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THE SOUND ENGINEERING MAGAZINE

db

APRIL 1979
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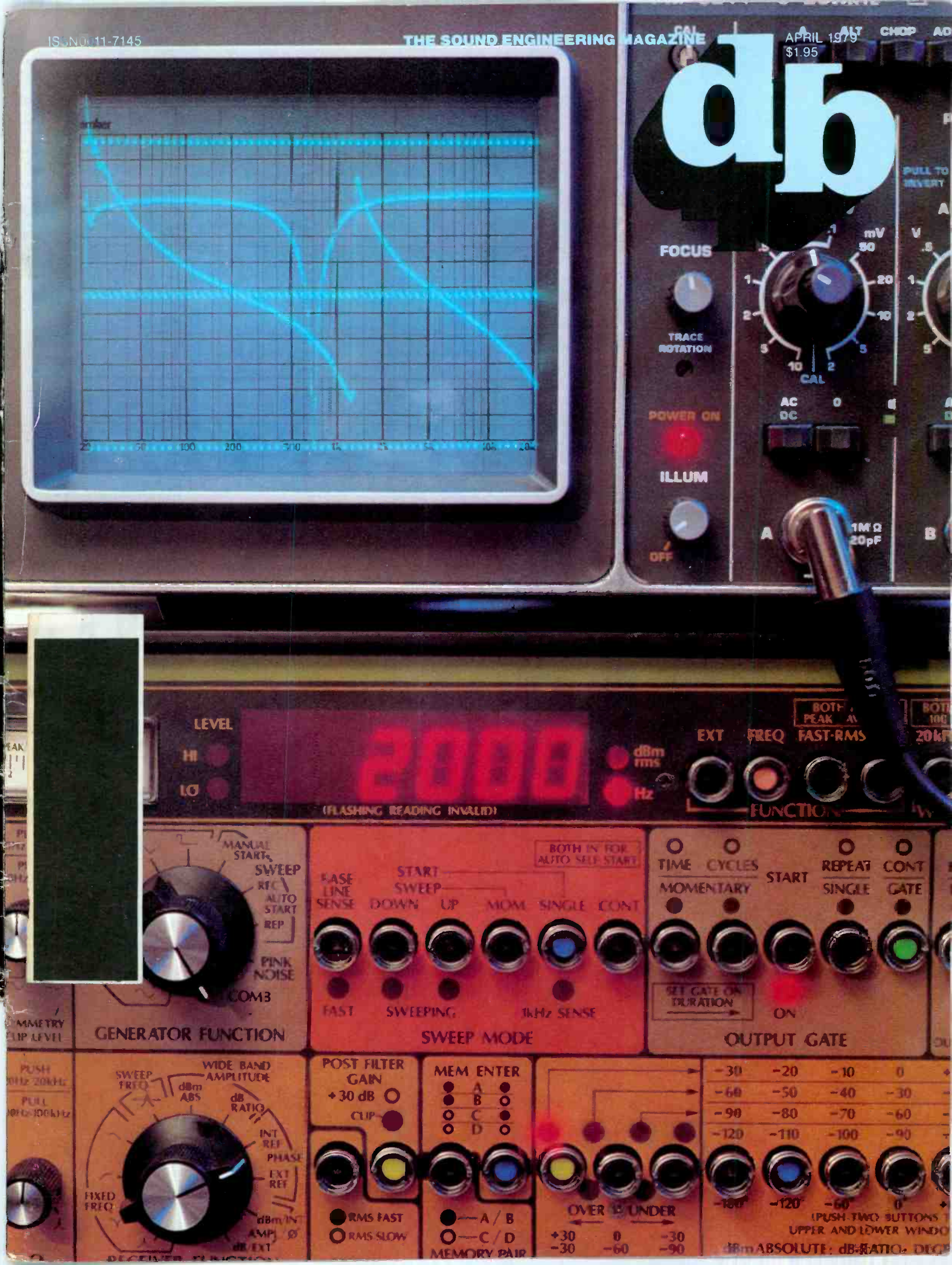

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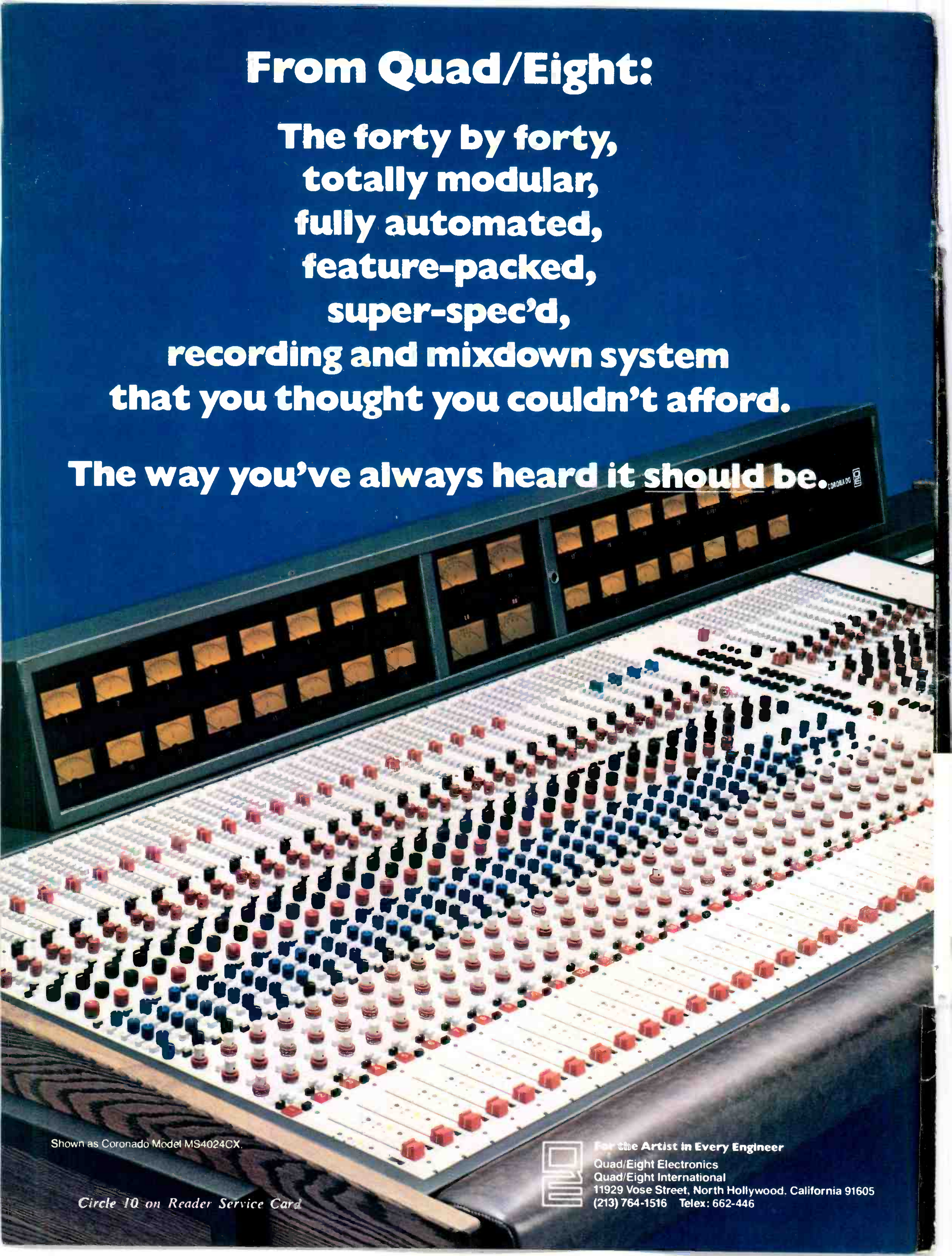
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Coming Next Month

- A little more about test equipment, and then its on to the world of signal processing. We'll have feature articles on choosing equalizers, and on the design of a dynamic range controller.
- Of course, digital signal processing will also be considered, with a general over-view of the subject, and a specific look at a shared access memory system.
- And, for the do-it-yourself crowd, Albert Hayes will be back with his programmable calculator, to help us design some notch filters.

About The Cover

- This month, we take a close look at test equipment, and for our cover, photographer Robert Wolsch has done so too. Experienced 'scope watchers are invited to identify the traces, and to let us know why the display is an improbable one.

By the way, the equipment is a Philips PM 3214 dual trace oscilloscope, and an Amber 4400a Audio-Test Set.

db

THE SOUND ENGINEERING MAGAZINE

APRIL 1979 VOLUME 13, NUMBER 4

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

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db Letters

TO THE EDITOR:

I was amused to see the perpetuation of the conventional, but inaccurate theory of the effect of bias in Peter Vogelgesang's article in the January, 1979 **db** Magazine. This theory obviously fails completely with the modern magnetic media which exhibit an almost rectangular magnetization characteristic.

My "bubble" theory, first published a dozen and a half years ago, is far more accurate. I am enclosing a copy of the original paper, and also a simple explanation which I published in the ISA Journal in 1965.

