentally short to a cable shield or the chassis.

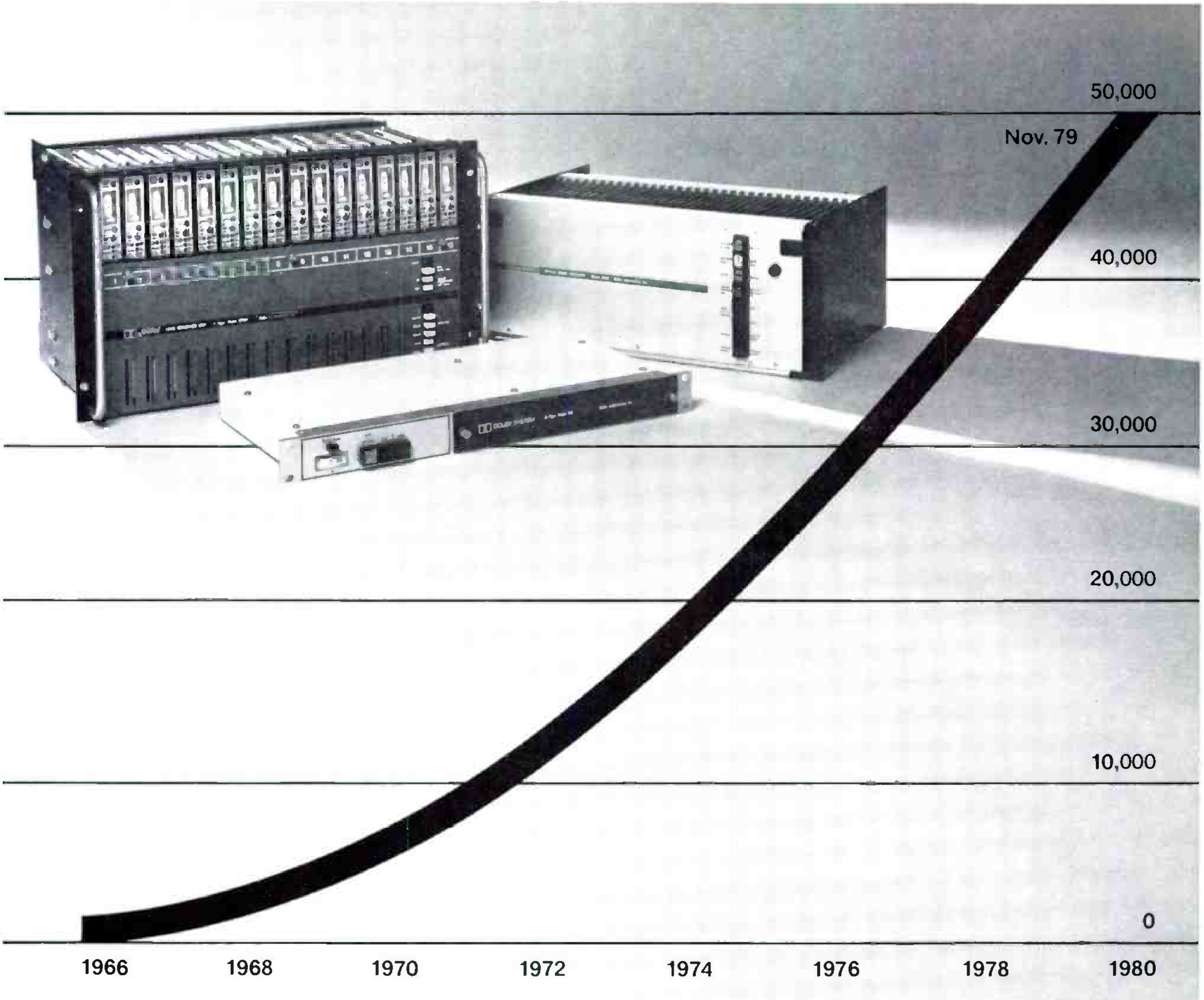
MEASUREMENT

When all is said and done, the performance of the pad in the circuit will generally show if the effort was successful or not. Since, in most applications, a difference of a few dB is immaterial to what is intended, the pad should produce acceptable results.

But if you desire to know what actual loss the pad will produce before you install it in the circuit—measure it. Set up the audio signal generator and the audio voltmeter. Set the generator output impedance to the value that will be found in the actual circuit, and add a resistor load across the output of the pad which is the same as will be found in the circuit. Feed some arbitrary (but known) value of sine wave into the pad, and measure the level of that sine wave at the output of the pad. The difference between the input signal level and the measured output signal level is the loss the pad is producing.

RECAP

Audio quality and operational convenience require signal level control throughout the system. The control is achieved by on-board variable gain controls, some precision fixed-pads, and many non-precision pads. By a little effort in calculation and work in assembling pads, the station engineer can construct inexpensive pads for non-critical areas that serve a very practical end in terms of audio control and operational convenience. ■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NORMAN H. CROWHURST

db Theory & Practice

Systems Design In Audio

• Recent affairs in my life have stressed the importance of the systems design concept in many contexts, not limited to engineering. And that, in turn got me thinking about how systems design—the same concepts—have influenced the growth of audio to its present stage of development.

When I was a boy in school, there were two audio systems: radio (a.m.) and phono (laterally-cut mono). But then engineers didn't even think about radio being a.m., because f.m. had not been introduced: and while the earlier phonographs had used vertical cut, lateral cut was generally conceded to be an improvement—much less distortion—so vertical cut was no longer thought of as a viable alternative.

But we did have equalization, and later there was vari-pitch recording, to get more onto a record, and then there were record changers, as an alternative to manual turntables. Perhaps the introduction of standardized equalization was the first step toward systems. Before that, recordings, or radio broadcasts did not pay much attention to frequency response.

CONSTANT AMPLITUDE & CONSTANT VELOCITY

Awareness of constant amplitude, or constant velocity, as applied to recording, and its equivalent in radio, led to finding an equalization characteristic, or curve, that would be a successful

compromise between the two. Constant amplitude means that the same sound pressure will produce the same amplitude of stylus movement, as frequency is changed. Constant velocity means that the same velocity of stylus movement will correspond with the same sound pressure at different frequencies.

These terms of reference did not come into existence before electrical transcription was invented. They would have little meaning, applied to the old acoustical phonograph. They might have a corresponding meaning applied to radio, but the relationship would be to diaphragm movement at the microphone or loudspeaker (then they used headphones rather than loudspeakers), so the whole concept of such a relationship was quite new.

TRANSDUCER TYPES

However, there are correspondences with transducer types. A dynamic type transducer, which uses an electromagnetic element such as a moving coil, produces a response which is inherently constant velocity, because the voltage generated by the pickup or microphone, or required to drive the cutter-head or loudspeaker, is proportional to the velocity of movement, whatever the frequency.

Static transducers, such as piezoelectric, or capacitor (then called "condenser") produce an inherently constant amplitude relationship: voltage is proportional to amplitude of movement, regardless of frequency. These prove to be oversimplifications in the real world, but they are the basic relationships on which each kind of transducer works.

Velocity and amplitude are related by frequency. Given constant amplitude, velocity is proportional to frequency: because moving over the same amplitude at a higher frequency increases the velocity needed to maintain that amplitude. And given constant velocity, amplitude must be inversely proportional to frequency, because the longer periods corresponding with lower frequencies require greater movement to maintain the same velocity.

From that relationship, expressed in its basic terms, it would seem that static transducers are the best choice. Use of dynamic transducers, with the naturally related constant velocity, requires quite large amplitudes for the lower frequencies. But that is a theoretical choice that does not actually "follow through" as well as might be expected. Control forces and electrical circuit values inevitably make devices that are basically of the "static" type, perform like constant velocity, below some critical frequency, generally known as "cut-off."

SYSTEM DESIGN

So, we begin to introduce the various parameters that make it necessary for us to think in terms of system design. How much program can we squeeze into a given recording space, measured in those days on disc? When magnetic recording came along later, the same question arose, in only a slightly different context. And on radio, how much information can be squeezed into a given radio band, or spectrum, by using different channels, or station-frequencies?

Earlier columns in this series have addressed dynamic range- and noise-versus-distortion. These are the factors that decided on the best choice of equalization characteristic: how could you modify response during recording and playback, so the best dynamic range could be squeezed into the minimum space? A systems design question, the first of many to follow it, in greater profusion.

The space question is tied to the level of recording, as well as to its frequency content. Lower frequencies require greater amplitudes, if one spiral of the groove is not to break through into the

The logic behind the Revox B77.

The logic is the logic which is built-in.

It's an ingenious and highly sophisticated system—much like the human nervous system—which controls the deck's functions.

You can push any button in any order with no chance of damaging your tapes. Our motion sensing system constantly feeds status reports to the logic circuitry which activates your commands in proper sequence.

The logic also permits full-function remote control, and an editing mode that keeps the playback circuitry live, even when the motors are stopped. You can make your splices right on-the-beat, and our built-in splicing block makes it easy.

The design and construction of the Revox B77 further guarantee smooth and accurate operation. To get the long-life advantage of ferrite without static build-up or heat degradation, we use Revox's exclusive Revodur heads, made of metal to dispel heat and static, and vacuum-coated with permalloy for durability.

The B77 has a unique capstan motor that's monitored by a tacho head to precisely control speed and limit wow and flutter to professional studio standards.

Revox offers many options with the B77 including a full range of speed configurations from 15/16 IPS to 15 IPS, variable speed control, ¼ track record/playback and more.

All this professional quality is neatly engineered to fit in a deck you can carry. After all, if you own a machine this good, it's logical to take it with you.

Experience the B77 and the full line of Revox and Studer professional products at your franchised dealer today.

STUDER REVOX America, Inc.
1819 Broadway, Nashville, TN 37203 615 329-9576/ In Canada: Studer Revox Canada, Ltd.

Circle 31 on Reader Service Card

next, and high levels also require greater amplitudes of stylus movement. But program varies in dynamic level, so that in quiet passages a lot of grooves can be squeezed in, while on loud passages each groove must be given more space.

VARI-PITCH

Thus the idea of vari-pitch was born. At First, this was achieved manually, by the record cutter changing the pitch, a little ahead of loud passages, to give more room, and closing them up again on quiet passages, to get more grooves onto the record. This depended on the skill of the operator who cut the records, and on his having an adequate score for cueing him when the loud and quiet passages were coming.

Systems design gets this kind of thing automated. How can it do that? Use of tape for mastering facilitates this. By running the tape past a "monitoring" head, before it gets to the playback head used to provide input for recording, the machine can "know" when the louder and quieter passages are coming, with a fixed "advance notification" set by the spacing between the heads, and the speed at which the tape passes them.

Now we come to how technological improvement influences system design. In the early days of phonograph design, a pickup required a force measured in ounces, to keep the stylus (then more often called a "needle") in the groove. Improvement in pickup design has brought this force down, until it is measured in grams, or even in fractions of a gram. This has been brought about, not only by changes in pickup design, but also by changes in record material.

The old records used a 3-mil stylus, or needle. Nowadays the maximum is 1-mil. The old needles wore down quickly, due to the abrasive in the record, part of whose purpose was to prevent the needle going out of shape—which it did, anyway, more quickly than its user liked: to get good performance, the needle should be changed after each record played (and that was less than 5 minutes' duration in those days). The advent, first of sapphire styli, and then diamond, changed this, as well as making the smaller radii practical.

This opened the way for longer playing discs, the EP and LP, each of which adopted its own systems design approach. RCA's Extended Play (EP) introduced the large hole in the center, adapted to a new system of record changing, suited either for juke boxes, or for the teenage record-player market. CBS' Long Playing (LP) used the same sized discs as before, but changed groove size and speed. Groove size dropped from 3 mil to 1 mil, and speed dropped from 78 rpm to 33 rpm. RCA's EP chose 45 rpm, also with a finer groove size, and a smaller disc, for greater convenience.

SYSTEM DESIGN APPLIED TO RADIO

All of these choices involved systems design concepts, to optimize what its designer wanted the system to do. Meanwhile—in fact long before that—radio was having its problems. While the disc recording was bothered with how much program could be squeezed into each disc, radio was bothered with how many channels could be squeezed into the "air space"—or radio spectrum.

This is what led to the adoption of f.m. for the higher radio frequencies: to use up more of the spectrum, to get audio modulation of higher dynamic range, rather than over-modulating on the very limited side-band range provided by a.m. Here again, systems design concepts led to choice of equalization, channel spacing and so forth.

WHAT ABOUT STEREO?

Now into this scene with already-proliferating systems, enter stereo. Up till

Circle 15 on Reader Service Card

MARTIN DICKSTEIN

db Sound With Images

A Bit More On The Visual Communications Congress

• You will recall that last month we discussed the meetings that took place at the Visual Communications Congress, held at the beginning of the summer. It was mentioned, in that column, that we would try to cover the exhibits that were shown during the Congress. Well, that time is here. We'll do it alphabetically; simply because we do not want to show any preference, nor do we mean to imply a "plug" for any of the equipment or services. The various exhibits are being mentioned and described, here, to indicate the scope of the overall coverage and to provide information about some of the items, of which you may not be aware. By the same token, we will not mention all of the exhibitors, as there were about 300 of them—and just listing them would run a full length column and more.

Among the "A's," there was American Innovative Marketing of Santa Ana, CA, showing how a "Digi-tiser programmable message center" worked; American Movie Network (NYC) explained their computerized closed-circuit pay tv system; Ampex; and Anvil Cases, manufacturer of custom made transportation and storage cases for video, audio-visual, camera equipment and delicate instruments.

Bergen Expo Systems (Clifton, NJ) was there displaying their Xenon slide and 16mm projectors, Quadapoint Light Valve, Eterna 16mm endless loop

(Continued on page 30)

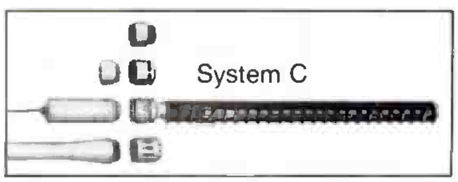
The CS15P condenser cardioid microphone is equally at home in a recording environment or broadcast studio. When hand-held it puts sex appeal in a voice with its bass-boosting proximity effect. With shaped high-frequency response and its ability to handle high sound pressure levels (140 dB with 1% THD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

When boom mounted, the CS15P has better gain-before-feedback and a better signal-to-noise ratio than most shotguns. It's phantom powered and it's rugged.

The CO15P condenser omni extends frequency response to the very limits of audibility, 20 to 20,000 Hz. Unlike other "omni's," the CO15P maintains its omnidirectional polar pattern at the very highest frequencies. Perfect for the distant miking of an entire orchestra as well as up close on individual instruments. And like the CS15P, it's phantom powered and it's rugged.

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

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



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a gulton company
600 Cecil Street, Buchanan, Michigan 49107

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OUR COMPETITORS SWEAR THIS KIND OF DIGITAL QUALITY IS STILL THREE YEARS OFF.

Almost everybody admits digital recording will someday be the ideal way to make records.

With a Sony PCM-1600 system, you can stop waiting for that someday to happen.

And start recording now.

IN THEORY, ALL DIGITAL SOUNDS TERRIFIC. IN REALITY, SONY ACTUALLY DOES.

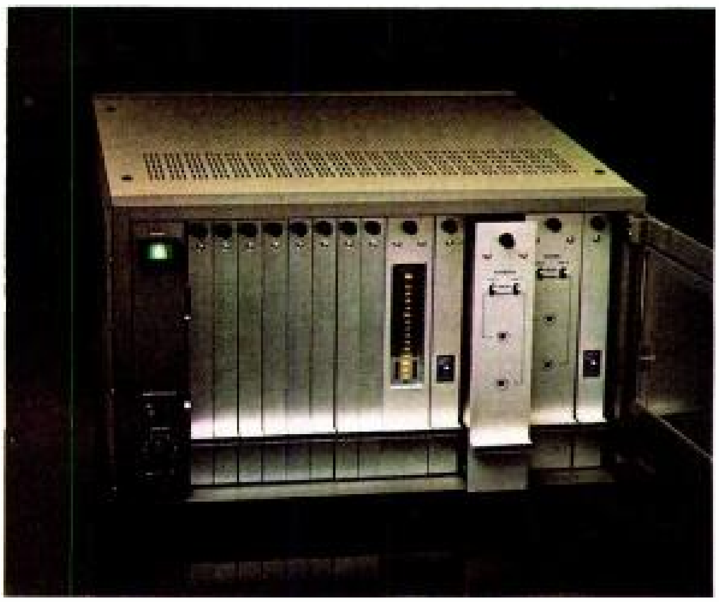
If your ears are even in semi-reliable condition, you've no doubt been exposed to incredible amounts of digital audio hype.

"The most dramatic change in audio since flat records," says one prestigious audio publication. "At last...perfect sound reproduction," notes another.

So instead of dwelling on how digital can improve the sound of your recordings, we'd rather dwell on how Sony—and only Sony—has finally made digital performance live up to its promise.

"WE'D LIKE TO DEDICATE THE NEXT FEW NUMBERS TO THE SONY ENGINEERING DEPARTMENT."

In order to convert regular audio signals into digital, the Sony PCM-1600 digital recorder uses a pulse code modulation system developed and perfected by Sony engineers.



The same people who have been behind virtually every audio and video breakthrough since the late 1940's.

Consequently, the dynamic range of the two track PCM-1600 is greater than 90dB.

Frequency response is 20 to 20kHz.

The total harmonic distortion of the PCM-1600 is inaudible over the entire audio spectrum.

There is no measurable wow and flutter.

And because the PCM-1600 puts binary code on tape instead of audio signals, it's not susceptible to tape hiss or print-through; there's no need to subject your music to noise reduction equipment or realign your recorder for bias and equalization.

EDITING PROBLEMS HAVE BEEN FADED OUT.

Since the PCM-1600 works in conjunction with two Sony video recorders, a standard Sony video editor, the BVE-500A or the new DEC-1000 will fill your every editing need.

DROP-OUTS HAVE BEEN DROPPED.

Based on a technique used by computers (where the loss of a single bit of information could mean millions), Sony engineers have created an ultra-sophisticated digital correcting code that can actually restore "dropped out" information.

And considering that Sony video recorders are virtually immune to this problem in the first place, the chances of it plaguing your sessions are all but negligible.

DIRECT-TO-DISC QUALITY WITHOUT DIRECT-TO-DISC LIMITATIONS.

Because digital quality doesn't deteriorate from one tape generation to another, the PCM-1600 lets you make generation after generation of lacquers, all with direct-to-disc quality—but without any limitation on the number of lacquers you can press.

And if you use the digital format for distribution to your foreign affiliates, you're guaranteed that your last digital master will be absolutely identical to your first.

SONY INTRODUCES TWO NEW WORDS TO THE DIGITAL VOCABULARY: AVAILABLE NOW.

Unlike other digital equipment, the PCM-1600 is actually available for delivery today. Not six months from now. Or "when we finally get all the bugs out."

PCM-1600's are currently being used by well-known recording stars, and are already in operation at Polygram, CBS, Spectrum Sound and Digital Recording Services in Los Angeles, and Digital Recording Systems in Philadelphia.

A FEW SONY DIGITAL ADVANTAGES YOU CAN'T HEAR.

Besides the quality and reliability Sony equipment is legendary for, we've set up a 24-hour digital hot line to service our customers.

We're prepared to provide as much technical expertise and engineering assistance as only Sony possibly can.

And we intend to continue our exhaustive research in digital audio—and share future insights with the growing Sony digital family.

If you'd like more information about any facet of Sony digital equipment, contact Roger Pryor at (415) 467-4900.

You may think you've heard a lot about digital in the past.

But until you've heard Sony, you haven't heard anything yet.

SONY.
DIGITAL AUDIO

We've never put our name on anything that wasn't the best.

sound with images (cont.)

attachment, and to introduce the Mini-X2 16mm film projector with Xenon lamp-house attached. Blackbourn, Inc. of St. Paul, MN, showed video storage albums for ½-inch and ¾-inch tapes, albums for 5-inch and 7-inch video reels, and their multi-media packaging; while Buhl had slide and film lenses, film and video multiplexers, and a high resolution slide projector.

Among the "C's," there was Cargo Case Division, ICOM, Inc. of Columbus, OH, manufacturers of heavy-duty protective shipping cases. Clear Light (Ft Lauderdale, FL) showed its multi-image control devices; Command Productions (White Plains, NY) displayed samples of their slides and filmstrips, and discussed their graphic and audio-visual production services; Creative Communications Group of Dallas, TX, also had a multi-image and live stage production to offer; and Crestron Electronics (Closter, NJ) had four models in their Executive Series of wireless audio-visual systems to show

Devlin Productions (NY) had a booth to discuss their new Mark 3 Flying Spot with Digiscan for film-to-tape transfer, their world-wide television standards conversion equipment and services, their CMX 340X hi-band computerized editing system.

Fiberbilt Photo Products, a division of Ikelheimer-Ernst (NY) showed their AV carrying cases, tv shipping cases, and monitor shipping cases; Freen Screen (Rockville, MD) discussed their rear screens; GE displayed its large-screen color video projector; Genigraphics showed the results of its slide and VU-graph process by computer; and Hitachi demonstrated its portable and studio camera systems.

Image Transform (NYC) talked about their videotape-to-film transforms in 16mm and super 8mm, and their film-to-tape transfer with the exclusive ITM telecine; Image West (Hollywood, CA) showed electronic animation; and "The Incredible Slidemakers," (NY) showed special effects slides and how they can help with multi-image presentations.

Kimchuk (Brookfield, CT) demonstrated their method for stacking slide projectors, StereoStand (for 2 projectors), 3-Hi for 3 projectors, and 4-Hi for guess-how-many-projectors, and their animator unit which comes with modules that can be added together for different functions.

Elsewhere, in the "L's," Lewis Lektronics (Santa Clara, CA) showed its portable programmable electronic moving-message sign, with a detachable keyboard programmer; while Loose Leaf Industries (South Plainfield, NJ) was exhibiting custom AV packaging for cassettes, filmstrips, records and slides.

Continuing with our survey of the Visual Communications Congress, random slide projectors and controls were displayed by Mast. Mediatech, from England, discussed their international equipment rental service; and Motion Message of Bohemia, NY showed their fully programmable electronic read-out sign. Lowell Nerge Filmstrips (Minneapolis, MN) explained their "direct process for making film strips from 35m slides;" Optical Radiation (Azusa, CA) showed a programmable high-intensity Xenon slide projector; Optisonics HEC (Tucson, AZ) showed slide/tape synchronizers, dissolve units, variable dissolve/synchronizers, remote projector controls, and portable slide show carrying cases; and Pak/Master (Princeton, NJ) had custom AV packaging and complete warehousing and shipping services to offer.

Reliance Plastics and Packaging (Forest Hills, NY) demonstrated custom packaging for audio-visual materials; The Silver Image (Washington, DC) discussed AV software; Spindler & Sauppe (Hollywood, CA) displayed multi-image presentation systems; and Theater Techniques (Newburgh, NY) showed backdrops for tv, films, etc.

There were many, many more, too many to mention all, but this just goes to show that you should not miss the next Visual Communications Congress if you're in the field of AV. ■

db New Products & Services

LOUDSPEAKER SYSTEM

• A variable dispersion loudspeaker system, the model 701 PRO MASTER yields big speaker performance in a compact, lightweight package. Each speaker system consists of a 15-inch woofer in a front-ported bass reflex cabinet and a high frequency horn and driver combination. Operation of the variable dispersion control involves turning a knob located in the mouth of the horn to settings of 60° (narrow, "long throw" coverage) or 120° (wide-area, "short throw" coverage. Power handling capacity is 150 watts of continuous program material. Overall enclosure dimensions are 27-5/8" x 23" x 15-13/16".

Mfr: Shure Brothers Inc.
Price: $495.00
Circle 70 on Reader Service Card

AUDIO CONSOLE

• A stereo console, the Howe 7000 handles 22 inputs through 12 channels, and utilizes d.c. voltage-controlled audio mixing. All line inputs are 10,000-ohm or greater, bridging, and all microphone inputs are 600-ohm. All outputs have a rated frequency response of ±1 dB, 20 Hz to 20 kHz, with 0.09 per cent, or lower, distortion. The console contains full rfi protection, with balanced-in/balanced-out electronics.

Mfr: Howe Audio Productions, Inc.
Price: $3,995
Circle 71 on Reader Service Card

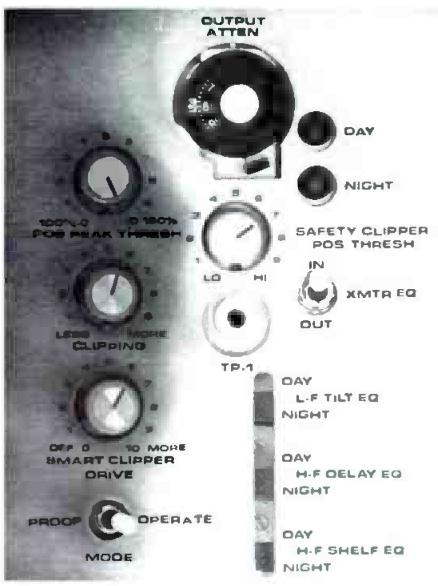
ELECTRONIC MICROPHONE SNAKE

• A system for transmitting low level signals from a stage to mixing or recording equipment, the MAINLINER utilizes time-domain multiplexing. Each input is encoded and transmitted over a length of standard microphone cable to the decoder, where the signals are reassembled at their respective outputs. Lo-Z mic inputs are standard on all models, however Hi-Z inputs are available as well. Cable runs of 700 feet are practical prior to any signal loss.

Mfr: JHD Audio

Circle 72 on Reader Service Card

TRANSMITTER EQUALIZER



• Substantially improving the average modulation and loudness capabilities of many transmitters using older modulation techniques, a new transmitter equalizer accessory is now available for use in the OPTIMOD-AM Compressor/Limiter/Equalizer system. Designed to compensate for low frequency tilt inherent in many transmitters, and also to compensate for transmitter-antenna system overshoot and ringing, the equalizer contains two separate sections (remotely switchable) permitting independent adjustments of day/night transmitters or day/night power levels. Each equalizer section has three controls: one for low-frequency tilt, and two for high-frequency compensation. When desired, the entire transmitter equalizer can be switched out. A front panel jack permits insertion of square wave test signals for initial set-up. Day/night status of the unit is indicated via led's. This new transmitter equalizer is available as an accessory kit for easy retrofit into existing units, and is being included on all current production units of the OPTIMOD-AM Compressor/Limiter/Equalizer system.

Mfr: Orban Associates, Inc.

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

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

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

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

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

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

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

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

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

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

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

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

OTARI®

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

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TAPE MACHINE

• This four channel, ¼-inch tape audio recorder/reproducer system incorporates an analog to digital—digital to analog converter with 16 bits quantizing accuracy. With an operational speed of 15 in/sec., the closed loop capstan tape transport system provides constant tape tension. The machine, equipped with thin film write and read heads, offers a frequency response of 20 Hz to 20 kHz within ±0.5 dB, a dynamic range of 90 dB minimum, and a total harmonic distortion of less than 0.05 per cent. Utilizing Miller Code modulation, each channel contains four tracks—3 data and 1 parity, with a data block rate of 1200 Hz.
Mfr: Technics
Circle 74 on Reader Service Card

SHIPPING CASES

• Available in nine standard EIA-RETMA rack sizes, the Rack-Pack is a case for shipping rack-mounted electronic instrumentation. The rack frame, contained inside the case, is constructed of welded aluminum and is mounted to the outer walls by shock mounts, to isolate the instrument from shock and vibration. The case, itself, is made of molded one-piece rugged plastic. Both the front and back covers can be removed quickly by quarter-turn latches, and the extruded aluminum tongue-and-groove rims on the covers form watertight seals. Male and female ribs on the top and bottom of each Rack-Pack interlock for stability when several cases are stacked.
Mfr: Thermodyne International Ltd.
Price: $490.00 to $975.00
Circle 75 on Reader Service Card

SHOCK-MOUNTED MICROPHONE

• Primarily intended for hand-held broadcast applications. the RE18 shock-mounted super-cardioid microphone is equally at home in any situation where ambient noise rejection and isolation from handling noise is a consideration. The RE18 maintains the frequency response and super-cardioid polar pattern of the RE15 microphone; while sharing the integral blast filter of the RE16. An added advantage of the RE18 is the Variable-D design which maintains frequency response regardless of mike-to-talent working distance.
Mfr: Electro-Voice, Inc.
Price: $226.00
Circle 76 on Reader Service Card

Concert Mk II Mixer – new dimensions in creativity and control

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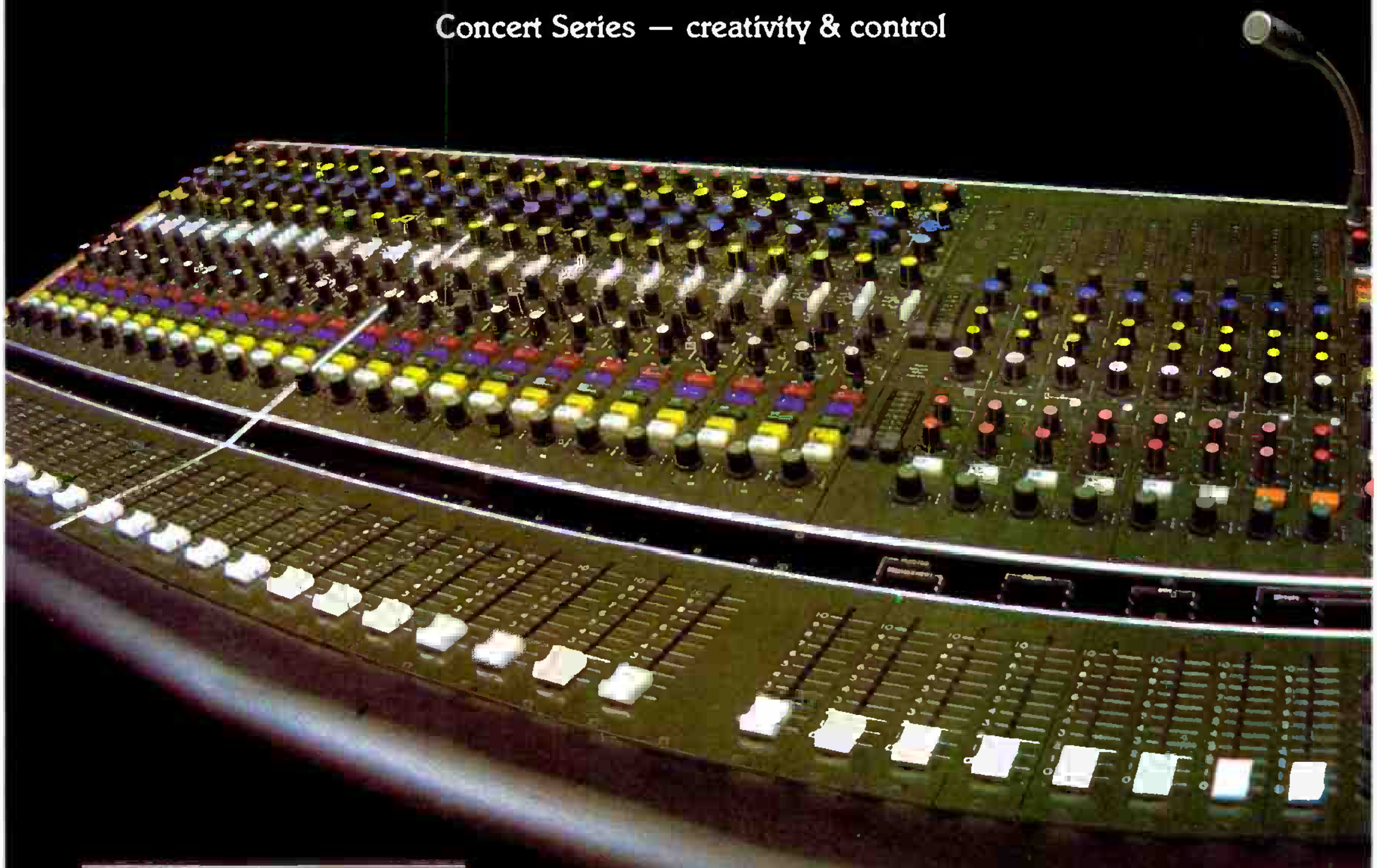

STRAND SOUND

A Division of Strand Century Company Limited — A Company Within the Rank Organisation
6520 Northam Drive, Mississauga, Ontario Canada L4V 1H9 Tel. (416) 677-7130

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● Specifically designed for use in moderately noisy conditions, the Sportscaster II boom-mic headset is ideal for live sports broadcasting. The unit features an omnidirectional broadcast-quality dynamic microphone and in-line push-to-cough switch for microphone muting. The binaural headphone receivers attenuate noise while allowing the announcer to monitor the program in one ear, and receive cues in the other. A snap-on foam filled headband cushion provides ventilation during prolonged periods of use.
Mfr: Telex Communications Inc.
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DIGITAL MULTIMETER