BENJAMIN B. BAUER
Consultant

Mr. Vogelgesang Replies:

The "bubble" theory is an imaginative visualization of the recording process which may be an aid for developing an intuitive grasp of magnetic recording. Unfortunately, it does not provide a quantitative measuring tool by which bulk magnetic properties of a magnetic material can be mathematically translated into tape performance. The modified anhysteretic magnetization model which I described in my article has been incorporated in a computer program which allows us to simulate and predict precisely how a magnetic material will perform in a tape. Far from failing, the model is supported by extensive empirically obtained data which matches closely the theoretical predictions derived from the simulation program. Such correlation could not be obtained by accident. The "bubble" model provides no such quantitative utility.

I submit that the anhysteretic model is *self*-perpetuating, and that the "bubble" model has not received general acceptance because of its limited usefulness.

PETER J. VOGELGESANG
Laboratory Manager
Advanced Recording Technology
3M

db Replies:

We've asked Mr. Bauer to give us an 'update' on his "bubble" theory, which we hope to publish in a future issue of **db**.

TO THE EDITOR:

I read the letter that appeared in your December '78 issue on the subject of the spelling of the word "disk"—in connection with disk records. As grandson of the inventor of the disk record, Emile Berliner, I thought I'd

index of advertisers

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fact: this condenser microphone sets a new standard of technical excellence. & it sounds superb!

The Shure SM81 cardioid condenser is a new breed of microphone. It is a truly high-performance studio instrument exceptionally well-suited to the critical requirements of professional recording, broadcast, motion picture recording, and highest quality sound reinforcement — and, in addition, is highly reliable for field use.

Shure engineers sought — and found — ingenious new solutions to common problems which, up to now, have restricted the use of condenser microphones. Years of operational tests were conducted in an exceptionally broad range of studio applications and under a wide variety of field conditions.

As the following specifications indicate, the new SM81 offers unprecedented performance capability — making it a new standard in high quality professional condenser microphones.

SM81 puts it all together!

- WIDE RANGE, 20 Hz to 20 kHz FLAT FREQUENCY RESPONSE.
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- WIDE RANGE SIMPLEX POWERING includes DIN 45 596 voltages of 12 and 48 Vdc.
- EXTREMELY LOW RF SUSCEPTIBILITY.
- SELECTABLE LOW FREQUENCY RESPONSE: Flat, 6 or 18 dB/octave rolloff.
- 10 dB CAPACITIVE ATTENUATOR accessible without disassembly and lockable.

Outstanding Ruggedness

Conventional condenser microphones have gained the reputation of being high quality, but often at the expense of mechanical and environmental ruggedness. This no longer need be the case. The SM81 transducer and electronics housing is of heavy-wall steel construction, and all internal components are rigidly supported. (Production line SM81's must be capable of withstanding at least six random drops from six feet onto a hardwood floor without significant performance degradation or structural damage.) It is reliable over a temperature range of −20° F to 165° F at relative humidities of 0 to 95%!

Send for a complete brochure on this remarkable new condenser microphone! (AL577)

SM81 Cardioid Condenser Microphone

SHURE®

Shure Brothers Inc., 222 Hartrey Ave., Evanston, IL 60204, In Canada: A. C. Simmonds & Sons Limited
Manufacturers of high fidelity components, microphones, sound systems and related circuitry.

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kind as to either let us have the address or forward our letterhead to them.

Thank you for whatever help you may give.

CLIFF JURANIS
Reading, PA

db Replies:

We'll do both. Cybersonics' address is:
P.O. Box 8327
Universal City, California 91608

TO THE EDITOR:

I am presently enrolled in a recording engineering course at the University of Sound Arts in Hollywood, CA. I expect to finish the entire course by mid-April of this year.

I am interested in becoming a student member of the Audio Engineering Society, and will very much appreciate if you can send me some information about being a member.

RICHIE MERLUZA
Los Angeles, CA

db Replies:

For an application for membership in the AES, you can write to the Society at:
60 East 42nd Street
New York, NY 10017

There is also a Los Angeles section, which may be able to supply applications, the address is:
P.O. Box 1806
Hollywood, CA 90028

TO THE EDITOR:

Perhaps you can tell me what application would require all four conductors of a three-conductor, 1 shield cable thereby utilizing that capability of the Switchcraft connector?

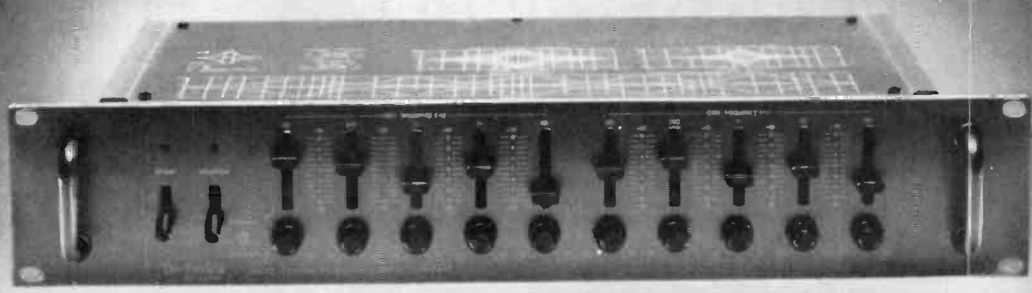
BRYANT ARRINGTON
Technical Consultant
Long Engineering Company

db Replies:

The same feature is also available in the Neutrik line of cable plugs—a fact we overlooked in our "Nuts-'n-Bolts" feature in the December, 1978 issue of **db**.

Several applications have been suggested for that third conductor. For ribbon microphones with center-tapped transformers, it's a reliable way to keep the center tap grounded. It's also a way for keeping phantom-power voltages off the shield. This is certainly not an absolute necessity—just a nice touch. In some applications, shields take a lot of physical abuse, and that extra wire may be a more-reliable conductor. For the true fanatic, this also allows the shield to remain unconnected at the microphone end of the cable, thus eliminating any chance of it becoming a (hum) conductor. Of course, this means you

All the features you'd expect from a 2-channel parametric equalizer. At a price you don't.



A radical departure in circuit principles, Technics SH-9010 universal frequency equalizer offers the experienced technician and demanding audiophile the flexibility of a 2-channel parametric equalizer with five bands per channel.

Each band has a center frequency that's continuously variable. By turning the control knob below each slide pot, the center frequency can be varied up or down by as much as 1.6 octaves. So, unlike conventional equalizers with a fixed-center frequency, the SH-9010 has no frequency "blind spots." What's more, each band of the SH-9010 can adjust to overlap the adjacent band to further boost or attenuate a selected frequency width.

Incredible for the price? You're right. But what's even more incredible is that variable center frequency is just one of the SH-9010's advantages. Variable "Q" or bandwidth is another. With it you can broaden or narrow any frequency band. Independently or both at the same time. Which means you can balance an entire string section or eliminate an annoying little hum.

Technics SH-9010. Compare specifications and prices. And you'll agree there's no comparison.

THD: 0.02%. FREQUENCY RESPONSE: 10 Hz—20 kHz (+0, −0.2 dB). 10 Hz—70 kHz (+0, −3 dB). GAIN: 0 ± 1 dB. S/N: 90 dB. BAND LEVEL CONTROL: +12 dB to −12 dB (5 elements x 2). CENTER FREQUENCY CONTROL: +1.6 oct. to −1.6 oct. BANDWIDTH (Q) CONTROL: 0.7 to 7.0. CENTER FREQUENCIES: 60 Hz (Variable 20 Hz ~ 180 Hz), 240 Hz (Variable 80 Hz ~ 720 Hz), 1 kHz (Variable 333 Hz ~ 3 kHz), 4 kHz (Variable 1.3 kHz ~ 12 kHz) and 16 kHz (Variable 5.3 kHz ~ 48 kHz). SUGGESTED RETAIL PRICE: $560.00.*

Technics SH-9010. A rare combination of audio technology. A rare standard of audio excellence.

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

Technics
Professional Series

Circle 45 on Reader Service Card

can't connect two such cables together, since there would now be no shield-continuity from cable-to-cable.

In the June 1970 issue of **db**, *Gerhard Bore's article on "Powering Condenser Microphones" discussed the potential (pardon the pun) problems of powering via the cable shield:*

"The d.c. through the cable shield produces a small voltage drop which might fluctuate slightly during operation if, for example, two connectors along extended cable runs of two adjacent microphones touch for a moment, shorting out this voltage drop for an undefinable period of time. The result is fluctuation of the supply voltage by the amount of this voltage drop.

"Besides this, undesirable currents might be caused to flow through the shield should a connector or the microphone housing come in contact with a metal object which is at a different potential from operating ground. This might occur if the console and its shields are at ground potential and a connector from a microphone cable comes in contact with a pipe of the central heating system or an electrical device grounded to the protective a.c. ground. This would cause a.c. with power-line frequency to flow through the cable shield."

The June issue is now out-of-print, but reprints of this article are available from Gotham Audio Corp. 741 Washington Street, New York, N.Y. 10014.

Department of Corrections

Murphy's Law, applied to Fourier analysis.

This is the law that states that, when old errors are corrected in the type set, new ones will be introduced. On page 39 of our February issue, the formulas *should* have looked like:

$$e = \frac{4}{\pi}[\sin \omega t + \tfrac{1}{3} \sin 3\omega t + \tfrac{1}{5} \sin 5\omega t + \ldots]$$

and

$$e = \frac{4}{\pi}[\sin (\omega t + \theta_1) + \tfrac{1}{3} \sin 3 (\omega t + \theta_2) + \tfrac{1}{5} \sin 5 (\omega t + \theta_3)]$$

Our thanks to Edsel Murphy, and apologies to everyone else.

The Bose 1800.

When you turn the power up, it won't let you down.

The Bose Model 1800 power amplifier delivers 400 watts RMS per channel with both channels driven. Its massive power transformer and filter capacitors prevent power supply voltage droop, allowing the amplifier to deliver large amounts of solid, sustained bass.

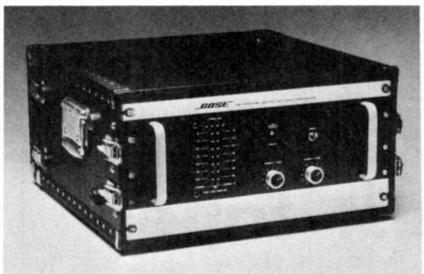
Not only that, the Bose 1800 is so reliable, we gave it the longest warranty of any professional amplifier on the market. What makes it so durable?

The use of 14 power transistors per channel results in unusually low thermal stress. And massive heat sinks reduce the operating temperature even further. Computer-grade electrolytic capacitors increase reliability by providing extra temperature and voltage safety margins. A turn-on delay circuit limits power supply inrush currents to extend the life of the components. Electronic current limiting acts instantly to protect the amplifier from short circuits and other abnormal load conditions.

If you want to take your Model 1800 traveling, it can take the beating of hard road use. Its optional transit case has built-in fans and air-flow baffles to keep it cool, even under continuous high-power use.

With all its ruggedness and reliability, and with enough reserve capacity even under the most demanding conditions, there is no compromise in what you need most from a power amplifier: lots of clean power, with absolutely no audible distortion or coloration.

The Bose Model 1800. Check it out soon.



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db Calendar

APRIL

10-12 Synergetic Audio Concepts Sound Engineering Seminar: Sheraton Harbor Island, San Diego, CA. For registration forms or information, contact: SYN-AUD-CON, P.O. Box 1134, Tustin, CA 92680. (714) 838-2288.

23-26 Audio-Visual '79. Wembley Conference Centre, London. Contact: British Information Services, 845 Third Avenue, New York, NY 10022. (212) 752-8400.

24-26 Electro '79. New York Coliseum, New York, N.Y.

MAY

12 1979 Midwest Acoustics Conference. Topic: Digital Technology: Impact on Recorded Sound. Norris Center, Northwestern University. Contact: William R. Bevan, Shure Bros., Inc., 222 Hartrey Ave., Evanston, Illinois 60204. (312) 866-2364.

15-18 63rd AES Convention (Los Angeles), Los Angeles Hilton, California; Chairman will be Martin Polon, Director, Audio Visual, U.C.L.A., C.A.S.O., Rice Hall 130, 405 Hilgard, Los Angeles, Calif. 90024, (213) 825-8981.

22-24 Synergetic Audio Concepts Sound Engineering Seminar: Sheraton-Universal Hotel, No. Hollywood, CA. SYN-AUD-CON, P.O. Box 1134, Tustin, CA 92680. (714) 838-2288.

JUNE

3-6 13th Annual Summer Consumer Electronics Show. McCormick Place, McCormick Inn, and the Pick Congress Hotel, Chicago, Illinois.

20-22 APRS '79—Annual Exh. of Professional Recording Equipment. Connaught Rooms, London.

"a professional studio recorder with a handle"

"ReVox new B77 is long on performance and short on Mickey Mouse features."

That's what Herb Friedman said about the ReVox B77 in Hi-Fi/Stereo Buyers' Guide.* If you do location recordings, you'll be interested in what Herb has to say.

In addition to evaluating products for magazines, Herb Friedman is Chief Engineer for Tridac Electronic Laboratories and a major New York radio station. As such, he produces taped programming and he knows the real differences between truly professional recorders and others that claim to have "professional features".

Differences like 18dB record headroom, flat response with no low frequency "head bumps", the highest usable dynamic range and the lowest noise of any portable recorder. Add to these such features as all-digital-logic-control of tape motion, large meters with LED peak level indication, self-contained tape splicer, and rugged 37-pound package with a handle and you've got the best location recorder in the world.

If you'd like to know what else Herb Friedman thinks about the B77, please circle reader service number or write to us for complete information including a reprint of his article and a list of professional audio dealers where you may see and hear the ReVox B77 demonstrated.

REVOX

Studer ReVox America, Inc., 1819 Broadway, Nashville, Tennessee 37203 (615) 329-9576
In Canada: Studer ReVox Canada, Ltd.

Circle 38 on Reader Service Card

PATRICK S. FINNEGAN

db Broadcast Sound

F.M. Modulation Monitor Interface

• The modulation monitor is a very important equipment item in any station. Not only does it provide a measurement of the actual amount of modulation taking place, it also provides test capabilities and an important source of air audio which can be used in the station for many purposes besides simple listening to what is on the air. When this audio is distributed throughout the station for these desired uses, proper interfacing to the monitor must be done if we are to retain the original audio quality. This month we will consider what the engineer should look for and be concerned about when he desires to attach external circuits to the station monitor.

KNOW THE MONITOR

The first consideration in the design of some special monitoring arrangement, or the installation of a brand new monitor into the present system, is knowledge of the particular monitor in question. This may seem like a simple, common sense factor but it is well worth mentioning at this point. While all monitors provide the basic functions as required by the FCC Rules, they are not all the same in method, circuitry, and physical layout. The available outputs on the old monitor, for example, may not be the same on the new monitor at all.

Besides the basic difference in monitors, some have many additional features which are optional and must be ordered when purchasing the unit. If the monitor is one which you inherited when you took over the job, the instruction manual may not, by itself, indicate if the monitor does in fact have any of these options. Because there are terminals on the rear of the instrument marked with this function, that does not mean that the terminals are actually connected inside to the non-existing circuit. The fact there are external connections to the terminals are not a sure bet either—the former engineer may have simply used these terminals as a convenient tie point for some homestyle modification he made. To determine if the options are present, you may have to visually inspect the insides of the instrument.

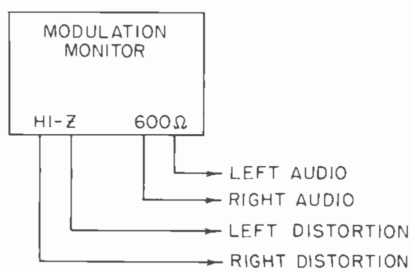
Anyone planning a new distribution, or other special circuitry to attach to the monitor, should take the time to study what that particular monitor actually provides in the way of outputs *before* designing the external circuitry. Without this knowledge he could discover that the special device or circuit he spent much time and effort in designing and building will not fit, won't work properly, or at all.

IMPEDANCE

In addition to checking the number of outputs, a study of the monitor circuits and manual will determine the impedances of the various outputs and what they are intended to work into. For basic monitoring, the output will usually be a 600 ohm circuit. But is it balanced? Is the output transformer coupled? While the audio output for monitoring is generally a balanced circuit, that particular unit may contain an unbalanced output. Some monitors do provide transformer coupled outputs, while in others this is an option that must be ordered or added. If the output is unbalanced and the engineer attaches his balanced system to it, then there may be hum and noise distributed throughout the system.

Most monitors will provide a high impedance, unbalanced output, in addition to the regular monitoring output. This high impedance drive is intended for test instruments, such as

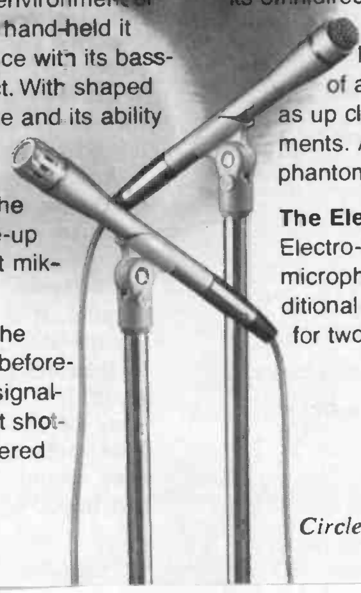
The two most basic outputs provided by the stereo modulation monitor.



Studio quality microphones that don't need a studio to survive.

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 (140dB with 1% THD at 1kHz), the CS15P is ideal for close-up vocal or solo instrument miking applications.

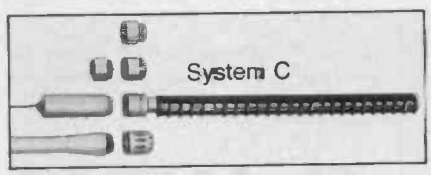
When boom mounted, the CS15P has better gain-before-feedback and a better signal-to-noise ratio than most shotguns. It's phantom powered and it's rugged.