● A low-cost 3½ digit benchtop digital multimeter, the model 1351 can measure a.c. or d.c. currents up to 20A, with a basic accuracy of 0.1 per cent. Pushbutton controls are provided for all 34 ranges and functions, and measurements are displayed on a bright, 0.43-inch led display. The model 1351 measures d.c. volts from ±100 μV to 1200V, a.c. volts from 100 μV to 1000V rms, and resistance from 100 MΩ to 20 MΩ with either high (2.8V) or low (300 mV) excitation. The unit comes complete with tilt stand and carrying handle, test leads, spare fuse and 1-year warranty.
Mfr: Data Precision Corporation
Price: $199.00
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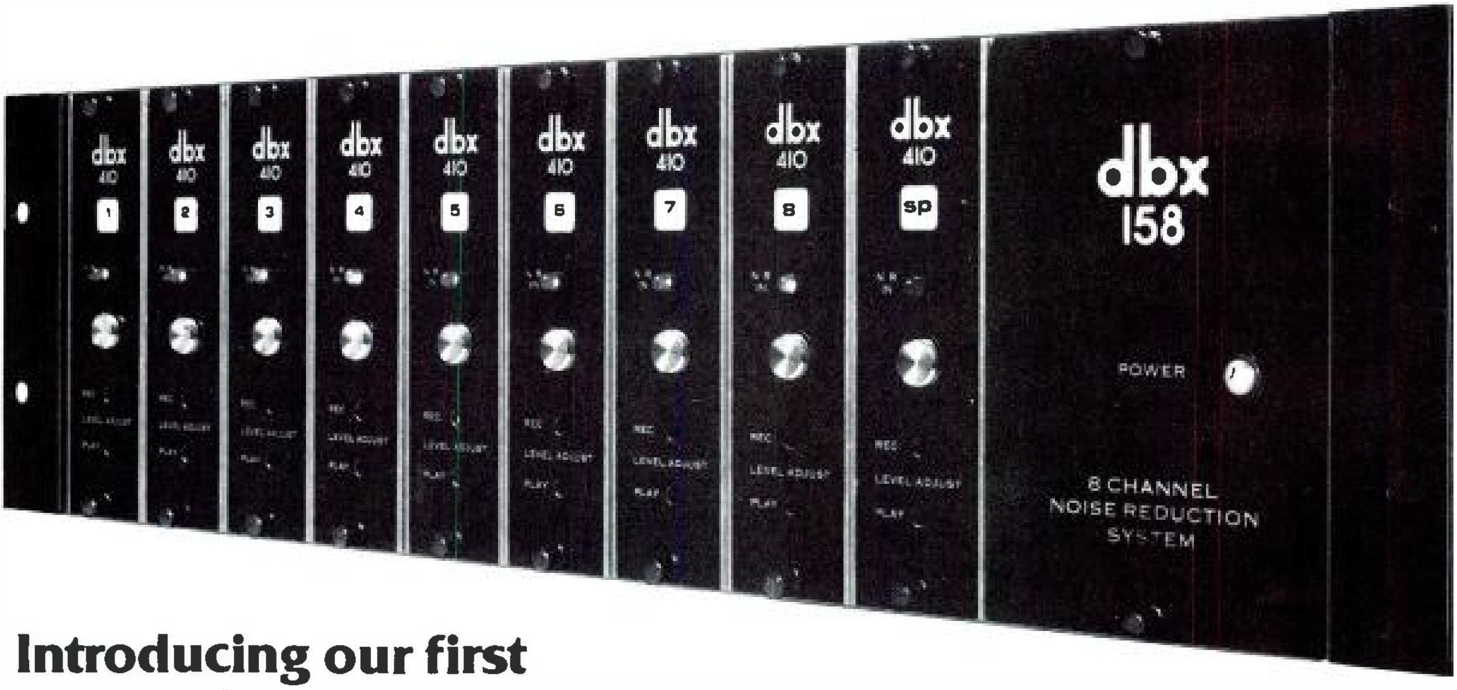
CONTROLLER-TIMER

● A compact tabletop controller-timer that makes cartridge production easier, faster, more accurate and consistent, the Upstart produces tight cartridges without clicks, pops or upcuts. The Upstart controller-timer, in sequence, will start and pre-roll turntables or reel-to-reel tape players (regardless of start-up time), start and pre-roll cartridge recording machines, noiselessly switch on the audio, digitally time the cartridge (while separately timing intro-to-vocal and intro-to-outro), and remove audio at the end of the program. With a front panel switch permitting two speed operation from turntable or tape machine, the unit features a large, bright, digital timer display—reading minutes, seconds, and 10ths of seconds. The Upstart interfaces with most turntables, reel-to-reel players and cart machines.
Mfr: Sharepoint Systems, Inc.
Price: $495.00
Circle 79 on Reader Service Card

dbx 158.
IT'LL GROW ALONG WITH YOU.



Introducing our first economical, expandable, modular, simultaneous tape noise reduction system.

Now you can have a tape noise reduction system that will stay with you from high-end audiophile, through semi-pro and into full professional equipment.

Our new dbx 158 system can start life in your place with the 158 main frame and as few as two modules or as many as eight modules for its full eight channel capacity. It also has storage space for a ninth spare module in its compact chassis. The rear panel has phono and multi-pin connectors that will interface directly to your cables. Additional 158's can be used for 16 or 24 track recording.

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

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

dbx

dbx, Incorporated
71 Chapel Street
Newton, Massachusetts
02195
617-964-3210

Here's a generous offer: buy all 8 channels up front, and we'll throw in the ninth module free.

Circle 21 on Reader Service Card

AUDIO PROCESSOR

• A multiple-option audio processor, the Model 215 offers radio stations a "building block" approach to audio processing through a number of equipment options. As an a.m. or f.m. peak controller, the 215 maintains program peaks within desired limits at the transmitter site. The 215 can also be equipped for use only as an AGC or as an AGC and compressor, with the AGC option gated to provide a slow, "gain-riding" function for wandering program levels. The compressor option features a "soft-knee" transfer function for smooth, unobtrusive control over average program dynamics. In addition, two peak controllers are available with the 215: a phase-following, asymmetrical version for a.m., and a 25/75 usec f.m. limiter. Of course, with all option included, the Model 215 stands alone as a total audio processing package.

Mfr: Inovonics, Inc.
Price: $375 to $785, depending on the options selected
Circle 80 on Reader Service Card

DISTRIBUTION AMPS

• Boasting the highest headroom of any unit on the market, the Model 815 and 8151 audio distribution amplifiers are separately powered modules with screw terminal connectors, providing easy installation without the need of lugs or soldering. The model 815 provides 10 outputs (with a maximum output of 26 dBm rms) and gain of 0 to 30 dBm. The 8151 provides 6 outputs with individual gain adjustment from 0 to 30 dB, and a maximum output of 21 dBm rms. Inputs may be bridging, balanced or unbalanced, and outputs are electrically balanced with 600-ohm source termination. Optional input transformers for external differences in ground potential are also available. The Distribution amps are available individually, or in rack frames holding up to 10 units.

Mfr: Dyma Engineering
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Professionals depend on their equipment.

Like their BGW amplifiers. Why is it so many have come to rely on BGW? Why in less than ten years have BGW amps become the number one choice among audio pros worldwide?

Because their legendary performance refuses to fail even under the most severe conditions you can throw at them. Rugged, awesome power that's been tamed by continuous common-sense engineering. That's why there are more BGW amps in discos than any other kind, and why there are so many in recording studios and on concert stages. BGW has earned a reputation for building superbly engineered products ... massive heat sinks, large safe operating area, redundant output stages, welded steel modular construction are all synonymous with a BGW product.

We are now proud to introduce a new cost-effective 175 watt per channel power amplifier... the Model 600. It's a quality basic power amp, built around our super reliable 750 B/C output modules. It's in a big 8¾" high rack-mount package so it runs cool and costs substantially less than a 750C. It's a quality BGW amp and the answer to the professional who wants BGW on a budget.

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Circle 52 on Reader Service Card

db Editorial

EVEN AS THE DEADLINE for this issue of **db** came closer and closer, we were still casting about for a way to properly cover the subject of education in audio. Should we once again offer features about different schools, we wondered? We tried that last year (February, 1978), but the problem remains; how can we accurately identify those schools that really give the student a sound (sorry about that!) education? And, what about the rest of our readers? After all, you're not *all* students, are you?

Think about that for a minute. Maybe all of use are— or, maybe, we *should* be. No, we don't have to quit work and go back to the classroom full-time. But, what about a bit of "continuing education" while still on the job? Hardly a month goes by that we don't encounter even more things to learn about—just to keep pace with our industry. As for getting ahead of it—well, maybe *next* month.

Well students, how many of us *really* know all there is to know about say, magnetic tape, computers, spectrum analyzers, digital technology, and such? If you're like most of us, you're having trouble enough recalling everything you learned back in school, and now you're being hassled by ROMs, A/D converters, aliasing, and bit streams, in addition to remembering oersteds.

When will it end? That's an easy one—never! Of course there's always retirement. But for as long as you want to stay active, you'll have to stay educated. That means remembering the past, and keeping up with the present— or at least, trying to.

So (this being education month here at **db**), we asked our November authors to keep the educational aspects of pro audio in mind as they prepared their manuscripts for publication. You'll find the results, and the education, on the following pages.

David Rubenstein concludes his two-part overview of magnetic tape specifications. For more "continuing ed" on various aspects of this subject, please look up our January, 1979 issue, where you'll find detailed accounts of tape performance, speed-versus-biasing, etc.

Next, Richard Factor turns his personal computer (a Commodore PET) into a spectrum analyzer. (Your editor continues to grapple with an Apple II, while the publisher is kept awake nights by a Radio Shack TRS-80.)

For some time now, we've all been duly impressed by the ease with which the personal computer can tackle audio projects. And, the p.c. can prove to be a relatively-painless introduction into the computerized studio of the future (the very near future, we might add). As more and more pro audio hardware becomes "computer-based," there's less and less chance of avoiding the subject. Here at **db**, its all part of our continuing education—yours too, we hope.

From Ampex comes news of yet another digital delay system. But this one is somewhat different in that it was designed for primary channel applications. In describing the system, authors David Haynes and John Brennan give us a brief education in DDL basic theory.

To conclude, Irwin Diehl reports on his recent one-week education at MCI's "basic training camp" in Fort Lauderdale, Florida. While there, Diehl and his fellow students went through a rigorous series of classroom and lab sessions, and came out knowing a lot more about state-of-the-art console and tape recorder theory and practice.

And speaking of state-of-the-art consoles, our recent directory of "super consoles" inadvertently failed to include the manufacturer of the "Memphis Machine." The company is perhaps best known for its "Son of 36 Grand" consoles, and if that's not enough of a clue, see our Super Console Update, below. And for still more information, stay tuned; we hope to be able to convince someone in Memphis to tell us more about the console's Auto-trak® Automation System in a future issue of **db.**

SUPER CONSOLE UPDATE

Auditronics, Inc.
3780 Old Getwell Road
Memphis, Tennessee 38116
(901) 362-1350

Trident Audio Developments, Ltd.
In addition to Sound 80, Inc., Trident consoles are also distributed by:
Empirical Audio
3A Todd Place
Ossining, New York 10562
(914) 762-3089

Studio Maintenance Services
12444 Magnolia Boulevard
North Hollywood, California 91607
(213) 877-3311 ■

db Special Report

JOHN M. WORAM

Education and Audio

THIS MONTH'S LETTERS to the editor confirm the fact that education in audio is still a timely topic. Once more, we wish there was a "yellow pages" of reputable schools that we could recommend without hesitation. However, when it comes to audio (and this especially applies to recording), *everyone* is teaching something.

In February, 1978, we published several features about audio schools. One reader took us to task for neglecting to mention the many other fine schools that are around. But the problem remains—how do we tell the good ones from the not-so-good?

Coming to the rescue of the prospective student in search of an education, is the Audio Engineering Society. The Society has compiled a Directory of Educational Institutions which is now available from them on request. But, it is very important to note that the Directory has been compiled from information received from the institutions themselves. Therefore, it is *not* a list of recommended schools, any more than the yellow pages contain only recommended phone numbers. The Directory simply lists the schools that have chosen to respond to the AES questionnaire.

We urgently suggest that the student investigate several schools. If possible, talk to the faculty and certainly, to the students. Some schools are so academic, the faculty wouldn't know a recording studio control room from an aerospace command center. Others are merely recording studios in search of a little cash flow. Regard both with healthy suspicion.

Most reputable studios gladly host occasional seminars, AES meetings and such, simply as a means of supporting the audio industry and contributing to its growth. But a good studio can make far more money from sessions than from students, so unless the tuition is astronomical, think it over carefully before enrolling. (If the tuition *is* astronomical, you don't need us to tell you to think it over carefully!)

Still more help may eventually be on the way. NARAS (The National Academy of Recording Arts and Sciences) recently sponsored a conference on education in Nashville, Tennessee. Out of the conference came the Music Industry Educators Association. A news release from David Leonard, the association's vice president, describes MIEA as, "...a non-profit organization whose purpose is to help maintain the highest possible professional and academic standards in preparing students for employment in the burgeoning recording industry."

Mr. Leonard is executive director of the Trebas Institute of Recording Arts in Montreal, Canada. Trebas offers a 600-hour (ten hours-per-week, for four 15-week terms) program in recording arts and sciences. As for MEIA, the news release doesn't specify how the association plans to function, or, what services it will offer. Hopefully, we shall have more information from them later, and will pass it along, when and if.

For a copy of the Directory of Educational Institutions, write to:

The Audio Engineering Society, Inc.
60 East 42nd Street
New York, N.Y. 10017
(212) 661-8528

For information on Trebas Institute and presumably, MIEA, Mr. Leonard may be reached at:

Trebas Institute of Recording Arts
1 Place Ville Marie, Suite 3235
Montreal, Quebec, Canada H3B 3M7
(514) 871-1067

■

IRWIN J. DIEHL

School Days at MCI

Our 'roving reporter' discovers how one company provides valuable continuing education for its customers, and others wishing to keep up-to-date.

In some industries, certain licensed professionals are required to periodically refresh and update their knowledge and skills. Of course, there are still no licensing or formal education requirements in the audio industry, but a strong personal desire to improve understanding still motivates many working audio pro's to continually update their on-the-job skills. So, although going "back to school" is not absolutely necessary, it is nevertheless an important activity for many people in our industry.

Probably one of the better bargains available in this category of education today is MCI's training program in Fort Lauderdale, Florida. A most-prolific manufacturer of professional multi-track recorders and consoles, MCI offers an *intensive* one-week technical course on the function, service and troubleshooting of its current product line.

A recent class of some fifteen students (an average enrollment for these sessions), gathered one week in September at the company's new headquarters. Formerly the international headquarters of STP Oil, this new MCI plant presently accommodates tape recorder engineering, drafting and manufacturing divisions, in addition to the training center. Two other buildings are now maintained for metal fabrication, console production, service department, executive offices, etc., however all operations will be brought into the new quarters by February, 1980.

"Intensive" is a term often bandied about to suggest the would-be depth or detail of short-term training programs seminars. In the case of MCI's course, the term applies with full weight. The five-days of classes begin at 8:00 A.M., and continue through the day, until 6:00 P.M. Except for a one-hour lunch break and one or two brief coffee breaks, in each day's work, the students steadily pace through signal-flow diagrams, schematics and operational theory of the company's current-model equipment. In addition, the evenings are *filled* with homework comprehensively testing the students' retention of the day's work. These tests are submitted and graded. Certificates awarded at the end of the week's work are not conferred unless all exams are correctly completed and submitted.

The school, which began several years ago more-or-less spontaneously as a one-on-one service department response to customers' requests has recently been reorganized by the Engineering Publications Department. Al Simons, a member of that department, is responsible for structuring the classes, and is often-present to maintain the course on schedule. With the aid of Don Czekanski, also a member of Engineering Publications, somehow all the topics listed in the extensive five-day course outline are covered.

The course is updated regularly to encompass new product additions, but always is aimed at teaching basic equipment functions, service procedures, and troubleshooting tips and methods. The students' class time is split about evenly between lecture and lab; labs being opportunities to work in small groups with field service or product engineering staff, to observe and perform various adjustment procedures.

MCI CURRICULUM

The principle course subjects correspond with the company's principle products: tape recorders and recording consoles. In addition to these main topics, a fair portion of class time is given to related subjects such as: Autolocator III, RTZ III (a return-to-zero locator), theory and servicing of plasma meter displays, plus theory and operation of several models of equalizers.

Intended prerequisite to the course is an Associate's Degree in electronics or equivalent experience. However, in most cases, the background of students is varied, and for this reason a certain amount of time is devoted to discussions of basics in the areas of: applications of op-amps, power supplies, and digital, and microprocessor technologies. For example, design engineer Ted Starros reviews basic theory of magnetic recording before delving into the particular refinements of the JH-110 and JH-16 audio electronics. Theory of operation of transports is also covered, and lectures thoroughly detail the function of control circuits that provide reeling tensions and deck logic. Approximately one-half day is scheduled to explain the theory and servicing of the Autolocator III and RTZ III. Typically, some half-dozen or more MCI staff-members participate as course instructors.

TAPE RECORDER SESSIONS

Some of the more interesting tape recorder sessions are the labs, conducted by various members of the Field Service department. The JH-110 and JH-16 transport set-ups, check-outs and adjustments are demonstrated step-by-step, augmented along the way by tips and hints for improving servicing and troubleshooting proficiency. Also, complete audio alignment procedures are explained, including physical alignment of head assemblies, guides, "dancer-arms," lifters etc., as well as bias and record reproduce electronics calibration.

Upon completing the first two days' lectures, labs and exams, attention shifts to consoles, automation and related subjects. Prior to embarking on these topics, the company's

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Irwin Diehl, founder of the Institute of Audio Research, New York, is currently employed as an independent consultant.

microprocessor design engineer, Manny Guerrero, joined our class, and offered a glimpse of MCI's "automation architecture." The discussion developed an appreciation for the automation system heirarchies which define branching, looping and other such routines. From this foundation, it's a more-or-less easy matter to follow the processes by which various automation-routines (writing, reading, updating etc.) are implemented in software.

A complete and thorough analysis of the JH-500 signal-flow block diagram explains function, logic, gain and levels within each board module. The inter-connection of modules and "switching logic" are also reviewed.

The fine points of transport logic are reviewed by MCI's Bob Bryant (seated) and Dan Czekanski (standing).

Company president "Jeep" Harned studies George Kuchmas' final adjustments on an MCI transport.

In a factory test area, Gregg Lamping tries out some "field service" procedures, complete with portable tool case close at hand.

A 24-track recorder and its Autolocator get their innards thoroughly checked-out.

Project engineers for MCI's most-recent board, the JH-600, contribute a lecture on theory of operation of the plasma meter display. The meter's central component, Burroughs' 100-element plasma display device is described by engineer Bill Houston. His remarks concern both field service and troubleshooting hints, as well as theory of operation of the metering system.

CONSOLE DESIGN & PERFORMANCE

On day four, the JH-600 console is introduced. This new board replaces the 400 series consoles, but is available as an automated-only system. Some of the more-fascinating circuit features of the JH-600, such as the transformerless, differential-input and differential-output amplifiers, are developed fully on the blackboard. The design choices made for these system components are discussed openly, providing insight into overall design and performance criteria.

This same type of scrutiny of circuit options and choices is also applied to the Vari-EQ equalizer, a new product with bandpass, low-pass and high-pass sections, each equipped with variable-Q adjustments. The analysis of these circuits proves to be a mini-course in itself, in the design of active filters.

The fifth and final day is spent in the console manufacturing section (Plant 1) where both JH-500 and JH-600 boards are assembled, tested and readied for shipment.

Customer service engineers with "specializations" in these consoles become the day's instructors. Board check-outs and troubleshooting procedures are highlights of the sessions. Lab groups alternated between both JH-500 and JH-600, in order to gain an understanding of each.

During these labs, students visit the final module check-out area, where every board module is put through its paces. The extensive testing techniques are demonstrated and, where necessary, components of modules are changed or adjusted to match performance with specifications.

The five-day course was over before we knew it. Somehow we covered a 26-page course outline and worked through much of the contents of seven large equipment manuals. Many of MCI's design engineers and field service staff had turned out at one time or another to lecture the class or assist during lab sessions. Each student had an ample opportunity to ask specific questions and receive answers based on the firsthand knowledge and experience of the MCI staff. All in all, this was a very positive educational experience.

But the best has been kept 'til-last. If a student were to enroll elsewhere, in a private school for example, where a dozen or so experts lecture for five days, and more than a half-dozen binders filled with current reference material are provided, he might expect to pay a fee of anywhere from $400 to $600. Not so at the MCI Training Center. The fee for this course is just $100. Surely this is one of the better bargains today in technical training and education. ■

RICHARD FACTOR

Spectrum Analysis On a Personal Computer

Serving as an educational tool, the computing spectrum analyzer can effectively introduce the audio engineering student to some of the fundamental audio concepts of frequency, amplitude, and sound perception.

WHEN THE EDITOR told me that the theme of this issue of **db** was Audio Education, I went casting about for suggestions on relating the issue topic to my topic, a computer spectrum analyzer combination. A co-worker, whose previous occupation was teaching in the New York City public school system pointed out, I think correctly, that perhaps fifty per cent of education lies in getting the student's attention. He proceeded to recommend hitting them on the head with the spectrum analyzer. While the latter point may well be valid in his teaching context, I feel that the first point, getting the student's attention, is valid even within the context of well-motivated students and impeccable decorum which may be expected in college and specialized-institution classrooms. Why should attention-getting mechanisms be necessary, and how does the spectrum analyzer relate to them? And while we're at it, how else can the computer-analyzer assist the educator? This article will present a case for making this instrument an indispensable teaching tool, instead of an afterthought brought in for one lesson to show how certain measurements are made.

GETTING THEIR ATTENTION

To an audience accustomed to obtaining information by seeing, the concept of Sound, regardless of the fact that it is readily detectable by at least one sense, (two, in the case of loud rock concerts and "disco"), must be regarded as "abstract." Despite the amazing acuity of the human ear, there is nothing intuitively analytical about sound. One cannot infer, by unaided observation, that a high-pitched tone corresponds to a certain pattern of molecular densities in the air and, that a lower-pitched tone corresponds to a different pattern with greater spacing between density maxima. We can much-less guess what these physical facts have to do with sonic perception, control room design, loudspeaker limitations, or any of the manifold phenomena directly related to the physical nature of the sound and its interaction with electronic equipment. The common response to this human "inadequacy" is to trot out the cathode-ray oscilloscope, and look at the time-versus-amplitude display of the signal in question. Unfortunately, this is not really what is desired. While the time amplitude display is a vital one for studying audio from the viewpoint of the engineering discipline, it is (at least initially) only peripheral to the concern of the musician, the producer, and frequently even the novice recording "engineer." These students are interested in the relationship between what they perceive and what they can control by means of the recording technology soon to be at their disposal. It is in this context that the oscilloscope falls drastically short.

The human ear is sensitive, typically, to the frequency range of 20 Hz to 20 kHz, and to an amplitude range of about 120 dB (although not all at once). Depicting two simultaneously-audible signals on a scope is usually impossible. If the signals differ in amplitude significantly, the lower-level one will be invisible.

Even in the case of one signal, it is necessary to manipulate two controls (sweep rate and gain) to achieve a usable display. And, when this display is obtained, it will look much like any other input signal with different frequency and amplitude parameters, since eye and CRT persistence eliminate the sensation of sweep speed at all but the lowest frequencies. By the way: I don't mean to denigrate the oscilloscope! It is one of the two most useful tools around (three, including the hammer), and we'd all be lost without it. I'm simply saying that it is far from the ideal instrument to employ to introduce the student to the physical world of sound. And, with the best will in the world, students who are not technically knowledgable, or inclined, will find oscilloscopic (or for that matter, drawn, chalk-scribbled, or even sky-written) time-versus-amplitude presentations intuitively dissatisfying from a musical point of view. Under such circumstances, the learning process is inhibited at best and frustrated at worst.

The spectrum analyzer suffers none of the disadvantages of the 'scope. Its frequency range is identical to that of the ear. Its displayable dynamic range, while typically less than that of the ear, is 10 to 1000 times greater than the 'scope. Several frequencies, of similar or differing amplitudes can be viewed simultaneously, and, in the case of the "real-time" analyzer, viewed dynamically as well. Finally, because all the information is displayed on the same, unchanging scale, it is not necessary to concern the student with necessary but spurious (insofar as understanding is concerned) time-and amplitude-scale changes.

Richard Factor is vice president of Eventide Clockworks Inc., New York.

In summary, the real-time spectrum analyzer is the ideal tool to use to introduce the concepts of frequency, amplitude, complex sound, and other aspects of sound perception to the audio engineer, and to the musician interested in some of the technical aspects of his art. Additional advantages include a method of simplifying the explanation of the notorious decibel! Unlike that of the cathode-ray oscilloscope, the analyzer display can be almost-instantly, and intuitively, understood by a wide cross-section of non-technical people.

I do not claim to be an educator, and my "methodology"—at least in terms of teaching procedures—will probably be faulty at best, but I shall attempt to show in this article some of the fundamental (and not-so-fundamental) audio concepts which can be explicated with the help of a computer/spectrum analyzer combination. In many cases, the computer portion of the combination will prove to be valuable, both for ancillary computations, and for clarifying some fairly-difficult concepts, such as equal-amplitude contours and the concept of "pink noise."

Before starting, I should offer a disclaimer. No specific programming examples are offered herein for an obvious reason: None have been written. Eventide Clock Works is *not* in the education business, and writing and testing software under realistic conditions takes a lot of time. Nonetheless, the examples given are not futuristic extrapolations of the art of programming. They are, rather, fairly simple and obvious combinations of techniques known to even the novice BASIC programmer. While the specific computer described here is the Commodore PET, other versions for the Apple and Radio Shack TRS-80 computers have virtually-identical capabilities.

THE REAL-TIME ANALYZER

A real-time spectrum analyzer is a device which gives, in most cases, a visual presentation of the amplitudes of the various frequency components present in a complex input signal. The horizontal axis represents the frequency of the component, and the vertical axis represents the amplitude of the component at that frequency. For most acoustical work, it is customary to use logarithmic axes, so that a wide range of frequencies can be accommodated on a single display. The "real-time" part of the description refers to the capability of the unit to fully analyze the input signal and present a new display frequently enough so that all components of the input signal are visible as they occur. Analyzers are also subdivided into bandwidth classifications. The most commonly-used are one-octave bandwidth and one-third-octave bandwidth. Subdividing the frequency range into octaves or one-third octaves is in keeping with the facts of psychoacoustics: The ear is generally sensitive to tones separated by percentages rather than by absolute amounts. At 100 Hz, a 10 Hz difference is instantly audible; at 10 kHz, it's almost unnoticeable. The audible frequency range is approximately ten octaves wide.

Eventide's analyzer circuit board.

The analyzer described here employs 31 filters to separate this range into individual bands, each of whose amplitude is measured and displayed on the CRT screen of a PET computer. The analyzer circuit board (manufactured by Eventide) is installed inside the computer and, for all practical purposes, becomes part of it. The computer CPU (Central Processing Unit) chip sends all the signals necessary to control the analyzer to it via an interface cable, and receives and processes the spectrum data over this same data path. It is interesting to note that there are NO operating controls on the Analyzer board. Everything is controlled by software written for the computer. The hierarchy of operation begins with the analyzer hardware: This comprises a variable-gain preamplifier stage, the 31 filters, 31 detectors, a multiplexer to sequentially sample the outputs of the detectors, and an analog-to-digital converter. A "Peripheral Interface Adaptor" (PIA) chip interfaces the CPU to the various control signals necessary to sequence the multiplexer, A/D converter, etc. The PIA also selects the preamplifier gain. The next step in the heirarchy is the "firmware." This is a set of "machine language" instructions necessary to perform the individual steps which operate the multiplexer, do mathematical operations such as logarithmic conversion, and rapidly generate the bargraph display. The instructions contained in the machine language ROM (Read Only Memory) are usually of little interest to the analyzer user. Each instruction does relatively little (such as load an 8-bit number in a register or perform a shift operation). However, these instructions are executed very rapidly (about 2 to 4 microseconds each), and they permit the analyzer to be operated about 100 faster than would an equivalent set of BASIC language instructions. This ROM also performs the function of linking the computer's operating system to the analyzer. Using instructions contained in the ROM, a complex operation such as drawing a bargraph is reduced to a single BASIC statement.

The final step in the hierarchy is the "application" program or the program written by the end user/purchaser which allows the analyzer to be used for something! The major application programs furnished by Eventide are the "Interactive Program," which in conjunction with a laminated overlay, turns the PET keyboard into a "control panel" for the analyzer, and the "Selftest Program," which, in conjunction with an audio oscillator allows the user to set up, calibrate, and verify

The third-octave real-time audio spectrum analyzer by Eventide Clock Works plugs right into the PET computer.

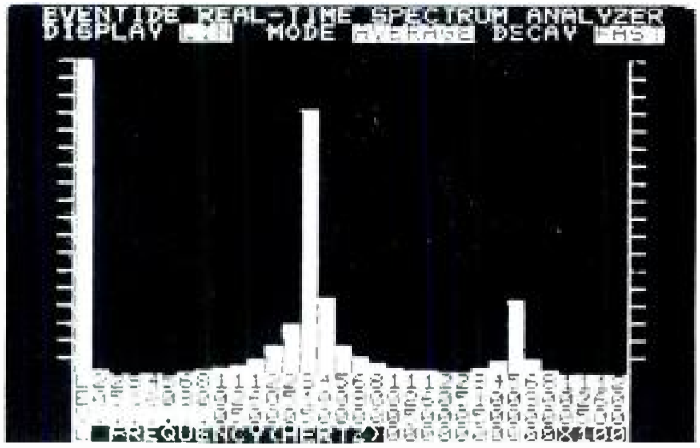

Figure 1. Sine waves at two frequencies and different levels applied to analyzer input. Characteristics of both signals may be determined instantly (and precise levels gotten from the computer for quantitative analysis. An oscilloscope display of the composite waveform would be difficult to interpret.

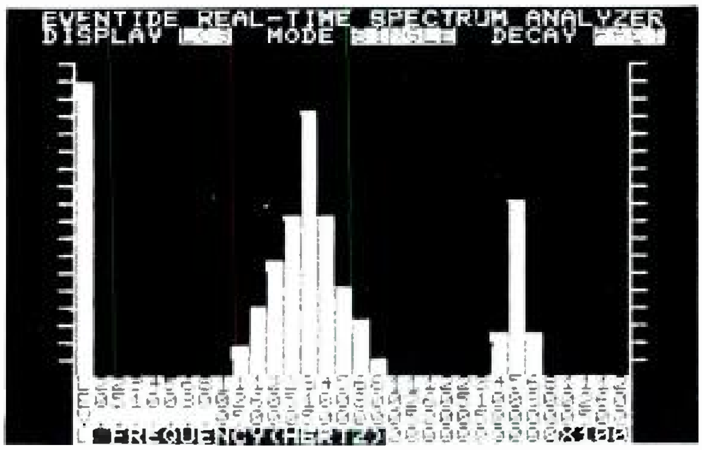

Figure 2. Same as Figure 1, except the display is logarithmic. Comparison between LIN and LOG displays, along with calculations based on the numerical values of the data help elucidate the meaning of decibel.

operation of the analyzer. Unlike the machine language program in ROM, application programs may be very unsophisticated. For instance, "operating" the analyzer in a single mode, so that the display continuously shows the input signal, requires a program only 4 statements long, two of which perform the analysis and two that are used for "housekeeping." At the other end of the scale, application programs may be quite complicated, including not only operation of the analyzer, but control of peripheral equipment, data storage on disc and cassette tape, and calculations involving data generated internally and externally. Most of the applications suggested in this article require application programs, either our "Interactive Program," or one written by the instructor. One could add that, depending upon the degree of "computer illiteracy" exhibited by the students, and their desire or requirement to combat it. It might also be advisable to suggest or assign programming problems which combine use of the spectrum analyzer and the computer.

"LESSONS"

While the analyzer computer may be used for all familiar applications and many unfamiliar ones, I am going to limit this discussion to a few examples which may be used to increase the student's knowledge and or aid his sonic intuition. In this highly-subjective field, it would be difficult to say which should take priority.

INTRODUCTION TO AUDIO CONCEPTS VIA THE ANALYZER

The perception of pitch and loudness may be related to the physical concepts of frequency and amplitude with the adjunct of a variable-frequency oscillator and attenuator set. Amongst the phenomena which can be demonstrated are:

The nature of musical octaves and harmonics: The tuning of the oscillator can be varied between, say 400 Hz and 800 Hz, and the fact that the peak of the bargraph has moved by 3 units should be noted. The student is then asked to adjust the oscillator to a frequency another octave higher, without watching the display. Most people will be able to come close without difficulty. Reference to the display will show a vital fact. The frequency has been multiplied by 2 (the bar above 1600 is now at a maximum), indicating that for the same perceived difference in sound, the absolute change has doubled. Repetition of this experiment with different starting frequencies will bring home the significance of percentages as opposed to absolutes in musical relationships. This simple, almost insignificant, demonstration is extremely effective in explaining this non-intuitive fact.

Another demonstration in this vein is a hands-on proof of the existence of "equal amplitude contours." Here we employ a (conceptual) application program for the PET.

This program will permit the PET to capture a frequency amplitude data point in response to pressing a key or an external switch closure. In operation, the instructor tunes the oscillator to the various filter center frequencies. (He, not the student, is watching the screen.) The student is then asked to adjust the attenuator until the tone sounds "as loud" as the previous tone. When the student is satisfied with the level, he may press a button (or nod OK if no external hardware is connected) and the instructor can press a key. The computer takes over now and stores both the frequency and the amplitude data automatically and instantaneously. The procedure is followed until the frequency becomes so high that it is no longer possible to achieve equal loudness without shattering windows. When the data set is complete, the analyzer firmware converts it to a bargraph showing the ear's frequency response throughout the audio range. It is only slightly-more-complicated to write a BASIC program to do this procedure with multiple students, saving all the data both individually and as a statistical summary. In fact, with a large enough group, it might be possible to get an indication of the decline in high frequency acuity with age. All of this data can be presented effectively (and even animated, if desired) on the computer screen. Data derived by experiment is usually more convincing than that which is read (or is the subject of lectures). The experiment might be effectively augmented with a presentation of the effects of listening to loud noise or music over long periods. Temporary threshold shift can be demonstrated by taking data before and after a practice mixing session. (The computer can store the first set of data on its cassette drive.)

I began the above paragraph by mentioning an "application program." The evolution of the paragraph simulates the evolution of most programs. You begin with a simple task (taking data for one student), and end up with a relatively complicated one, taking data from a group, performing statistical summaries, storing it, re-loading it, and comparing it with a new set of data to be taken! Fortunately, none of the steps is particularly difficult if attacked logically. Programs, like hardware, benefit from modular construction. Of course, once a set of teaching programs is developed, it may be reused indefinitely.

THE DECIBEL

Judging from the number of articles appearing on the subject, it would seem that the decibel is either the most misunderstood or most abstruse concept this side of relativistic time dilation.

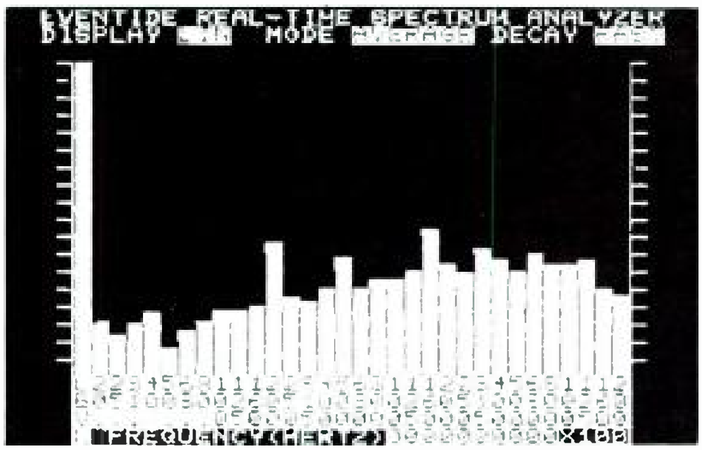

Figure 3. "Pink Noise" applied to the input of the analyzer. This ragged graph represents a "snapshot" or single analysis. Normally the eye integrates the data over readings to obtain a "flat" line.