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

Circle 33 on Reader Service Card

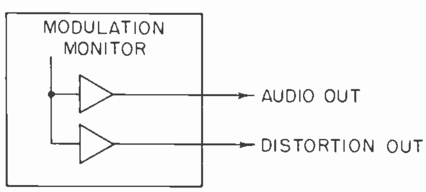
distortion analyzers and noise meters. Since this output is unbalanced, it is important to know which terminal is the high side and which is ground. In some test and trouble-shooting situations reversed leads can produce frustration. Although this output is intended for test instruments, it is an audio output and can be used for other audio purposes so long as the external circuitry interfaces to the high impedance properly. In some models, the distortion and audio-out are fed from separate stages, while in others it is the same stage with a resistor dividing network. If it is a single stage, then adding a low impedance circuit across the distortion terminals will load-down and effect the regular audio monitoring system of the station.

SIGNAL LEVELS

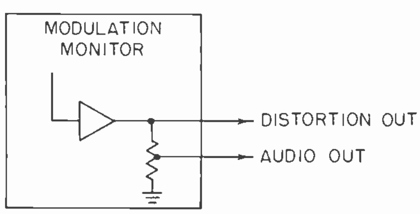
Another important consideration in what we hope to accomplish in the external system is the amplitude of the audio signal available at each of the outputs. The usual level at the Left and Right audio outputs is 0 or +4 dBm at 100 per cent modulation of the carrier. This is at normal distribution levels and adequate for earphone monitoring, but certainly not enough to drive a speaker directly.

The high impedance output will be rated in rms volts for a high impedance load. The voltage level at 100 per cent modulation will be shown in the specification sheet for the model. There may be several other outputs of various impedances that appear on terminals or connectors at the rear of the unit. These too are available for external uses, although some are intended to couple into input ports of

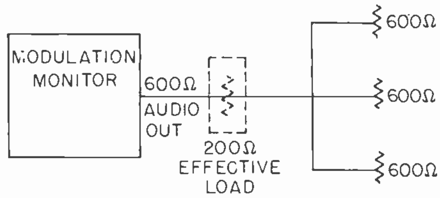
The audio output and the distortion output may be from (A) separate stages, or as in (B) from the same stage.



(A)



(B)



Loads in parallel act the same as resistances in parallel. The stage may soon be overloaded.

other sections of the monitor. Through proper interfacing and decoupling, even these can be used for external purposes, so long as the basic functions of the monitor are not impaired. For all these type outputs, the signal voltage will usually be specified in rms volts, or some may be in peak-to-peak values for oscilloscope measurement. Some of these outputs contain more signal components than just the audio components; for example, the composite output of the baseband monitor. The amount of audio available (even with the appropriate filtering) is not as much as the full rated output level. That specified level contains the supersonic components also.

Still another consideration when selecting audio outputs, is whether it's pre-emphasized or de-emphasized audio? Although the specifications may show the terminals as audio (at the normal audio impedance and levels), it will also specify whether it is pre- or de-emphasized audio. If the normal audio monitoring system were attached to the pre-emphasized audio output terminals, the system's speakers would sound very poor, due to the distorted bandpass of the audio signal.

INTERFACE

Once the particular monitor's capabilities, outputs and their impedances, and signal levels have been determined—and the intended use of the available audio determined; then proper interface can be worked out which will accomplish the desired results and, at the same time, not effect the monitor.

An important principle is to avoid overloading any output with too many common impedances that aren't properly decoupled. Impedances in parallel will act much as resistances in parallel. It is very easy to soon load the monitor circuit very much below its load design specifications. The end result in most cases would be that *none* of the external circuits would have sufficient signal; and performance would be far below what we had hoped to accomplish.

professional quality delay plus special effects

Lexicon's new Model 93 "Prime Time" digital delay processor gives recording studios and entertainers an easy-to-use professional quality time delay with special effects and convenient mixing all at a price you can afford. It combines a degree of flexibility and versatility never before offered in equipment of full professional quality.

- Two delay outputs independently adjustable from 0 to 256 ms
- Complete mixing for delay and reverb processing, freeing up main console channels and tape tracks
- Integral VCO for special effects like vibrato, doppler pitch shift, flanging and automatic double and triple tracking
- Long delay special effects — up to 2 seconds
- All dynamic functions can be footswitch controlled
- 90 dB dynamic range, total distortion below 0.08% at all delay settings

lexicon Lexicon, Inc., 60 Turner Street
Waltham, MA 02154 (617) 891-6790

Export: Gotham Export Corporation, New York, New York

Circle 44 on Reader Service Card

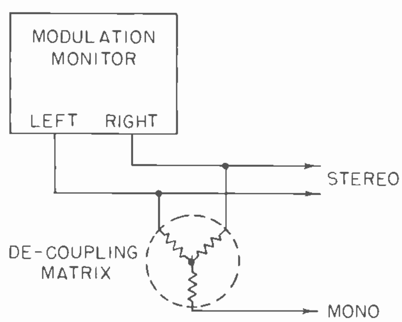
In the case of the basic 600 ohm audio monitoring output, this should run as a single bus, terminated at its end, in a 600 ohm impedance or a resistor. The load may be just a resistor or it may be the monitor input of the console. Any other use of this audio, whether for recording, house monitors or the like, should tap onto this bus by a bridging connection—20K bridging is usually sufficient. This bridging provides isolation, but it will also reduce the signal level available at the output of the bridge by about 20 dB. Your external system should have sufficient gain to overcome this loss. There will always be a loss at the output of the bridging connection. This bridge provides isolation of the feedpoint by high impedance (which is another way of saying it will only tap off a small amount of signal from the feedpoint). Always be prepared to make up the loss in the external system.

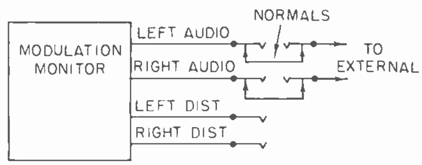
MONO FEED

Although the station is transmitting full time in stereo, there are many cases when only a mono audio signal is required. Such station purposes, for a mono feed, may be: a simple monitor, to check the mono signal that the Left + Right stereo signals provide for the mono listener; a feed to a mono a.m. sister station for a simulcast; or a feed to a silence sensor on a program automation system, and so forth.

Obtaining this mono feed depends upon the monitor, and a study of the manual and circuit diagram will show what is available. A monitor which only proivdes Left and Right audio outputs, can have these outputs combined with a resistor matrix to produce the mono signal at the output of the matrix. If the isolation were not used, simply paralleling the Left and Right outputs will provide mono, but it will also destroy the separate outputs for stereo monitoring.

When combining Left and Right audio to obtain mono, always use a resistor matrix so separate stereo is still available.





Run outputs to jacks for convenience in making patches and tests.

Many monitors also provide a summed Left + Right as well as the separate stereo outputs. This output, however, comes from the main baseband monitor and stems from the fact that the main monitor is also functioning as a standard mono monitor for the total modulation of the carrier. By using this output for mono purposes, there is automatic isolation from the stereo outputs.

JACKS

Whenever it is possible, most of the monitor outputs should be run to a jack strip. This will make it more convenient to run tests such as the proof-of-performance—in addition to operational patch-ups for special recordings, trouble-shooting problems, and so forth. Those outputs which are driving everyday monitoring and recording circuits should run through a set of jacks that are "normaled through." If additional jacks are available, connect at least one pair on the monitor side as a multiple. By the use of multiples, spot checks or tap-ins to the circuit can be easily made without disrupting the circuit.

Other outputs which are not ordinarily used for everyday purposes (i.e., the distortion analyzer output) should be terminated on jacks also. Tests can then be far more convenient, than if they had to be made by attaching temporary wiring to the terminals at the rear of the monitor.

RECAP

The modulation monitor often provides a variety of audio outputs and other outputs which can serve valuable purposes in the station. But all monitors are not alike, and some may or may not contain various options. Before deciding on external circuitry and uses, study the particular monitor to determine exactly what it provides—including signal levels and impedances. Always provide proper interfacing between the monitor and any external circuits you attach, so as not to effect the monitor's operation and the hoped for results in the external system. ■

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NORMAN S. CROWHURST

db Theory & Practice

When Feedback Doesn't!