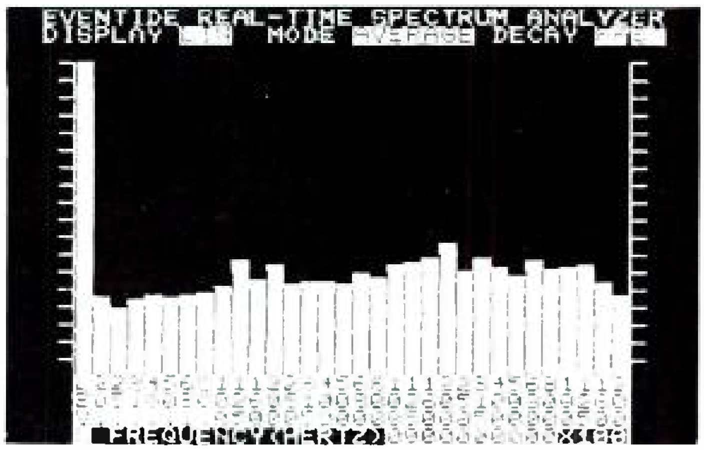

Figure 4. "Pink Noise" applied to the input of the analyzer results in a much flatter graph after multiple averages. Because the scale is linear, the actual roughness is less than ±1 dB.

Why this should be so I don't chose to guess, but the spectrum analyzer should make explanations a bit easier. The PET/analyzer combination has the capability of displaying a spectrum in either a linear or logarithmic mode (and of switching simultaneously between modes). With proper signal sources (such as two independent sine-wave sources, and perhaps a few noise sources as well), one can demonstrate such fundamentals as linear-versus-logarithmic voltage ratios and coherent and incoherent addition of signals. Remember that the mathematical capabilities of the computer allow the student to simultaneously experiment with real-world signals and perform calculations upon the data he measures and the results he derives.

Another subject (difficult, and with far-better reason), is the interaction between physical spaces filled with arbitrarily-shaped objects, and sound generated and reverberating within such spaces. Analytical solution of such problems is, with minor exceptions, impossible even with high-speed modern computers. Despite this fact, studio workers of all persuasions are called upon to function in this environment, and any steps that can be taken to aid them in developing their general understanding and intuition regarding everyday problems like microphone placement and the physical arrangement of sound sources must be classified as one of the most important tasks facing the instructor. To this end the spectrum analyzer, a noisy source, and a few application programs will find use.

ACOUSTIC MEASUREMENT

Before continuing, I should point out a few additional capabilities of the analyzer: built into the ROM firmware are the machine language routines necessary to do "ensemble averaging." This is a process by which the results from successive analyses are added together, frequency cell-by-frequency cell. The result of this process is to drastically reduce the statistical uncertainty of a measurement, by averaging the readings of many measurements. Because noise sources typically have significant variations in their low frequency output over short intervals, analyzer users are accustomed to unsteady displays in the lower octaves. While the eye and brain can integrate a moving display to a reasonable extent, it is certainly better to have actual numbers to work with and these the averaging process provides. The other capability is to substitute BASIC routines in certain cases for the machine language routines resident in the ROM. Thus, if the user does not wish to perform a complete spectrum analysis, he can write routines to analyze the results from a single filter or group of filters.

With this explanation out of the way, the procedures and experiments are left pretty much in the hands of the instructor. An obligatory one is "room tuning," in which the room is excited with pink noise and the "room curve" is observed on the analyzer screen. Of course, most classrooms will be very poor acoustically, but this doesn't affect the validity of the experiment. A quick variant to give insight into absorption is to save this curve in the computer, temporarily have everybody leave the room, and take another curve. The computer can then rapidly switch between curves while everybody tries to understand why they are different. If a graphic equalizer is available, it would also be instructive to attempt to flatten the room curve. Of course, the room may be so poorly designed that the attempt will fail, or, if it succeeds, the "equalized" room may well sound worse than when unequalized. This brings home quite quickly the importance of having proper physical design for acoustic spaces.

The previous experiment used the averaging capability to increase the accuracy of the measurement. The next uses the capability for selective analysis to perform an amplitude-versus-time measurement at a given frequency.

REVERBERATION MEASUREMENTS

The PET computer has a built-in clock (which measures in "jiffies," 60 to the second). This allows programs to determine how long a particular procedure has taken, to print out the time, etc. An always-interesting characteristic of acoustic spaces is the reverberation time or the length of time required for a sound impulse to decay to a fraction (usually -60dB) of its original value. Even more interesting is this characteristic in individual frequency bands, since the capability to vary these parameters leads to a more "natural" sounding reverb than does, say, a long reverb time due to a horrendous resonance at 500 Hz! The PET/analyzer combination permits a complete set of such measurements to be made automatically, even to the extent of controlling the sound source and printing the results. In the teaching environment, this allows rapid experimentation with different techniques for reducing (or creating) resonances, and giving the student a feel for the characteristics of different spaces, such as "live" and "dead" portions of rooms.

There are many other instructional applications of the real-time analyzer, and the computer itself is becoming more nearly vital as a teaching tool. Combining the two instruments provides an opportunity to familiarize the student with many new—and some, fairly difficult—concepts effectively. The computational capabilities of the analyzer permit data to be presented in a manner that aids intuition and auditory experience and thus provides immediate reinforcement of lessons that might otherwise be lost. The complexity of interrelating the audio and video inputs to the human brain is substantially reduced by using a computing spectrum analyzer, and this aids the student in integrating new information for his future use. ■

DAVID L. HAYNES & JOHN M. BRENNAN

A New Digital Delay System

Recent advances in digital technology now make it practical to use digital delay lines in primary-channel applications.

THE SUCCESSFUL CREATION of a master disc from a carefully-produced master tape is a key process in assuring the overall quality of the final product, the phonograph record. A great deal of specialized equipment is necessary to insure a high-quality master disc; among this equipment is the variable-pitch lathe.

A variable-pitch lathe has a mechanism for varying the groove spacing for the cutting head. Using this type of lathe, the pitch becomes dependent upon the character of the program material. A variable-pitch lathe requires two audio signals: the original "preview" signal, sent to a computer which controls the pitch-adjusting servo system, and, a delayed "program" version of the original signal, actually cut onto the master disc. The purpose of the delay is to allow the pitch to be properly adjusted by the time the program signal reaches the cutting head. FIGURE 2 shows a block diagram of a typical variable-pitch disc mastering system.

This variable-pitch feature allows an increase in the program length, and/or the dynamic range, that can be put onto a disc. Since the preview signal controls the pitch (lines per inch) of the grooves engraved by the cutting head, groove spacing becomes a function of the amplitude and dynamics of the source material. For low-level passages, the pitch controller will make the grooves closer together. For higher-level passages, the grooves will be farther apart. This allows optimum packing of program material onto a record side. Variable-pitch also provides another 2-to-3 dB "hotter" recording level—thus, dynamic range—than would be possible for the same amount of program material using a constant-pitch lathe.

THE OLDEN DAYS

The variable-pitch lathe is by no means a new device. In the late 1940's, custom variable-pitch adapters could be seen in some mastering suites. Around 1951, the Reeves-Fairchild margin control—an adapter usually fitted to Scully lathes—was marketed. This Scully/Reeves-Fairchild set-up became the first widely used variable-pitch lathe. To produce the required delay, several methods have been used. The most common method used today is a special tape recorder with two playback heads. Between the two heads, a set of tape guides provide a path for a tape loop. If one were cutting an LP lacquer using a 30 in/sec. master tape and the Scully lathe (which requires one revolution of delay), 54 inches of tape would be required between the two playback heads. Complicated guide schemes required for this excess amount of tape open the door for possible degradation of wow and flutter characteristics. Also, most preview transports do not accommodate the 14 inch reels which may be required at 30 in/sec. As a result, slower tape speeds are often used for the master tapes.

Of course, direct-to-disc recordings (by definition) cannot use this method, therefore a highly experienced lathe operator must be present at all times. By reading the score he anticipates program content, and manually adjusts the lathe pitch. This can be a tricky situation because if he guesses wrong an overcut can occur and the session must be re-done.

The advent of digital recording has created its own problems. Because of the intimate head-to-tape contact required, complex guide schemes, as mentioned above, are very difficult to implement. Therefore, one solution currently used is to record the program material onto two channels of a four-channel digital recorder. The program is also stored in a computer, read out the required delay time later, and recorded onto the other two channels. This method of preview requires a knowledge of the type of lathe to be used later, and is only valid for one cutting speed. In other words, a single four channel tape could not be used to cut 33 and 45 rpm versions of the same tune.

WHY ANOTHER DIGITAL DELAY UNIT?

It became apparent that a more-versatile form of previewing for disc mastering could be developed. At Ampex, after reviewing the economic trends of digital hardware, and the company's own research into digital audio technology, it was felt that an ideal solution would be a digital delay system, providing accurate analog-to-digital-to-analog conversion, appropriate delay, and optional digital inputs and outputs.

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Seated at a Neve disc-mastering console at Filmways/ Heider studios, engineer Phil Cross oversees a tape-to-disc mastering session. Between the lathe and an Ampex ATR-100 is the new Ampex ADD-1 digital delay system.

There are many types of digital delay units currently available. These are secondary-channel systems—excellent for providing room ambience enhancement and for adding special effects to specific recorded tracks. However, a primary-channel delay, which can be inserted into the main audio line, requires electronics comparable to that of a state-of-the-art digital tape recorder.

For a system like the Ampex Digital Delay, degradation of the signal is not dependent upon the delay time (for the same reason that the degradation in a digital tape recorder is not dependent upon the tape). Once the signal has been digitized, the length of the delay depends only upon the amount of memory available to store the digital samples. This type of system is also appealing from a standpoint of versatility, because it is compatible with any type of audio source. This means that the ½-inch two-channel tape format, as well as direct-to-disc albums, can take full advantage of the added dynamic range and increased program material possible with a variable-pitch lathe. The delay required for preview using a half-speed mastering system is also programmable and a real-time bandwidth of 20 kHz or 40 kHz is possible.

WHAT'S INSIDE?

FIGURE 3 shows a block diagram of the ADD-1. Analog signals enter by way of a balanced line input where they go through an input low-pass (anti-alias) filter, a sample-and-hold, and an analog-to-digital (A/D) converter, where the signal is digitized. The digital representation of the sample is then stored in memory for the desired length of time, recalled, and passed through a digital-to-analog (D/A) converter, a "de-glitcher," and an output low-pass (reconstruction, or anti-imaging) filter. The "system control" oversees the system and generates the timing as well as doing the memory management.

Because the ADD-1 is a digital system requiring A/D conversion, the analog signal must be sampled, and then held for a finite amount of time. This hold-time must be long enough to allow the A/D converter to complete its conversion. The sample-and-hold can be considered to be a simple analog multiplier. As such, the original frequency, the sample frequency, its multiples, and all the sums and differences would be present at the output of the sample-and-hold. As an example, if a 10 kHz sine wave is sampled at 50 kHz, signals at 10 kHz, 40 kHz, 50 kHz, 60 kHz, 90 kHz, 100 kHz, 110 kHz, etc. will all be generated. This is shown in FIGURE 4. The frequency spectrum of a passage of music before and after sampling is shown in FIGURE 5. FIGURE 6 shows the time-versus-amplitude response of the same passage of music. Because of the lowest-order difference frequency (sample frequency minus original frequency) generated by the sampling process, no frequency applied to the sample-and-hold should exceed half the sampling frequency. Otherwise a false component, known as an alias frequency, will be generated. Thus, an input low-pass filter, usually called an anti-alias filter, is required. This type of filter can have a profound effect upon the overall sound of the system, and an appropriate compromise between aliasing, group delay distortion, harmonic distortion and noise generation must be made. The above distortions are directly related to the cut-off slope of the filter. Because of this, the sampling frequency generated by the internal master clock was chosen to be 50 kHz. This allows the band between 20 kHz and 25 kHz to be used for a reasonable filter roll-off.

The sample-and-hold, being an analog device, does not quantize the audio signal. The value it holds is dependent only upon the amplitude of the signal at the instant it is sampled.

The A/D converter has a finite amount of numbers which it can choose to assign to that value. In the case of a 16-bit converter (the type used in the ADD-1), the amount is 65536(2^{16}). The process of the A/D converter choosing the value closest to the one held by the sample-and-hold, and giving it a binary representation, is known as quantizing.

Upon completion of the A/D conversion process, the digital representation of this sample is stored in a memory location. The memory is addressed in sequential order, starting at the beginning of memory (location 0), and extending to the memory location that corresponds to the desired time delay. Therefore, if one second of delay is desired (using a sample frequency of 50 kHz), samples would be stored in memory starting at location 0 and extending to location 49,999. Once the final location is reached, the process is repeated. When a location is addressed, the sample stored in the memory location is sent to the D/A converter, and a new sample from the A/D converter is stored in the same memory location. Thus, the amount of delay is determined by how much of the available

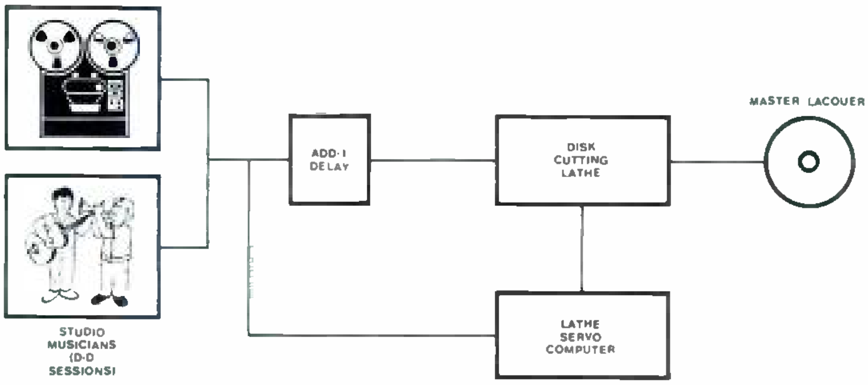


Figure 2. Disk Mastering system block diagram.

memory is used. After the desired time delay, the sample is recalled from its memory location and sent to the D/A converter.

"DE-GLITCHING"

During the conversion process, switching transients can occur in the output D/A during the settling time. These transients, or glitches, are removed by the de-glitcher. The de-glitcher is similar to a sample-and-hold, since it must acquire a sample from the D/A, after the D/A has settled, and hold it until the next sample is available. Because the output of the de-glitcher is exposed to the analog output during those sample-and-hold times, the de-glitcher must be fast enough not to induce slew-limit distortions, while at the same time must be quiet enough to preserve the dynamic range of a 16-bit digital system. After the signal is de-glitched, the higher-frequency components shown earlier are still present. These high-frequency components must be removed, for they may cause distortions in subsequent analog equipment due to possible slew-rate limitations. Therefore, an anti-image filter with a cut-off at 20 kHz is used after the de-glitcher. Finally, the de-glitched, filtered signal appears at the system output, as a delayed replica of the input signal.

The system control is the coordinator of the entire process. It is a micro-programmed controller, similar to the type used in many modern computers. The function of the system control is to generate the appropriate timing for the system, manage the memory and oversee the distribution of data within the system. This circuitry also contains the information about how much delay the user requires and uses the memory accordingly. The master clock generates the signals which provide the timing for the digital logic in the system and also generates the sampling clock (which is crystal referenced).

For applications where a sample clock other than that internally generated is desired, the ADD-1 provides an external sample rate input jack. The electronics in the system will automatically synchronize with this clock and adapt the system to maintain and display the correct amount of delay.

The system can also interface with other digital audio equipment via a serial digital input and output. This allows digital data (from, say, a digital tape recorder) to be delayed without having to go through an extra conversion process.

HOW GOOD IS IT?

Whenever an analog signal is sampled and converted into a digital bit stream, two types of distortions occur. The first is dependent upon how many samples are taken per second. This sets an upper frequency limit for the signals which can be accurately reproduced. The other type of distortion consists of how fine the amplitude of the signal is divided or quantized. In a properly-designed system, this determines the dynamic range.

Other types of distortion occur in less-than-ideal systems. Our old friends, harmonic and I.M. distortion are still present. There are, however, some new ones to be on the lookout for. One of these is non-linear phase shift with frequency, commonly known as group delay distortion. The amount and character of the phase shift varies with the type of filters being used.

Another distortion occurs during the sampling process. Frequency components which are seemingly unrelated to the original frequency appear. These so-called convolved frequency components come from harmonic distortion in the sample-and-hold and de-glitcher, which are aliased down into the audio band. (Remember, the sample-and-hold and de-glitcher come after the anti-aliasing filter.)

FIGURE 7 shows the specifications for dynamic range and total distortion-plus-noise for an ideal digital system. The dynamic range of any n-bit system can be determined by the following equation:

$$X\ dB = (20 \log 2^n) + C$$

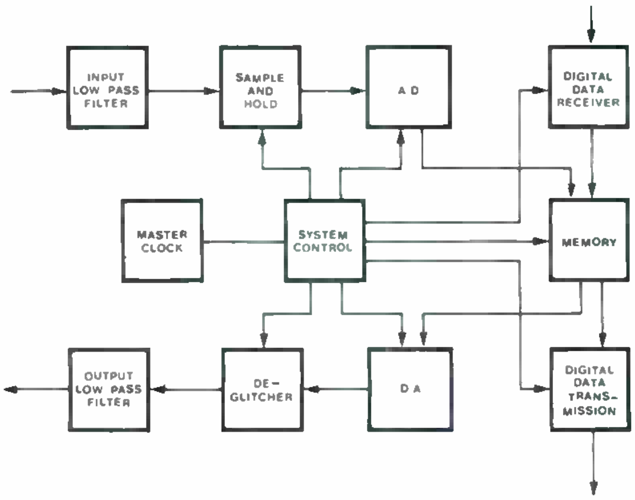
C is related to the power distribution of the signal, the bandwidth of the filters and the sample frequency of the system. For most audio systems, this number is between 2 and 4.