• After my December column appeared, a prestigious reader picked up on the second paragraph. I was talking about misconceptions about damping factor, and that paragraph opened,

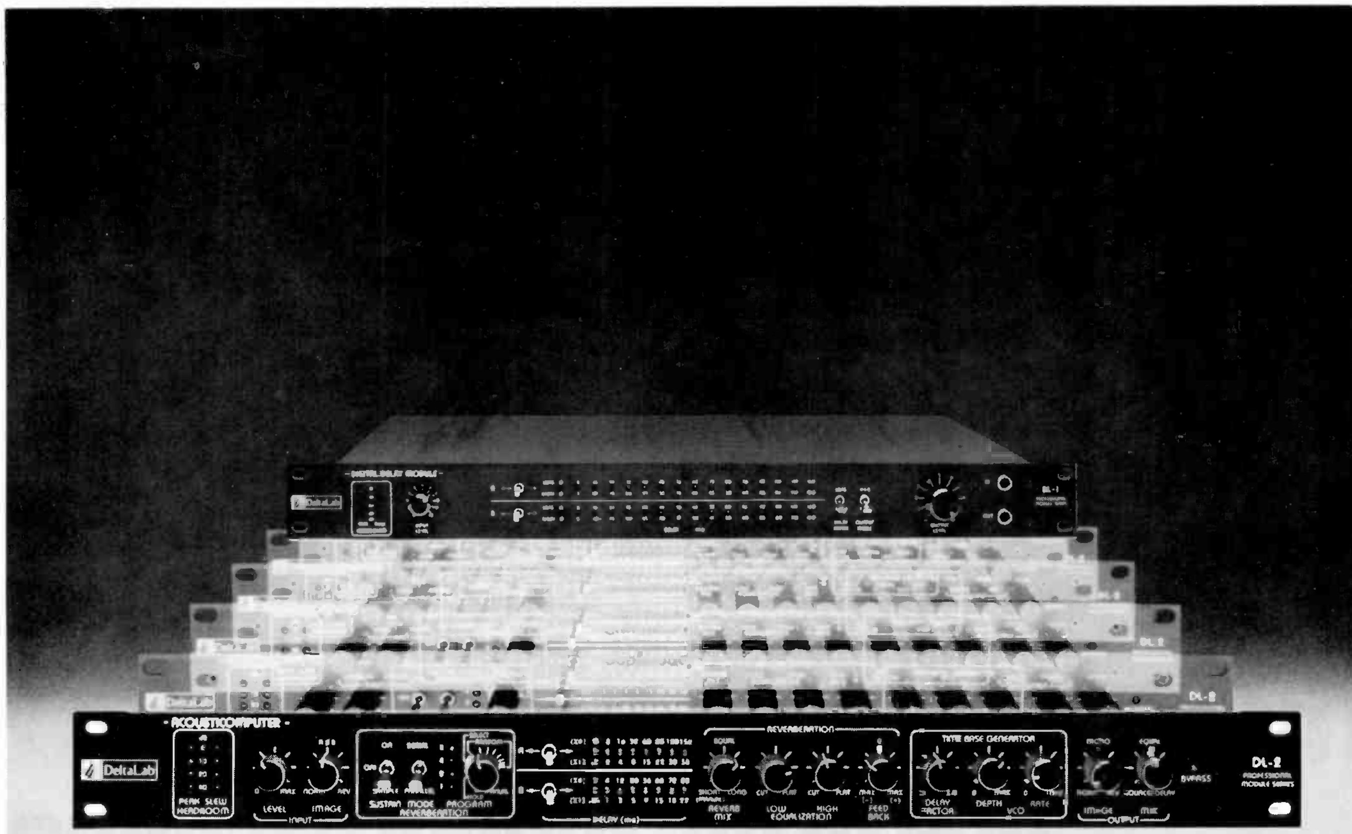
> *The simple fact is that no connection is made from the loudspeaker voice coil to the amplifier input, unless you have a system with what is known as motional feedback.*

The whole thing is a matter of viewpoint, and what it is you are talking about. Damping factor relates to the effect that the amplifier has on the loudspeaker's behavior, and for that purpose I think you are entitled to consider, for the moment, that the amplifier behaves as "good" amplifier, always doing what it is supposed to—just amplify.

Of course, I am aware that they do not always do so, and that is another story. But assuming it does, the December column dealt with its effect on the loudspeaker, resulting from the property known as damping factor. The feedback makes the output impedance "look like" a source resistance that is small, compared to the loudspeaker's own impedance.

Thus, when the amplifier has been driving the loudspeaker at a resonant frequency, and the drive from the amplifier, at that frequency is cut off, the loudspeaker tends to go on moving at that frequency, the amplifier now behaves almost like a short-circuit across the voice coil, damping that spurious sustained movement.

The sentence repeated above was basis for showing that, unless the loudspeaker is equipped with a separate motion-sensing coil, for motional feedback, the *best* that the amplifier can do toward damping, is to provide an equivalent short-circuit across the voice coil terminals. If other frequencies are present, which is usually the case with an ordinary program, the amplifier continues delivering those other



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frequencies, while providing an equivalent short-circuit to the resonant frequency. I hope that "catches up" with what I was saying in December.

My correspondent called attention to a paper delivered at the Hamburg (Germany) Audio Convention in March 1978. While I had not seen or heard that paper, I was glad to see that someone is following up on material I presented in a paper before the Audio Convention in October 1957. It takes a while, I guess!

INTERFACE INTERMODULATION

So let me now try to relate that to what I was saying in the December column. The Hamburg paper was treating what it called "dynamic intermodulation at the amplifier-loudspeaker interface." As I described in a book published in 1952, all the feedback theory, with its $\mu\beta$ formulas, is based on said $\mu\beta$ being a constant. Now if μ, which is the forward amplification of the amplifier, was constant, the amplifier would be perfectly linear, and there would be no need for β, the fraction fed back.

Of course, that further depends on why you want the feedback. The first purpose was to linearize the amplifier—to reduce distortion. So the assumption that μ is constant is fallaceous from the outset. If it were, the feedback would be unnecessary, for its first purpose. But the theory goes on to show that feedback can also modify impedances, both input and output, and stabilize gain.

Now, to illustrate what happens, let us work through some figures, as most people get the point better that way, than using the algebra. Suppose a power amplifier, or the output section of an integrated system, is designed to deliver 50 watts into an 8-ohm loudspeaker, with an input of 1 volt. Suppose it also has what is described as 40 dB feedback.

Now, skipping the relationship between peak and rms, 50 watts into 8 ohms must have the square root of 50 x 8, which is 400, or 20 volts. So 1 volt input produces 20 volts output. But there is 40 dB feedback, which means that the 1 volt input consists of 1 volt external input, balanced out by 0.99 volt fed back, so that the internal amplifier gets 10 millivolts.

That is all fine and dandy, so long as you have a closed feedback loop. But who is going to disconnect it? There is more, before we get into that. The output transistors may have an internal impedance (a.c.) that is 10 times the optimum load—maybe more. So that 20 volts output is coming from an equivalent source resistance of 80 ohms or more. But all the while the feedback is working that is not what it looks like.

The 40 dB feedback makes the source resistance look like 1/100th of the load impedance, or 0.08 ohm. Now let's see what could happen when the 1 volt drive, at the loudspeaker's resonant frequency, suddenly stops. The input stage of the amplifier is designed to accept 10 millivolts, which is what it gets while the 1 volt input is being applied. The 0.99 volts fed back balances it off so the input stage gets only its 10 millivolts.

But now the 1 volt disappears, and the voice coil keeps moving, feeding back its 0.99 volts, or a large part thereof. Unless the damping factor acts quickly enough to "kill" that movement, down to less than 10 millivolts' worth, that input stage is going to be handling a much bigger level than normal, just at that instant.

The input stage can probably handle a lot more than 10 millivolts. But

Circle 37 on Reader Service Card

successive stages build the voltage up, until you get 20 volts at the output, from the 10 millivolt input. Suppose some intermediate stage normally handles 1 volt, from the 10 millivolt input. When the 10 millivolts jumps to 0.99 volt, that stage will suddenly get 99 volts, if nothing overloads, which is obviously unlikely. Something will overload before that.

"LINEAR MODE" TRANSISTORS

Now we have a fact of life that often gets forgotten, because the amplifier uses transistors that are termed "linear," or that are operated in their "linear mode." Every transistor is a semiconductor, which means it conducts current one way, but not the other. It behaves in an approximately linear fashion, only if you keep the base-emitter junction in its conducting region, and the base-collector junction in its non-conducting region, so the current it passes is controlled by that in the base-emitter junction.

Now obviously, a signal that would go to 99 volts, if something did not stop it, is going to run that transistor out of its linear regions and what happens then? The semiconductor properties come into play. Either the base-emitter junction runs into saturation and/or cut-off (non-conduction) or the base-collector junction runs into its conducting region.

Either way, the transistor now looks like a diode or two, rectifying the signal, and swinging the coupling interface, far beyond the linear region. To all intents and purposes, the amplifier feedback loop is now open. Amplification has become zero. This means that the apparent output source resistance, which is connected across the loudspeaker voice coil, jumps from 0.08 ohm to more than 80 ohms.

The 0.08 ohm, across an 8 ohm voice coil, is to all intents and purposes a short-circuit, damping out the resonance. But 80 ohms is a virtual open-circuit, allowing the resonant after-movement to continue virtually unimpeded. At the same time, if other frequencies were present, they have disappeared, when the amplification stopped. All our nice feedback theory went out the window!

Admittedly, that is an extreme case of interface intermodulation. But if that happens when the transistors suddenly go all the way into semiconduction, what happens when they nearly do?

Of course, you will see, when you start looking in detail, that a lot of things could happen—every case is different. When it does begin to happen, does it have a "trigger" or cumulative effect, or does it have a tendency to self-correct?

OTHER FEEDBACK VALUES

Let us look at some of the different values of feedback you could use on that deceptively simple example we took. We said it was 40 dB feedback, and based our figuring on that. We were talking about the amplifier with an 8-ohm load connected. In that condition, if you removed the feedback, 10 millivolts input would produce 20 volts output.

But what if you took the 8-ohm load off, had the amplifier working open-circuit? The fact that its actual, open loop internal impedance is 80 ohms (let us say, it could be more than that) means that it has an equivalent internal output voltage, when delivering 20 volts into 8 ohms, of 220 volts. Now, removing the 8 ohm load would probably not result in the output going up to 220 volts, even with the feedback removed, because something would "go over" before that.

So to find out, you turn down the input even more. You already have it down to 10 millivolts, when you disconnect the feedback. Now, to bring the output down to 20 volts, when you take off the 8-ohm load, you

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must reduce the input by a further factor of 11, to about 0.9 millivolt. So, if you put the feedback on again, to get 20 volts out, you still need 1 volt in, or very close to it.

So in these terms, the change of gain is more than 1000:1, or better than 60 dB feedback. And for the purposes of output impedance reduction, that is what it is—it changes an actual, open-loop source resistance from 80 ohms, to 0.08 ohm, which is a 1000:1 change. But for distortion-reducing, or gain stabilizing purposes, with the 8-ohm output load connected, it is only 40 dB feedback.

And as we said at the outset, those dB figures are based on an assumption that μ has a fixed value, which it has not, in practice. So as well as different "ball park" figures being applicable, in describing the amount of feedback effective for different purposes, the figure changes all the time, even during the fluctuations of an audio waveform.

As the classic theory shows, or should show, ß should be a constant, and it usually is, so that, by making the feedback large enough, it can have a stabilizing effect on μ, the gain with feedback, whichever way you figure it. Now, can you regard the loudspeaker as having an electrical equivalent impedance? We do most of our calculations, assuming the loudspeaker has a "nominal" impedance of, say 8 ohms. We know it doesn't—that it has different impedances at different frequencies. But does that tell the whole story?

That gets into some very interesting questions. The loudspeaker's acoustic resonance does reflect back into the electrical circuit, as an impedance you can measure at the voice coil terminals. Could that impedance be represented as purely electrical elements, resistance, inductance, capacitance, in some configuration or other? And, if it were, would that rather complex impedance, connected to the amplifier in place of the actual loudspeaker, produce the same performance idiosyncracies in the amplifier?

ANALOG CIRCUITS

Here we go again. That is another whole story—of analog circuits. But to be brief—and maybe that is as dangerous as the comment I made in the December column—if the complete acoustical performance can be represented as a measurable electrical equivalent, then connecting that electrical equivalent to the amplifier would have an identical effect, on the amplifier's performance.

Again, from some viewpoints it may be possible, from others not. If the electrical circuit was coupled, 100% through the transducer device (the electromagnetic motor mechanism), then it would be universally applicable. But, as any loudspeaker designer knows, the most efficient loudspeaker ever built, achieved around 50% efficiency. In everyday use, a loudspeaker with an efficiency of 10% is rated as "high efficiency."

This means that the acoustic resonance is not very efficiently reflected into the electrical circuit, so that damping by electrical means can never be very complete. But it can help, which is only why loudspeaker designers recommend using it. ■

MARTIN DICKSTEIN

Sound With Images

Back to Basics Again

• Recently, a booklet on making presentations was written for the personnel under the command of the author. In this magazine-size pamphlet, which has been mentioned before in this corner, some advice is given on the use of audio-visuals. Remember that this material was written for the presenter, not the audio-visual specialist, and so it is sparse in precise detail. Nevertheless, it can act as a good beginning on which to build a more complete paper. Some of these points we may have mentioned in the past, but they are worth repeating to keep the list complete.

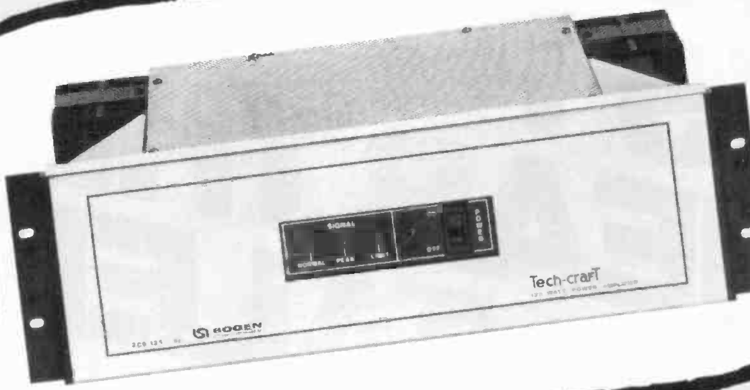
For starters—survey the site of the presentation—if it is to be given on "foreign" ground. Check on things like: positioning of electrical outlets; location of light controls (and whether they are switches or dimmer); whether or not there is a built-in screen available (and how big it is); whether there is a projection booth for the equipment; and finally, a point often overlooked—whether the windows can be darkened. These, of course, are excellent points, but there is more than the a/v man has to think about.

ELECTRICAL REQUIREMENTS

Noticing where the electrical outlets are is only the beginning. Is the outlet for a two-prong or three-prong plug? If there are three-prong outlets, this will facilitate plugging in things like a film projector; if not, provision must be made by bringing 3-to-2 adapters. Is there a sufficient number of outlets for the many pieces of equipment required in the set-up—or must some provision be made; like electrical boxes with four outlets, or extension cords with three inputs? Speaking of extension cords, are receptacles positioned near the equipment set-up or will several extension cords be needed?

LIGHT CONDITIONS

Are the light controls near the projection equipment or will it be necessary to run over to darken the room, and then run back to the projection devices? Possibly, someone else may be needed to man the switches. If there are dimmers, set a light level which will allow the audience to see the slides and films, while at the same time enabling them to read all the copy and allowing the presenter to see his audience. If there are no dimmers (only on/off switches), set up lamps around the room, in such a



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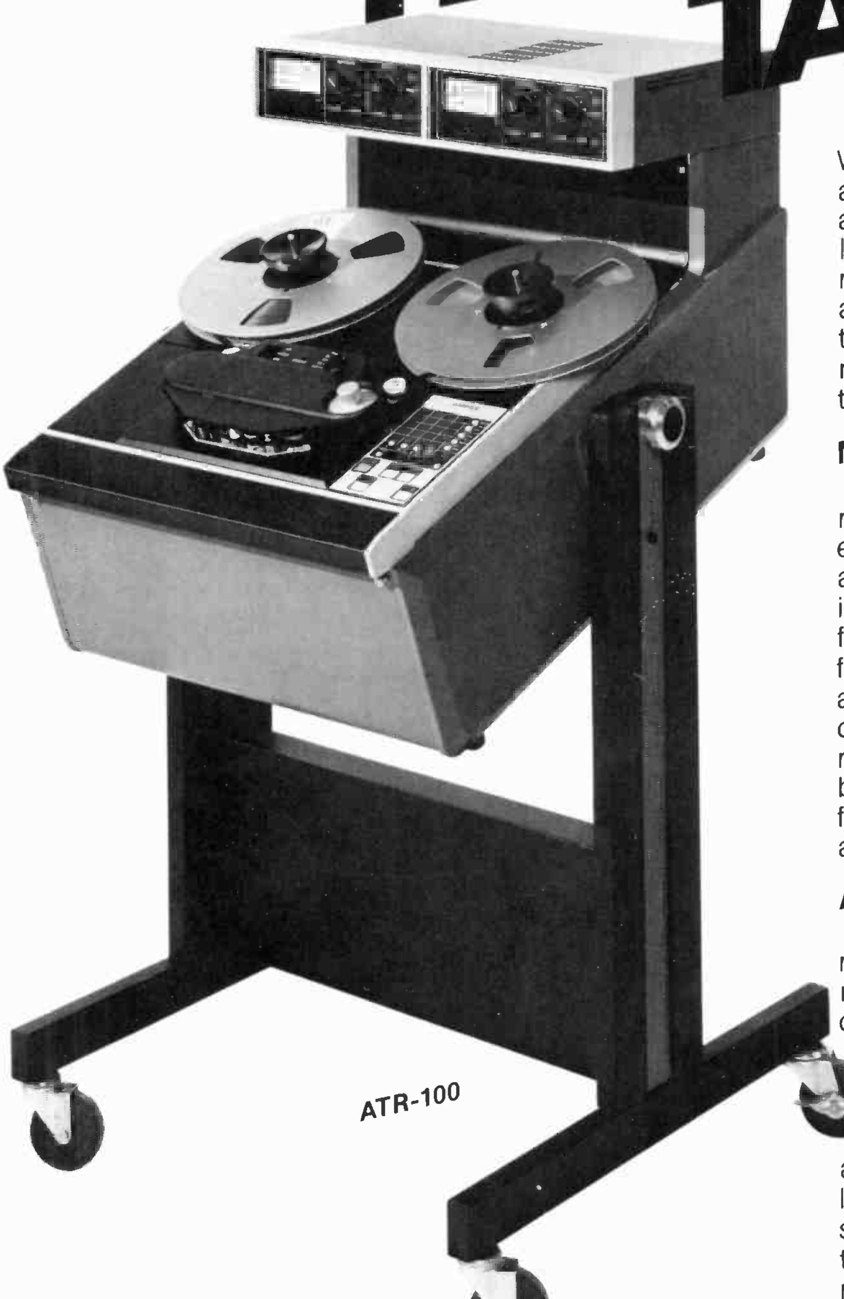
WHAT IT TAKES TO REALLY HEAR THE TALENT



ATR-100

When you hear recording engineers talk about the "Ampex Sound," they're talking about the sound of transparency. To the limits of technology, Ampex recorders and mastering tape let the production qualities and the sound of the talent come through the recording process intact. Here's a brief review of Ampex professional products for the serious studio...