Because of present technological limitations, digital systems available today do not meet these theoretically-ideal specifications. Therefore, any digital device to be used within a primary channel should be very-carefully investigated.

OTHER APPLICATIONS

Of course, the Ampex Digital Delay can be used for any application where a high-quality delayed signal is necessary. Truly-transparent delays can be used to provide the acoustical time difference necessary for a properly-designed P.A. system for large auditoriums or concert halls. Slap forward, unattainable without degradation before, is now easily possible.

Figure 3. ADD-1 system block diagram.



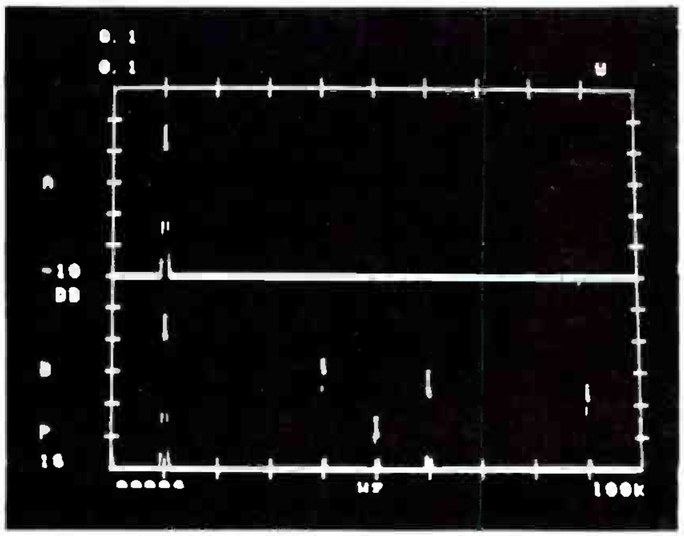


Figure 4. (A) Frequency spectrum of 10 kHz before being sampled. (B) Frequency spectrum of 10 kHz after sampling.

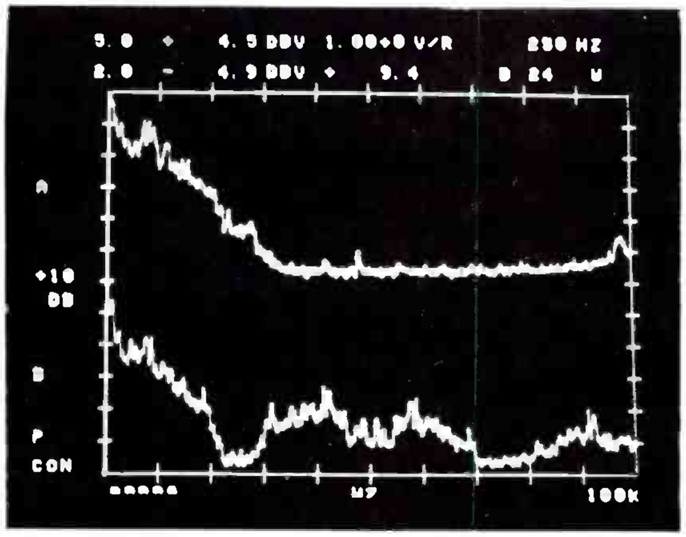
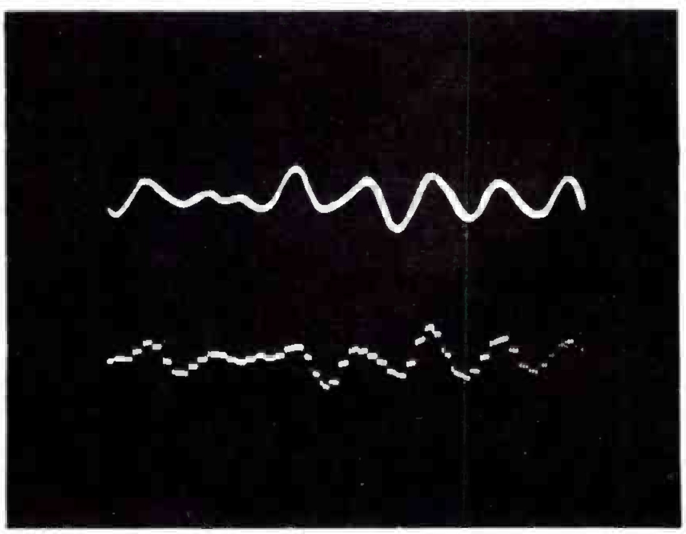


Figure 5. (A) Frequency spectrum of a typical musical passage. (B) The same passage after being filtered and sampled.

Figure 6. The top trace shows a passage of music before sampling, the bottom trace shows a similar passage after being sampled.



NUMBER OF BITS PER SAMPLE	DYNAMIC RANGE (dB)	DISTORTION & NOISE (%)
8	51	.28
9	57	.14
10	63	.070
11	69	.035
12	75	.018
13	81	.006
14	87	.0045
15	93	.0022
16	99	.0011

Figure 7. Table showing theoretical dynamic range and distortion and noise for a given amount of bits.

SUMMARY

Digital technology is becoming increasingly important in the field of professional audio equipment. As the price of digital hardware drops, and the speed and precision of A/D and D/A converters increase, more and more primary-channel digital equipment will appear on the professional audio scene. The ADD-1 is one such application of digital technology for primary-channel audio use.

The ADD-1 will bring new versatility into cutting suites equipped with variable-pitch lathes, allowing any audio source to be cut onto a master disc, while being able to take advantage of the added packing density (or dynamic range) that the variable-pitch lathe can provide. ■

DAVE RUBENSTEIN

Understanding Magnetic Tape Specifications—Part II

A review of the electro-acoustic properties of magnetic tape—how they are derived, and what they represent in terms of tape evaluation and performance.

When comparing one tape with another it is important to compare them under the same circumstances. In other words, you must compare apples with apples, not apples with oranges. Simply stated, the tape speed must be the same, the record track width the same, the record level must be the same, and the equalization must be the same. Only when these four factors are equal can one compare measurement data. Furthermore, in order to properly test any tape, first make sure that the machine is optimized for that tape, starting with the bias.

In order to bias a machine, one must look at the specifications given by the manufacturer as to the sensitivity reduction of a given signal. In the case of the graph in Figure 1, that signal is 10 kHz (at 15 in/sec.) The method for biasing a tape is to feed the required signal into the machine at a low level, place the machine into record and monitor the playback on the VU meter. Increase the bias until a maximum level is achieved on the meter; then increase the bias further until the reduction of the applied signal is equal to the specifications supplied by the manufacturer. This method of sensitivity reduction to arrive at the proper amount of bias current can be visually depicted on the graph. Looking at the E 10 kHz curve, find the peak and then—given the specification of a recommended 3½ dB sensitivity reduction—count this amount downwards from the peak, and arrive at +2 dB of bias current, which is shown at the bottom of the graph. It is important to note that sensitivity reduction is the fall-off in frequency response of the tape, *not* an actual reduction in bias current. For example, when someone comments that one tape requires 1 dB more of bias than another, they are speaking of increasing the bias current, not of reducing the applied signal level to achieve the proper bias setting.

The second specification normally given is the deviation of the recommended bias from the bias setting of a DIN *(Deutsche Industrie Normen)* reference tape. This German Industrial Standard is published by the German Standards Committee *(DNA, Deutscher Normen-ausschus),* and is used as a worldwide zero-reference point. The DIN reference tape is PER 525, manufactured by Agfa-Gevaert AG. The DIN committee has defined this tape as having zero (0) reference bias. Therefore, all other mastering tapes are related to this tape, and the graph shown in Figure 1 depicts a tape with a bias of +2 dB over the DIN standard.

At this point, it is important to note that for any given tape there is a manufacturer-recommended bias, and that no matter

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what speed or what operating level the consumer chooses to use, the amount of bias current is always the same; only the method to derive the amount of bias is varied. It is also important to note that the width of the record gap is also a factor which will vary the method used to derive the proper amount of bias.

Due to an effect known as self-erasure, the high-end response decreases as the bias increases. (The bias current begins to erase the tape—high frequency first.) The narrower the record gap, the greater the sensitivity reduction necessary to achieve the proper bias. The wider the record gap, the less sensitivity reduction necessary. To state it more fundamentally, a record head with a narrow gap is more efficient than one with a wider gap.

The next factor, and an extremely important one at that, is third harmonic distortion (thd). It is given as a figure in percentage of the total signal. Third harmonic distortion, as we discussed previously, is bias-dependent. The greater the bias, the lower the distortion (up to a certain point). After that point, the law of diminishing returns takes place, and distortion once again begins to increase. Hence, careful setting of the bias, for a low point in the distortion, is very important. On the graph the curve thd = 320 is the curve for low frequency distortion. The measurement can be read on the vertical percentile line to the left of the graph. Looking at the graph, we notice that the minimum point on the distortion curve is the exact point of 2 dB of bias current. Therefore, an alternate method of biasing the tape (if one had the proper equipment to measure distortion), would be to bias for minimum distortion. The percentage distortion at the proper bias point is 0.2 percent; an excellent distortion figure.

The next point to be looked at on the graph is the curve marked MOL (thd = 3%). This stands for Maximum Output Level (the highest output one could receive from the tape, given a maximum of three percent third harmonic distortion using a 1 kHz tone. The point of maximum output level is also bias-dependent. In the case of this graph, the maximum output level is more-or-less at the optimum point of bias. Note that at the bias where you found the minimum thd, the MOL curve for 3% thd flattens out, and the 1% MOL curve reaches its maximum.

If the MOL 3% (or, sometimes 5%) thd curve reaches maximum and then drops more than 1 dB without rising again, you can assume that the record head is saturated by bias and record current, and you are measuring the distortion of the head.

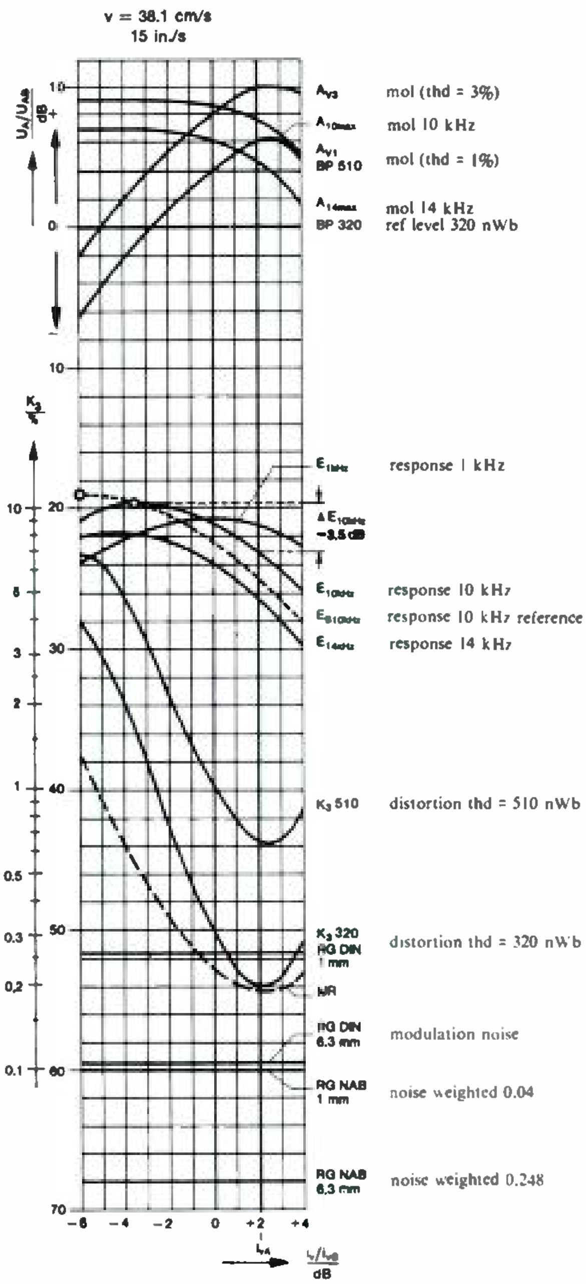
The MOL 1% thd curve must follow the thd curve at the reference level; that means when thd at reference level shows a minimum, MOL 1% thd has its maximum. One can determine the maximum output level by counting how many dB above the reference level the graph depicts. In order to fully understand maximum output level, one must first understand the term reference level.

The reference level represents a working point so that the user can receive the most output from the tape and still keep distortion at a practical minimum. Or, to coin a phrase, "the hotter, the better." However, if the record level gets *too* hot, the distortion increases due to non-linearities in the remanence curve (which was discussed in Part I). Hence, reference level and record level are synonymous, and the tape manufacturer will recommend a reference level for the given tape. Therefore, when we compare maximum output level from one tape to another, make sure that the two tapes are measured from the same reference level. In the graph provided, the maximum output level at the point of optimum bias is +10 dB (for 3% thd). According to the graph, the reference level is 320 nanowebers-per-meter (nWb/m). A nanoweber is a measurement of magnetic fluxivity. To put it more plainly, it means that one would set the record level high enough to achieve an output of 320 nWB reference fluxivity from the tape (fixed with a reference tape).

The maximum output level at 3% thd is measured with a 1 kHz tone. If the maximum output level was measured at a different frequency, another result would occur. Therefore, an industry standard specifies that maximum output level at 3 per cent is always measured at 1 kHz.

The next two points deal with relative sensitivity and three different frequencies; 1 kHz, 10 kHz, and 14 kHz. Relative sensitivity is measured with a given frequency with a constant record current which results in a replay voltage 20 dB below reference level. In simpler terms, a relative sensitivity is a measurement to determine if the tape truly puts out what is fed in. As you can see by the graph, sensitivity is extremely bias-dependent. Earlier, we used the sensitivity curve of 10 kHz to bias the machine. We can now look at the difference between the sensitivity curve of a 1 kHz tone and those of the 10 and 14 kHz tones. Until recently, some tape manufacturers have recommended that their tape be biased with a 1 kHz tone. In reality, this is not very practical, because the slope of the 1 kHz tone curve is not very steep, and therefore it is difficult to find the peak on a VU meter. However, when we use a 10 kHz tone, the slope is steep and the peak can be found easily. Hence, with a 10 kHz tone, we can bias a machine with greater accuracy. Relative sensitivity of the three tones previously mentioned can be seen on the graph as E 1, E 10 and E 14 kHz. Many times a

Figure 1. Graphic representation of electro-acoustic properties of magnetic tape.



tape manufacturer will give a sensitivity specification relative to a reference tape. This is perfectly acceptable if we know the sensitivity of the reference tape. (If all tape manufacturers used the same reference tape, and the test methods were identical, the relative sensitivity measurements could be directly compared.) One can see from the data that, at certain frequencies, the tape has a higher sensitivity than the reference tape, and at other frequencies it has a lower sensitivity. As far as sensitivity is concerned, the higher, the better. One can see by the sensitivity curves that the bias adjustment can greatly affect the output of the tape. This is another point to make in favor of proper bias adjustment. If the bias is incorrectly adjusted, further equalization may be necessary, therefore adding more noise.

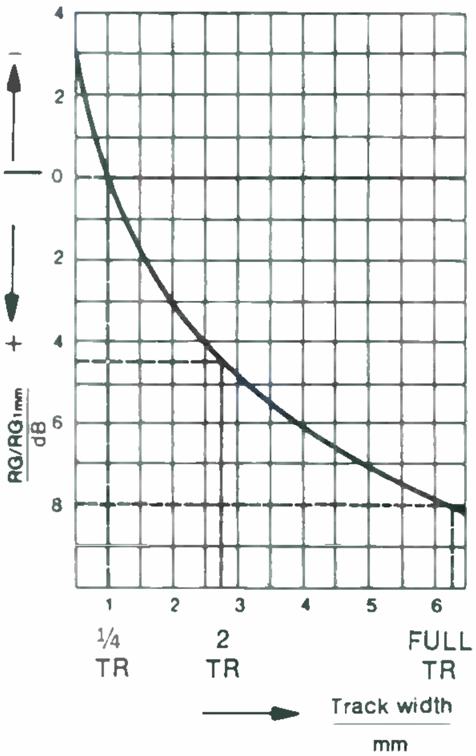
To delve a little further into what really happens on the tape to produce a sensitivity specification, let's consider three different parameters. The first is the quantity of magnetic particles within the same area of the tape. The greater the number of particles in the area (track width times coating thickness), the greater the low frequency output will be. (Shorter wave lengths do not contribute to the output from the depth of the coating; only longer wave lengths will add to the output from the lower sections of the coating.)

The second parameter is the squareness factor, which was discussed earlier in the section on hysteresis curves. This depends on the orientation of the oxide particles. If all the particles have different orientations, the squareness factor is approximately 50 per cent. The remanent magnetization is only 50 per cent of saturation. Modern tapes have a squareness factor of approximately 80 to 90 per cent of saturation.

The third and final parameter is surface roughness. This factor greatly influences short wave length (high frequency) response, due to distance attenuation, which was also discussed previously.