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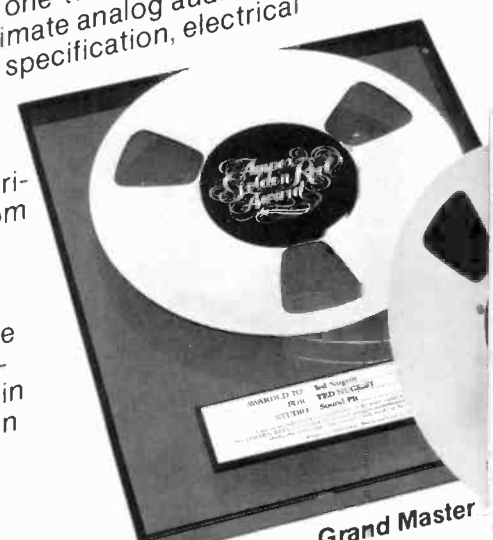
Ampex designed the MM-1200 as a money-making proposition for studio owners. It'll work around the clock as group after group lays down the audio, and then it'll keep right on working as you go back for sweetening, for a vocal, or even for a final mix to a video production (with the addition of a video layback head). Quick-change heads make the MM-1200 even more useful—you can go back and forth between 8, 16 and 24 channels, and switch from one-inch to two-inch tape as easily as a clarinet player changes reeds.

ATR-100. Mix Down the Ampex Sound

This modular one-two-or-four-track machine is the ultimate analog audio tape recorder. In every specification, electrical or mechanical, it **is** the state of the art. Tape handling superiority comes from a unique closed-loop triple servo system that protects your valuable masters. Convenience features begin with the wide-open



Grand Master

way as to keep the front of the room dark (for the sake of the screen image), while leaving some light in the rear of the room. When setting up for the presentation, *never* have a presentation in a totally dark room. (And make sure there is a podium light so the presenter can see his notes. The reflected light will also help illuminate the presenter to some extent.) A daytime presentation, of course, has its own unique problems —always look for curtains on the windows, or venetian blinds. It's almost impossible to show slides or films on a normal screen in broad daylight with the sunlight streaming in. If you survey the room late in the day, or at night, remember that the presentation might be given during the day.

PROJECTION BOOTH

The mention of the projection booth brings up several points to consider. First, if there is no projection booth, the equipment will have to be set up in the conference room. If there is a built-in screen, the positioning of the equipment is pretty well determined—unless, of course, you determine that it would be better to set up your own screen at a different position. (Sometimes this is the smart thing to do in spite of the extra trouble and work.) By repositioning the screen, you can then set up the projectors at a more convenient position. If you're using the built-in screen, and you're shooting over the heads of the audience, make sure you raise the equipment sufficiently over their heads and that no obstruction is in the way. Project a sampling of your slides or films and sit in the questionable seats. Reposition the equipment and/or the chair to avoid any silhouettes on the image.

If there is a projection booth, make sure there are ample provisions for you, the projectionist, to hear what's going on outside in the conference room. It's a lot smoother to hear the cue for a film than to guess and goof. If there is no monitoring system, keeping a door open will probably allow you to hear the speaker. In any event, be sure you can hear what's going on. You might have to have someone throw you a finger cue from the doorway, so prepare yourself for whatever is needed. The presenter and the audience will appreciate it. and you'll look like a genius.

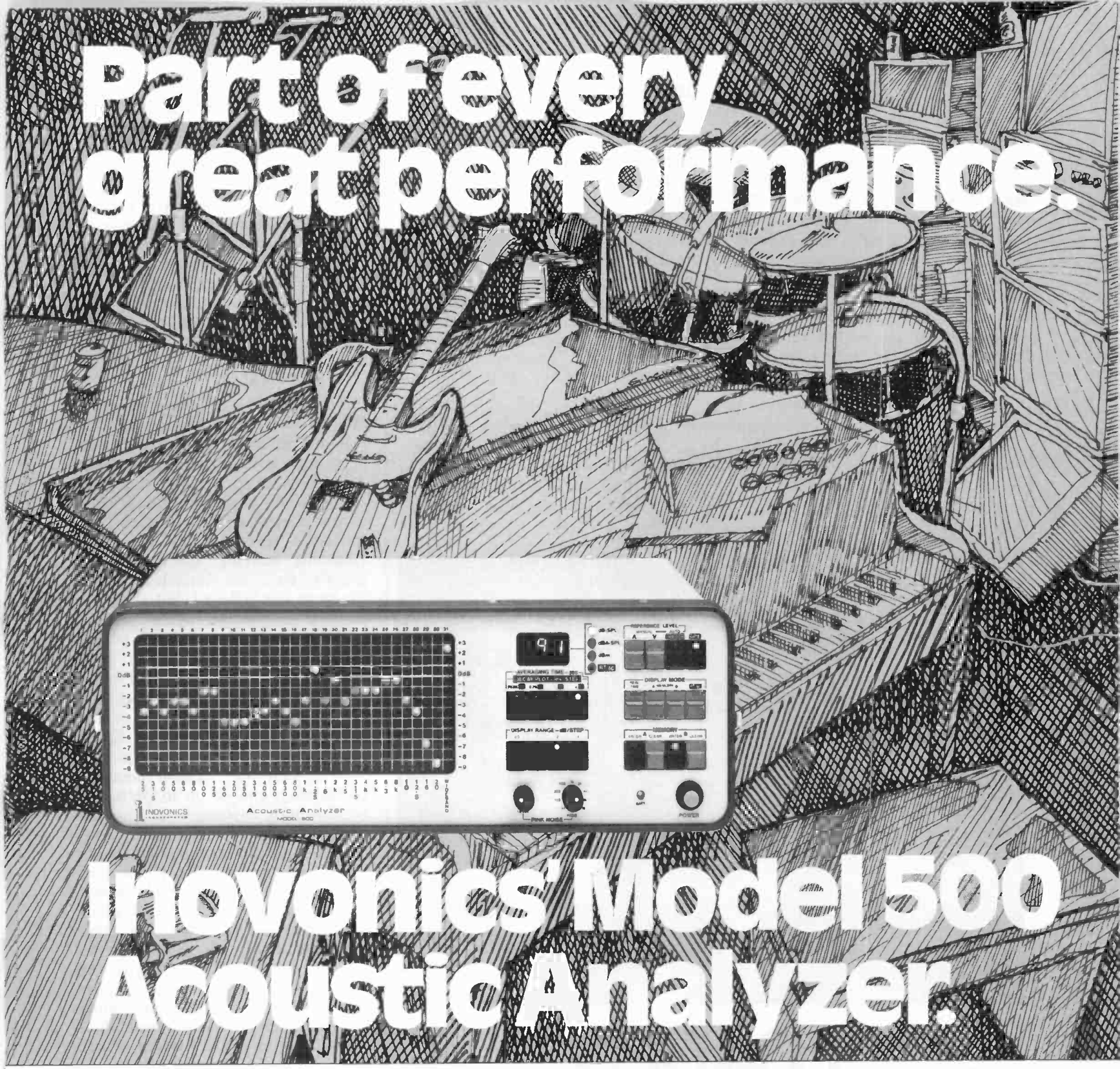
POSITIONING EQUIPMENT

Another advantage of the booth over the conference room set-up is that projector noise can't be heard with the equipment in a booth. If you have to set up within the conference room, try to avoid positioning the projectors in the middle of the room. Not only will you block the view of those behind you, but the noise will distract those closest to you—and it is extremely annoying. Get yourself as far as possible toward the back of the room, behind all the seats, if possible. Not only will it improve viewing conditions, but you'll find yourself with more room to move around. Now that you are back out of the way, you will need to know the size of the screen so you can determine the proper lens for slides and film. There are printed tables you can use as guides. Be sure to appropriate the lenses you determined necessary, for proper throw, and maybe one or two others around those just in case.

ESSENTIAL ACCESSORIES

The carrying of spare bulbs, pins (push pins for posting up material on the wall), batteries, and tape are part of the a/v specialist's "survival kit." Batteries are needed for any equip-

Part of every great performance.



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"See us at AES Booth #54"

ment that may use them—such as r.f. mikes or portable tape recorders or portable video equipment. The bulbs are, obviously, for replacement (although you know you should always start any presentation with brand new or very fresh bulbs). The tape is to lay over any cables that are run around the room. This includes audio cable to wall connectors, a.c. cables for power, remote control cables for the speaker to operate his own slides, etc. In addition, all connections, within the cable run, should be taped to prevent them from loosening during audience movement.