The curve right below the third harmonic distortion curve is the curve of modulation noise. This curve is also extremely bias-dependent, as is depicted on the graph. If a tape is under-or over-biased even as little as 2 dB, the modulation noise becomes unbearable. Another method of biasing the tape is to bias for the minimum point of modulation noise. According to the graph, the minimum point of modulation noise is the exact point of optimum bias at 15 in/sec. This biasing technique works for 15 in/sec., but not at any other speeds. It is also dependent upon the record head. The faster a tape runs, the greater the modulation noise, due to the speed at which the imperfections in coating pass across the head gap.

The next topic concerns a property of magnetic recording which is one of the greatest deterrents of an analog system. We are, of course, speaking of random noise—the swish or hiss that is heard underneath the program material. Noise is created from random charges of the particles on the tape. The size and shape of a particle can have a large effect on the noise. The smaller the particles, the lower the noise; however, print-through becomes worse as particles get smaller. It is extremely important that one records at a fairly high level in order to magnetize as many particles as possible in the oxide layer. If many particles are left untouched by the magnetic field from the gap, the noise will be that much greater to the ear. The more particles the magnetic field influences, the more the noise is masked. The noise, as one can see from the graph, is not bias dependent and is a property of the given tape. The noise is, however, track-width dependent. As you increase the track width from 1 mm. to 2 mm. the noise voltage increases by 3 dB. But the output voltage doubles, because twice the number of particles pass across the head. Hence, the output voltage rises by a factor of 2, which equals 6 dB. In other words, the output voltage is twice as high, while the noise is increased by only one-half as much. Therefore, the wider the track width, the better. Hence, a half-track machine will have a better dynamic range than a quarter-track machine. Furthermore, a 16-track



The signal to noise ratio of a magnetic recording is dependent, amongst other things, upon the track width. This interdependence is figured in the diagram opposite.
The signal to noise values stated in this data sheet are given for track widths of .248″ and .04″ respectively. In practice however other track widths are used.
In order to determine the S/N value, one must add the value obtained from this diagram to the S/N value for 1 mm of track width.
Should a signal to noise ratio of a desired output be specified, the difference in level relative to the DIN reference level should be taken into account.
Signal to noise relative to maximum output with a distortion level of $K_3 =$ 3% (RGA) is specified according to DIN 45 512, sheet 2.

Figure 2. Variation of signal-to-noise ratio in relation to track width.

machine using 2 inch tape would have a better noise characteristic than a 24 track machine using 2 inch tape due to the fact that each track width is wider.

FIGURE 2 depicts the relationship between track width and noise. Utilizing this graph, we can determine the noise of a given tape in conjunction with the track width that the machine is working with. Shown on the graph are the European standard track widths for quarter-, half-, and full-track. The values obtained from this graph must be added to the value obtained for the noise of a 1 mm. track width.

SIGNAL-TO-NOISE RATIO

After covering the subject of noise, the next obvious specification to be dealt with is the term signal-to-noise ratio, or dynamic range. Dynamic range is a simple calculation, made by adding the weighted noise characteristic to the maximum output level at three per cent third harmonic distortion. Therefore, at 15 in sec., if the noise is 68 dB and the maximum output level is +10 dB, the signal-to-noise ratio is 78 dB. This can be calculated using the figures supplied or by counting dB on the graph. Signal-to-noise ratio or dynamic range is the range in which one can record, from the quietest, lowest signal to the loudest, highest signal. Below the noise floor one could not hear what was recorded and above the MOL peak, the signal would be distorted. Therefore, the signal-to-noise ratio expresses the effective working range of a given tape. The wider the ratio, the better, because one could record at a higher level with low distortion and record tones at a lower level and they would still be discernable above the noise floor.

PRINT-THROUGH

Print-through has plagued recording engineers ever since the onset of magnetic tape recording. Print-through is the pre- or post-echo heard before and after a recorded tone. It occurs after a recording has been made and the tape is wound on a reel. Each layer of tape has adjacent layers, full of tiny magnetic particles, which act upon the particles in the neighboring layers. As was discussed earlier, the particles on the tape are not all of the same size. Larger-size particles have greater coercivity than the smaller ones. Therefore, if you record a tone on the tape and the adjacent layers are blank, the larger particles on the recorded portion of the tape will influence the smaller particles on the unrecorded section of the tape, creating a similar magnetic pattern on the adjacent layers. These similar charges result in a tone exactly the same frequency as the one originally recorded, only at a much lower level.

Print-through may be heard on recorded, as well as unrecorded, segments of tape. Usually, it is the pre-echo that is heard and it sounds as if, for example, the downbeat of the orchestra occurs very softly, shortly before the real downbeat. The loudest print-through occurs on that segment of tape whose magnetic coating comes in direct contact with the loud segment of recorded tape. On standard-winding transports, this means the greatest print-through radiates outward, (away from the center of the reel). Tapes stored tails-out will therefore have greater post-print. Heads-out storage will produce greater pre-print. The difference between the two is approximately 1-to-2 dB. Print-through is wave-length dependent, and is obviously an unwanted characteristic. There are certain ways of reducing its effects. Because the printed tone is recorded without a strong

Figure 3. A typical magnetic tape specification sheet. Represented are the electro-acoustic and magnetic properties for Agfa-Gevaert PEM 468 Mastertape.

TECHNICAL DATA

ELECTRO-ACOUSTIC PROPERTIES		Unit		
MEASURING CONDITIONS:				
Tape speed		ips	30	15
Track width		mm	6.3	6.3
Tape flux per 1 mm track width for reference level		pWb	320	320
1. Sensitivity reduction for recommended bias	ΔE_{10} kHz ΔE_{14} kHz	dB	2	3½
2. Deviation of the recommended bias from the bias setting of the DIN reference tape	ivA/ivB	dB	+2	+2
3. Third harmonic distortion	K_3	%	0.15	0.2
4. Maximum output level (THD = 3%)	Av_3	dB	+10	+10
5. Relative sensitivity at 1 kHz	E_1 kHz	dB	1.5	1
6. Relative sensitivity at 10 kHz Relative sensitivity at 14 kHz	E_{10} kHz E_{14} kHz	dB	- 0.5	+ 0.5
7. Maximum output level at 10 kHz Maximum output level at 14 kHz	A_{10} max A_{14} max	dB dB	+2	+ 0.5 + 0.5
8. Modulation noise.	MR	dB	51	54
9. Noise weighted	RG	dB	66 5	68
10. Signal to noise weighted	RGA	dB	76.5	78
11. Print-through			60	58
12. Signal to erase ratio			70	
13. Uniformity at 1 kHz a. within a reel b. roll to roll			 $\leq \pm 0.25$ $\leq \pm 0.5$	
MECHANICAL PROPERTIES				
14. Coercivity	iHc	Oe	380	
15. Retentivity	Brs	G	1060	
16. Residual saturation flux per 1 mm	ϕ rs	pWb	1700	
17. Surface resistance of back coating		Ω	$< 10^6$	

TEST METHODS AND DEFINITIONS

1. SENSITIVITY REDUCTION FOR RECOMMENDED BIAS SETTING USING RECORD HEAD OF 7 MICRON (APPROX. 0.3 MIL) GAP LENGTH

Alteration of the high frequency bias causes distortion and modulation noise to approach their minima at nearly the same bias. The amount of that bias is dependent on the gap of the record head.

Sensitivity reduction for this tape under the defined conditions is

$$\Delta E_{10} \text{ kHz} = 3.5 \text{ dB at } v = (15 \text{ in/s})$$
$$\Delta E_{14} \text{ kHz} = 2 \text{ dB at } v = (30 \text{ in/s})$$

Deviations of the stated figures are possible and depend on the type of recorder and record head. We therefore recommend the adoption of a THD—or modulation noise minimum setting on each machine type to obtain the correct bias setting.

2. DEVIATION OF THE RECOMMENDED BIAS FROM THE BIAS SETTING OF THE UNRECORDED SECTION OF THE DIN REFERENCE TAPE (IN THE DIAGRAMS = 0 db)

3. THIRD HARMONIC DISTORTION

is the ratio between replay voltages of the third harmonic and the basic frequency when modulated to reference level. Even numbered harmonics are insignificantly small in magnetic sound recording if the high frequency erase and bias currents are symmetrical and if no DC magnetism of the tape results due to remanence of the magnetic heads and tape guides. Higher odd numbered harmonics are also insignificant.

4. MAXIMUM OUTPUT LEVEL AT 3%

is defined as the ratio of the replay voltage with a 3% harmonic distortion to the replay voltage of the reference level at 1 kHz.

5. SENSITIVITY AT 1 kHz

Sensitivity is achieved with a constant record current which results in a replay voltage 20 db below reference level derived from the unrecorded section of the DIN reference tape at a bias setting according to DIN 45 512, sheet 2. Relative sensitivity is related to the unrecorded section of the reference tape.

6. SENSITIVITY AT 10 kHz

This is the measurement at short wave lengths and is determined by means of a constant record current which has the same value as that used in paragraph 5. Relative sensitivity is related to the unrecorded section of the reference tape. The sensitivity $E_{8\ 10}$ kHz is derived under the same conditions as for E_{10} kHz from the unrecorded section of the DIN reference tape. It is used for the definition of bias setting of this reference tape.

7. MAXIMUM OUTPUT LEVEL

The maximum output level is the maximum replay level at 10 kHz (14 kHz) related to the corresponding value of the unrecorded section of the reference tape.

8. MODULATION NOISE ACCORDING TO DIN 45 519, SHEET 2

is determined as the noise voltage of a DC recording the strength of which is equal to the effective value of the current required to obtain the reference level. This noise voltage is measured with a volt meter and a masking effect filter (see DIN 45 519 — in preparation — and DIN 45 405). The modulation noise is related to the reference level.

9. NOISE, WEIGHTED, RELATED TO REFERENCE LEVEL

The noise of a well erased symmetrically biased tape is measured according to NAB standard (r.m.s. value and psophometric filter) and related to the reference level.

10. SIGNAL TO NOISE RATIO, WEIGHTED, RELATED TO MAXIMUM OUTPUT LEVEL (DYNAMIC RANGE)

is the ratio of the weighted noise level related to maximum output level measured according to above.

11. PRINT-THROUGH

is the ratio of the replay voltage of a signal recorded at reference level to the signal printed on the adjacent layer of the tape after storage for 24 hours at 20°C (DIN 45 519, sheet 1).

12. SIGNAL TO ERASE RATIO

is the ratio of replay voltage of a signal recorded at reference level to the residual signal after erase.

13. UNIFORMITY AT 1 kHz

Level fluctuations and variations in sensitivity from tape to tape are kept as low as possible by means of modern production techniques.

14. COERCIVITY

Coercivity is the measure of the magnetic field strength which is required to achieve a zero magnetization value.

15. RETENTIVITY

is the remaining flux density after saturation of the tape.

16. RESIDUAL SATURATION FLUX PER 1 MM

is the flux density at saturation multiplied by the cross section of the coating on a tape 1 mm wide.

17. SURFACE RESISTANCE

The length of the measured sample is equal to the width of the tape (DIN 45 512, sheet 1).

magnetic field (bias) and the remanent magnetism is very low, the printed tone is not stable. If one runs the tape over a sharp edge, around tape guides, or if the tape is subjected to a small magnetic field, the print-through will be lowered approximatley 4 dB. Therefore, it is good practice to store the recorded tape on the takeup reel, and rewind it before playing. The rewinding of the tape, due to mechanical stress, will lower the echo by approximately 3-to-4 dB. If an "echo eraser" is used (a small magnetic field) which can be a small piece of recorded tape that contacts the tape in use, the print-through can be effectively lowered 8-to-10 dB. The only drawback to this method is that the high frequencies are also attenuated 1-to-1½ dB. The field from this echo-eraser should be approximately 100 oersteds (Oe), not more, and one must be very careful to keep the tape moving or a second print-through will result. In reality, these methods of reducing print-through cause those particles that had their charges changed by adjacent layers to flip back to their original charge. Rewinding the tape also aligns the tape with the guides of the machine, resulting in a smoother motion during playback.

SIGNAL-TO-ERASE RATIO

The characteristic that follows print-through is signal-to-erase ratio. This is a ratio of a recorded reference-level tone, to the same tone, after erasure. An erased tape is a tape whose particles have completely random charges. Therefore, when this tape passes across the gap of the playback head, it cannot induce any particular charge, plus or minus. Theoretically, the erased tape should have a neutral charge, with the pluses cancelling out the minuses and vice-versa. In order to create this random state of flux, one uses a strong high frequency field which is constantly changing the charge of the particles in the field from plus-to-minus and minus-to-plus. The tape must then slowly leave this very strong field, so that as it moves away from the magnetic field, the magnetic flux decreases, thereby leaving a completely random charge on the tape.

Having a cleanly-erased tape for recording is essential, and of course most recorders have an erase head as the first head the tape comes in contact with. As the tape passes across this head, it enters and leaves the high-frequency erasure field; as it leaves the field, it leaves with completely-random charges. If one tries to record on an un-erased tape, the result would be both signals existing at the same time, somewhat analogous to a double exposure in photography. The erasure field has to be strong enough to overcome the coercivity of the tape. Normally, one would need a field that is a minimum three-times-higher than the coercivity of a given tape. This field must penetrate the entire coating depth in order to fully erase the tape. Therefore, it is normally safe to say that a tape requiring more bias (having a higher coercivity) will also require a greater erasure field in order to achieve a random magnetization on the tape.

Another specification that is quoted on many spec sheets is uniformity within a reel and from roll-to-roll. Sensitivity of the tape should be as uniform as possible throughout the roll in question and, from roll-to-roll from the same manufacturer. This is an important specification since one wants the recorded level at the beginning of the roll to be the same as that at the end. We would also like the level to be unchanged from roll to roll, so that when many rolls are used to create an album, the recorded level remains constant. A good specification for uniformity at 1 kHz would be less than, or equal to, ± ¼ dB and from roll-to-roll less than, or equal to, ± ½ dB. Of course, the less level fluctuation, the better.

TAPE WIDTH TOLERANCES

Two tolerances for tape width are usually stated: one for quarter-inch tape, and another for half-, one-, and two-inch tapes. The higher the mid-point of the tolerance, and the narrower the tolerance, the better the slit. Agfa and 3M have the same slitting tolerances: that is, +0/-0.0025 and +0/-0.004. Ampex slitting tolerances are +0.001/-0.001 and +0.002/-0. Slitting is an extremely difficult operation, and keeping the tape within tolerance is each manufacturer's own trade secret.

All specifications that are printed for a given tape are derived from tests done by the manufacturer under optimum operating conditions, with perfectly-aligned equipment and experienced engineers, so the tests reveal the best results possible. In the field, however, we all know that this is not necessarily the case. Variations in temperature and humidity can affect the performance of a tape; equipment is not always perfectly-aligned or adjusted; tape guides can wear; heads can wear; electronic parts become old and off-tolerance. Therefore, using the finest software is very important because the hardware is far from perfect. Specification sheets can only offer one so much information. Specifications are in-house tested results which can only give a guideline as to how the product should perform. Another important aspect to consider is that a specification sheet may show data from hand-selected samples or it can show what the customer can buy every day, off-the-shelf. It is good for the consumer to seriously question the manufacturer as to how the company justifies its data. It is best for the user to test each product and make his own evaluations and comparisons in order to determine which product will suit him best.

There are many characteristics of magnetic tape that do not show up in the specification sheets. These are subjective qualities that only the user can determine for himself. The manufacturing of magnetic tape is something like 30 per cent science and 70 per cent art. Therefore, it is important to stress using one's ears and one's own equipment to evaluate the product. Use the specification sheet, and the information contained in this paper, to properly adjust the equipment and, as an outline to test the tape to see if it really does perform as well as the manufacturer states.

In the magnetic tape field, plus-or-minus X dB is not always the main concern. Consistency of product and ease of use is the winning formulation.

CONCLUSION

The purpose of this paper was to review the electro-acoustic properties of audio recording tape, help the reader to understand what the properties are, how they are derived, how the curves indicate each specification and to evaluate and put each specification in perspective.

It should be understood that a document such as this contains many limitations. Manufacturers develop new test methods and production techniques; equipment is constantly being updated and re-engineered. The two largest limitations appear to be that the concept of magnetic recording is still in its relative infancy and, secondly, tape, as was mentioned earlier, is more an art than a science. Recording a piece of music entails a great deal of artistry and what the user hears and imagines during playback is, for the most part, impossible to measure, but of the utmost importance. ■

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	8-Tr.	Cas.
1 min. to 45 min. any quantity	80¢	65¢
46 min. to 65 min. any quantity	90¢	72¢
66 min. to 80 min. any quantity	98¢	85¢
81 min. to 90 min. any quantity	$1.08	89¢
Shrink Wrapped & Labeled add	14¢	13¢

Reel-to-Reel 3m tape 1800′ $5.00
Blank VHS 2/4 hr. video tapes $19.00

Low Cost Shrink-Wrap Equipment Available
PROFESSIONAL 8-TRACK DUPLICATORS
$1,495.00
CASSETTE & 8-TRACK CALIBRATORS & ERASERS ($35.00 minimum order)
BAZZY ELECTRONICS CORP.
39 N. Rose, Mt. Clemens, Mich. 48043
Phone: (313) 463-2592
Master Charge and Visa Welcome

BGW: FOR IMMEDIATE DELIVERY. **UAR Professional Systems, 8535 Fairhaven. San Antonio, TX 78229. 512-690-8888.**

STEPHENS 24 TRACK tape recorder with 32 tracks of built-in electronics and 40 track meter bridge; can record 16, 24, 32 or 40 tracks on two inch tape. Has 16 and 24 track heads. 1 year old. Under warranty, excellent condition. **Queen Village Recording Studios, 800 S. 4th St., Phila., PA 19147. (215) 463-2200.**

IVIE ELECTRONICS REAL-TIME ANALYZERS, etc. Very slightly used demonstrators at discount. Full factory warranty. Money-back guarantee. **JML Company, 39,000 Highway 128, Cloverdale, CA 95425.**

AKG, E/V, Sennheiser, Shure, Neuman: FOR IMMEDIATE DELIVERY most models. **UAR Professional Systems, 8535 Fairhaven, San Antonio. TX 78229. 512-690-8888.**

SERVICES

CUTTERHEAD REPAIR SERVICE for all models Westrex, HAECO, Grampian. Modifications done on Westrex. Avoid costly down time; **3-day** turnaround upon receipt. Send for free brochure: **International Cutterhead Repair, 194 Kings Ct., Teaneck, N.J. 07666. (201) 837-1289.**

ACOUSTIC CONSULTATION—Specializing in studios, control rooms, discos. Qualified personnel, reasonable rates. Acoustilog, Bruel & Kjaer, HP, Tektronix, Ivie; equipment calibrated on premises. Reverberation timer and RTA rentals. **Acoustilog, 19 Mercer Street, New York, NY 10013 (212) 925-1365.**

MAGNETIC HEAD relapping—24 hour service. Replacement heads for professional recorders. **IEM, 350 N. Eric Drive, Palatine, IL 60067. (312) 358-4622.**

JBL AND GAUSS SPEAKER WARRANTY CENTER. Full lines stocked. Instant recone service, compression driver diaphrams for immediate shipment. **Newcome Sound, 4684 Indianola Ave., Columbus, Ohio 43214 (614) 268-5605.**

AMPEX SERVICE COMPANY: Complete factory service and parts for Ampex equipment; professional audio; one-inch helical scan video systems; video closed circuit cameras; instrumentation consumer audio; professional audio motor and head assembly rebuilding. **2201 Lunt Ave., Elk Grove Village, IL 60007; 500 Rodier Dr., Glendale, CA 91201; 75 Commerce Way, Hackensack, NJ 07601.**

MULTI-TRACK RECORDING specialists—1-2-4-8-16-24 tracks, authorized dealer for Tascam, Otari, Ampex, Teac, Technics, AKG, AB Systems, Crest, SAE Pro, dbx, Orban, Tapco C12, Audioarts, Loft, Lexicon, Ashly Audio, Altec, PAS, PSL, Shure, and many more. Single items or complete studio packages. Studio design and construction. Phone or write for a prompt written quotation. **Professional Sound Labs, Inc., 42 North Franklin St., Hempstead, NY 11550. (516) 486-5813.**

WANTED

WANTED: USED RECORDING equipment of any kind. Expanding studio will pay cash. **Dennis Reed, Box 50022, Palo Alto, CA 94303 (415) 494-9344.**

WANTED: RECORDING EQUIPMENT—mikes, recorders, consoles, outboard gear. **Greg Field, Box 243, San Mateo, CA 94401 (415) 343-1353.**

WANTED: MCI JH-16 parts (heads, headstacks, guides, amps, control cards, etc.) and Ampex 440 control cards. Also, 8 or 16 track console for $8,000 to $10,000. Call **Bill Kenner—(901) 726-1900.**

EMPLOYMENT

WANTED: FULL TIME maintenance assistant for major Long Island recording studio. Experience a must! Send resume to: **P.O. Box 192, East Norwich, NY 11732.**

TM PRODUCTIONS/PROGRAMMING seeks additional maintenance engineer. Degree, five years experience, and familiarity with MCI multi-track gear preferred. Send resume to **Ken Justiss, Vice President, TM Productions; 1349 Regal Row; Dallas, TX 75247.** No phone calls.

FACTORY SALES

We are searching for a personable, well organized, enthusiastic, individual to join our Professional Products sales team. The ability to communicate effectively with studio engineers, audio equipment dealers, and factory representatives is essential. If you enjoy traveling approximately 80% of the time and have a working knowledge of professional audio equipment, send your resume to:

Larry Jaffe
dbx, Inc.
71 Chapel St.
Newton, MA 02195

ENGINEERS/TECHNICIANS—Expanding NYC re-recording facility has opportunities for experienced maintenance engineers/technicians and recording engineers. **Dept. 101, db Magazine, 1120 Old Country Road, Plainview, NY 11803.**

SALES MANAGER

We are a leading manufacturer of professional audio products seeking a Regional Sales Manager to direct our Eastern Sales activities both directly and through manufacturer representatives.

The ideal candidate would have good sales and marketing experience, a background in commercial sound and high level music systems and be willing to travel.

We provide an excellent compensation program including a base salary and incentive program, car and expenses. Send resume and salary requirements to:

Dept. 111
db Magazine
1120 Old Country Road
Plainview, NY 11803

Equal Opportunity Employer M/F

JBL

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We're growing and in need of an additional qualified Applications Engineer with a minimum of 5 years professional experience in engineered sound systems design.

Please submit resume including salary history for prompt and confidential consideration to:

Linda Herod

JAMES B. LANSING SOUND, INC.

8500 Balboa Blvd.
Northridge, CA 91329

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Product Coordinator

SHURE

SHURE . . . a leader in professional sound reinforcement, PA, and audio products . . . is seeking the right individual to fill a highly visible and challenging position in our Marketing Division.

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- Experience in researching, analyzing and predicting market trends
- Ability to make recommendations to upper management for formulating new product marketing strategy
- Retail audio sales experience a plus

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Shure Brothers Incorporated,
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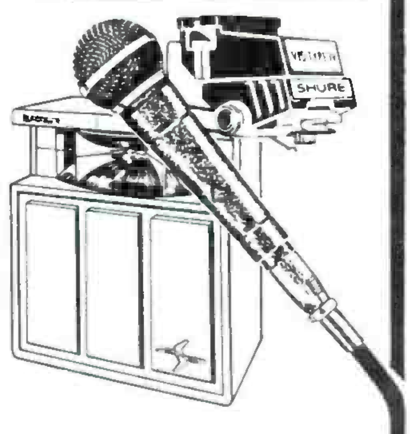

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People/Places/Happenings

• Completing their recent plant expansion and facilities renovation, **Harrison Systems, Inc.**, Nashville, Tennessee, has added about 4,000 square feet of new space to their module production areas. Extensive remodeling and improvements in existing space utilization include: provision of a new area for software development, expanded and remodeled administrative office space, and numerous enhancements to new-product development facilities. In addition, Harrison has also quadrupled the size of its engineering, documentation, and printed circuit board layout areas.

• **GenRad, Inc.**, Concord, MA, has established a wholly-owned subsidiary designated **GenRad Benelux** to succeed **Geveke Elektronica B.V.** and its affiliate, **SHV Belgium N.V.**, as distributor of GenRad products in the Benelux countries of Belgium and the Netherlands. Geveke will continue its association with GenRad by providing maintenance service on behalf of GenRad Benelux.

• Responsible for formulating advanced broadcast systems for the industry, **Michael Ziomko** has been promoted to the position of vice president, as well as joining the Board of Directors at **DYMA Engineering**, Taos, New Mexico. Prior to his promotion, Mr. Ziomko served as sales manager for the company.

• A new marketing and sales management team has been formed by **Analog & Digital Systems, Inc.**, Wilmington, MA. Comprising this team are three newcomers to the ADS organization: **Harron K. Appleman**, marketing manager; **Christopher C. Browder**, sales manager for the Western half of the country; and **William R. Duvall, Jr.**, sales manager in charge of the Eastern United States and Canada.

• **Dr. Donald S. McCoy**, a distinguished engineer, known for his recent work in advanced technical development of video disc systems (as well as for his earlier research in audio and television technology), has been appointed vice president and general manager of **CBS Technology Center**, Stamford, CT. Succeeding the late **J. Kenneth Moore**, Dr. McCoy comes to CBS Technology Center from **RCA's David Sarnoff Research Center**, in Princeton, N.J. With RCA for 22 years, Dr. McCoy was most recently division vice president, technical liaison for the company's video disc system.

• A twenty-year veteran of **Shure Brothers Inc.**, Evanston, IL, **Bernhard W. Jakobs** has been promoted to the position of vice president, engineering. In his new post, Mr. Jakobs will manage all of the company's design and development engineering departments. Prior to his promotion, Mr. Jakobs served as director of engineering and head of the engineering division.

• Named vice president of marketing for **Koss Corporation**, Milwaukee, WI, **Robert C. Bukowsky** will be accountable for all areas of marketing, including new product development, marketing planning and marketing services for the company's operations. Previously, Mr. Bukowsky was general manager of the Gerber health care division of **Gerber Products**, Fremont, MI.

• Recently merging with **Jonsson Communications Corporation, Heavenly Recording Studios**, Sacramento, CA, is expanding with the addition of 24-track recording capabilities. Included in the installation of the 24-track equipment will be 24-tracks of professional dbx noise reduction equipment; making Heavenly Sacramento's first 24-track dbx studio.

• The **Bedford Stuyvesant Restoration Corporation**, Brooklyn, NY, announces the grand opening of its new 24-track studio, **The Platinum Factory**. The new 3,000 square foot facility—constructed with a $265,000 grant from **CBS, Inc.** and designed by **John Storyk**, of **Sugarloaf View**—is Brooklyn's first 24-track recording studio.

• Responsible for all national and international sales, **Robert R. West** has been appointed sales and marketing manager for **Waters Manufacturing, Inc.**, Wayland, MA. Prior to joining Waters, Mr. West was a product manager for **Unex Laboratories**, Danvers, MA.

• **RCA Custom Recording** has now joined the rest of the Record Division of **RCA Ltd.** in its new headquarters at 1, Bedford Avenue, London W. C. 1. The new facility, designed by **Jack Edwards** of Glendale, CA, comprises two master cutting rooms utilizing Neumann lathes with Ortofon cutterheads. An editing dubbing studio and a cassette sub-mastering suite complete the new complex, with JBL and Tannoy speakers used throughout.

• **Ralph E. Green** has been promoted to the position of vice president, engineering, **CBS Radio Division.** Joining the company in 1950, Mr. Green has been CBS Radio's director of engineering for the past ten years. Responsible for the design and construction of all broadcast facilities required by the Network and the 14 CBS Owned a.m. and f.m. stations. Mr. Green also negotiates with technicians' unions having contracts with CBS Radio and represents the Division in FCC technical matters which may affect the operation of the Network and the 14 stations.

• **James B. Lansing Sound, Inc.**, undergoing a major expansion of its International Division, has announced three key departmental appointments. **Randy Patton**, formerly on the European staff of **Harman International**, has joined the division as sales manager, consumer products; **Ruth McNevin**, most recently assistant manager of the international division, has been promoted to operations manager; and **Garry Margolis**, an applications engineer for JBL's professional products marketing teams, has been promoted to the post of sales manager, professional products.

• Specializing in equipping recording studio facilities and in providing sophisticated home recording systems for music industry pros, **Vision-Sound Professional Audio Inc.**, a new audio consulting firm and dealership, has been formed by audio consultant **Michael Salafia**. Mr. Salafia previously served for two years as a sales representative for **Audiotechniques.** Representing a full range of state-of-the-art equipment, Vision-Sound is also the exclusive East Coast representative for **Neotek Recording Consoles.** In addition, the firm's new headquarters are being acoustically treated to provide clients with the opportunity to audit equipment functions in a studio environment. Vision-Sound is located at 110 Grand Avenue, Englewood Cliffs, New Jersey. (201) 871-4101.

• **James C. Leu** has been named president of **UMC Electronics Co.**, North Haven, CT, assuming total responsibility for all operations at UMC and the Broadcast Products and United Manufacturing Divisions. Mr. Leu brings to UMC some 30 years' experience at the U.S. Electrical Motors Division of **Emerson Electric Company**; where he most recently served as the president of the Division's International Operation.

Technics RS-M95. When price is no object. But performance is.

There are countless cassette decks to choose from when your object is price. But when your only object is performance, look at the RS-M95 with metal tape capability.

The heart of the RS-M95 lies in its quartz-locked direct-drive capstan motor and its computer-controlled tape tension system. Together, they team up to provide constant tape tension, remarkably low wow and flutter and complete immunity from speed inaccuracy. So when you record an A flat, you'll hear an A flat. No more, no less.

But if more is what you want, you'll get it with metal tape. Because, compared to conventional tape, it gives you a frequency response with more dynamic headroom and more high end extension.

You'll also get the advantage of the RS-M95's three-head configuration? Source/tape comparison. What you don't get is azimuth error, because our HPF record and playback heads are not only precisely gapped, they're also precisely enclosed in a single housing.

To add to the RS-M95's three-head configuration, we added double Dolby*. So you can monitor your tapes with the full effects of Dolby noise reduction.

Another glowing feature of the RS-M95 is its two-colored fluorescent (FL) bar-graph meters. They're fast, thanks to a device attack time of just 5 millionths of a second. And accurate, thanks to a Technics high-slew-factor operational amp for absolute wave and peak detection.

Also included are 4-position tape selectors with bias fine adjustments. A built-in 400 Hz/8 kHz test tone oscillator. A separate, coreless DC motor for reel drive. A microprocessor tape counter with triple memory functions and feather-touch controls.

Technics RS-M95. Its performance is its only reason for being.

FREQUENCY RESPONSE (Metal): 20-20,000 Hz (±3dB). WOW AND FLUTTER: 0.03% WRMS. S/N RATIO (Dolby in): 70dB. SPEED DEVIATION: ±0.1%.

*Dolby is a trademark of Dolby Laboratories.

Technics
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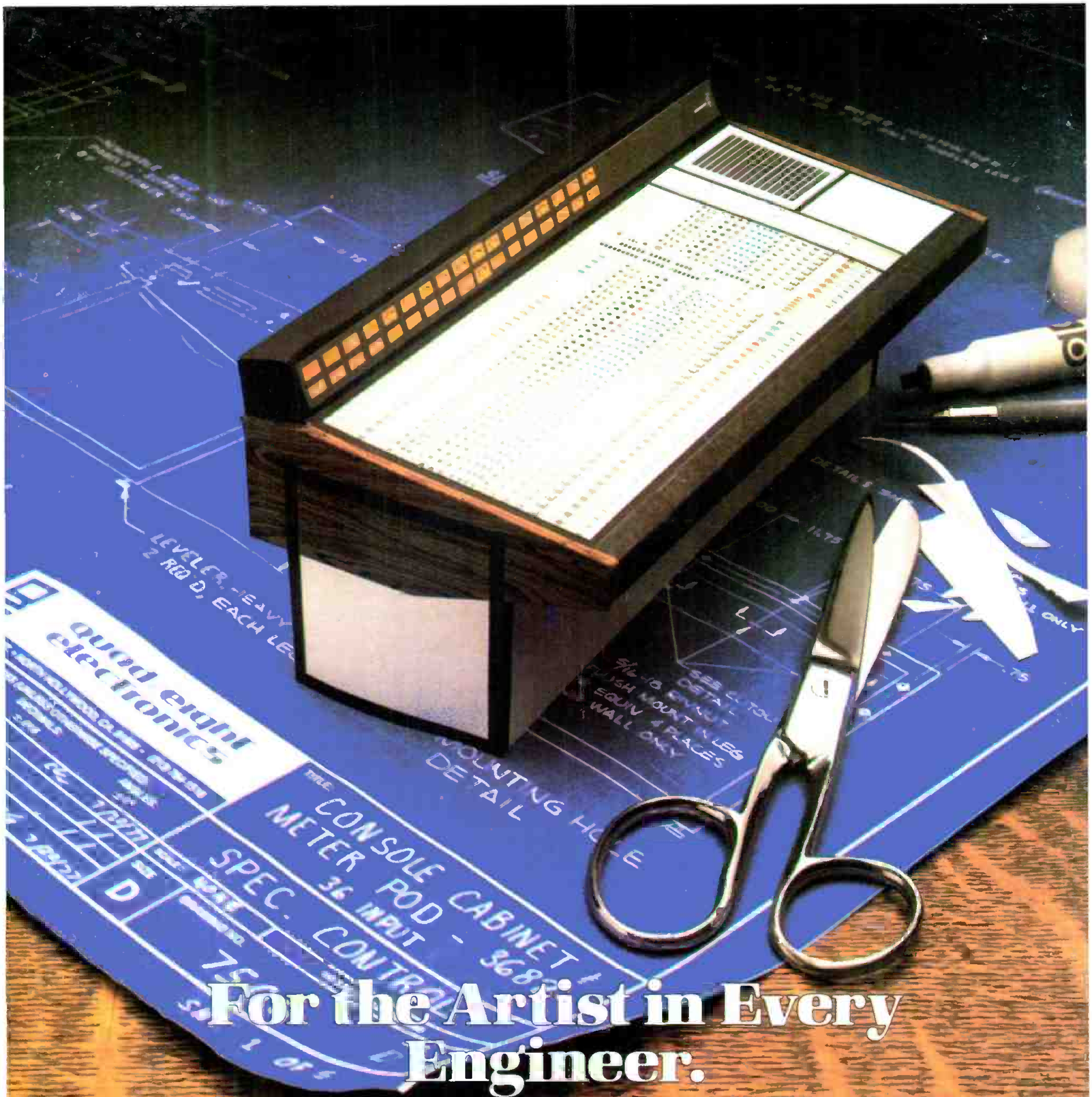

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