OTHER CONSIDERATIONS

You may only be responsible for the projection equipment, but it might be smart to judge whether the speaker might need a microphone. Question the client if he wants one . . . question the local houseman or electrician if they have one and alert them to set it up if the client wants it. Check the sound system and how you can feed your film projection sound and/or tape recorder into it. If there isn't a sound system available, you will need one. Sound from the film projector should always come from the front of the room where the screen is or even from overhead loudspeakers in the room's system, but never from the back where the projector is. As an emergency measure, in case there is some trouble with the sound system, or with the one you have set up, pull the plug from the projector output and let the sound come from the loudspeaker in the projector—but that is strictly in emergencies. With the film image in the front, it can really sound bad when sound comes from the back. Do your best to avoid it.

A/V SURVIVAL KIT

The afore-mentioned pamphlet offers several suggestions that might be included in a "survival kit." In the a/v line, such items as extension cords (meaning a.c., slide remote, audio, etc.), spare projection bulbs, exciter lamps, emergency splicer (both for audio tape and film), and tape (meaning masking, splicing, Scotch, gaffer's, etc.) are included. Among the items not included, but which are essential to smooth operation, are: take up reels (for audio tape and film), blank slides, from/to audio adapters (such as from ¼-inch phone plug-to-RCA phono plug, phone plug-to-mini-plug, phono plug-to-mini. from female mini-to-female phono, female ¼-inch-to-phono plug, phone plug-to-3 pin cannon male, cannon female-to-phone plug, and so on to permit interconnecting anything-to-anything including any audio device to any house system input). There should also be "Y" connectors such as phone plug-to-2 phone females, female phono-to-2 male phono, etc. These permit connecting one cable to two others (the normal function of the "Y") and also to act as "feed-throughs" between similar connectors. For example, a male phone-to-2 phone female "Y" can be used to join 2 male phone plugs by using the 2 females without using the male on the "Y".

All, or most, of this material might seem basic to the expert, seasoned a/v specialist, but consider some of the advantages of reviewing this information. It is usually true, for example, that rented audio tape recorders and film projectors come with take-up reels. Film projectors usually come with the largest reel size available. This may prove helpful if the film to be shown is already put together in one long reel, or if the individual small films can be taped together during the presentation to save time in re-threading successive films. However, rewinding the film to return the original take up reel with the projector takes time. It might be more convenient to bring your own take-up reels so that the film can be taken away tails-out and rewound back at the shop or office.

AUTO-LOAD OR MANUAL

Consider even such a minor item as which type of film projector to rent or bring. The auto-load, or self-threader, type is the most commonly used. It makes it easy to get the film on the projector properly without too much work. However, if the film snaps or breaks, it might be easier to manipulate the film past the break with the old-fashioned manual loading type of projector where the film can be handled along its entire path. This is also true in the event a film reel is stopped in the middle and the rest of the film is not shown. With the auto-load, the film has to be rewound through the path (a slow process) or removed from the projector manually (a tricky and tedious job). With the manual load, the film can be easily removed from the running path and set up for rewind. There are models of auto-loads available which permit the removal of the film easily—simply by releasing the holding mechanism. These models solve some of the above problems. These, and the manual loading types, can also solve another rather annoying problem. If the projector is in the same room as the meeting, and there is more than one film to be shown, the use of some types of auto-loads make it necessary to run the successive films through the path to cue up the film. This makes for annoying noise during the meeting. The "release" type of self-threader and the manual-load machines solve this problem by permitting the projectionist to set up the next film without running the motor each time. Separate take-up reels can be used, or films can be taped together in succession past the projector's sprocketed rollers. The annoyance of projector motor sound during a quiet portion of the meeting can thus be avoided.

Still think this is all too basic? Maybe, but think about the embarrassment when you rent a film projector for your "out-of-town" presentation and are told that you would be getting a certain model using the standard ¼-inch speaker output, only to find that, when you arrive—only a short while before the presentation was to take place—that the last one of that type in stock was either on another job or defective. The one you get, is the old type which utilizes the smaller Bell-&-Howell output connector, and, of course, the supplier has no cables to lend you. If you stacked your "survival kit" properly, you brought along just such a plug mated to a standard ¼-inch or phono plug, and even, maybe, a phone plug mated in a "Y" to two alligator clips. Either of these 2 cables, or both together, can save the day and give you a feed to a separate loudspeaker or sound system. Along with standard female-to-male cables, the length of any cable can be extended for any desired distance. It's a great feeling to come in with a *complete* "survival kit" so you can shrug your shoulders at any emergency.

And to complete your "bag of tricks," you might also include a remote switch cable to allow you to start a film projector from some distance away from the machine, a "Y" to connect 2 slide projectors to operate simultaneously in side-by-side projection from one remote control device, a "Y" to permit two speakers to operate the same slide projector in a dual-presentation, and even an extra "clicker" in case the one being used by the presenter fails. Basics, sure, but think of the hero you'll be if you can think of everything beforehand. Remember the old a/v prayer: "If only I could always be fully prepared for the totally unexpected." AMEN! ■

New Products & Services

NOISE FILTER

● A stereo hiss reduction device, the 8800 Dynamic Noise Filter removes noise from tapes, records, and broadcasts without the use of encoding or decoding. A gyrator-type, variable low pass filter provides up to 15 dB hiss reduction.
Mfr: Logical Systems
Price: $249.00
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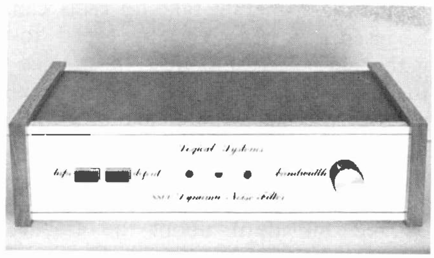

RECORD DE-WARPER

● Sure it looks like a hockey puck, but it's really a record de-warper. Just slip the AudioMate Record De-Warper over the turntable spindle, and its weight flattens and straightens out your warped records.
Mfr: ELPA Marketing Industries, Inc.
Price: Approximately $5.00
Circle 62 on Reader Service Card

SOUND LEVEL METER

● Providing both slow and fast response, the Model CS181A sound level meter utilizes a 1-inch diameter condenser microphone, and has a measuring range from 40 to 130 dBA. Designed to operate for over 100 hours on a single set of standard 9-volt batteries, the meter may be coupled with a chart recorder. The CS181A is available separately, or with a kit, including a calibrator, windscreen, spare batteries, manual, and a carrying case.
Mfr: Castle Associates
Price: Model CS181A, $218.00
Kit, $180.00 additional
Circle 63 on Reader Service Card

PARAMETRIC EQUALIZER

● Each band, in the PQ-6 stereo parametric equalizer, can be boosted up to 20 dB, or cut to complete cancellation (more than 40 dB). The unit features three continuously variable and broadly over-lapping frequency controls per channel, with a bandwidth of 1/3 to 4 octaves, boosting; and 1/10 to 1 octave cutting. Frequency ranges: bass 25-500 Hz, midrange 150-2500 Hz, treble 600-10,000 Hz. Each channel has a separate bypass switch, and all IC's are socket mounted for easy serviceability.
Mfr: Furman Sound, Inc.
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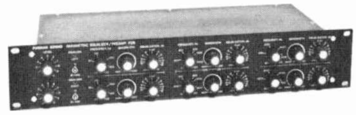
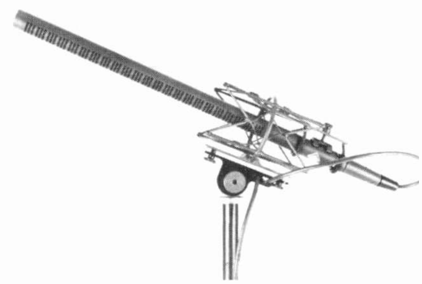
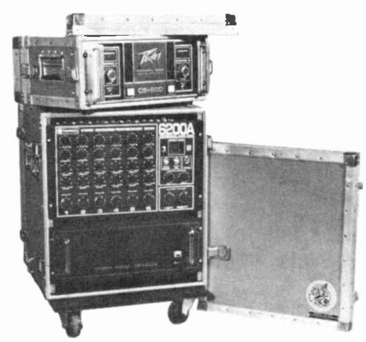


CONDENSER MICROPHONE



● Displaying a smooth frequency response, the Model KMR-82i is an ultra-directional ("shot-gun") line condenser microphone. The unit is less susceptible to off-axis sound coloration as a result of a directional pattern which differentiates pattern *vs* frequency, less severely than traditional models. Accessories for the KMR-82i include a foam wind screen, elastic suspension, wind-proof "blimp" and a unique "active handle" for hand held use, containing the 9V battery for the 48V phantom powering converter.
Mfr: Neumann Company
Price: $795.00
Circle 66 on Reader Service Card

RACK MOUNT CASES



● Providing protection and ease in transport, rack mount cases are available in a variety of designs (clamp-on lid, case-within-a-case, pullover clamp lid, etc.). Each case is supplied with an ample amount of 10/32" chrome-plated screws and nylon washers for mounting.
Mfr: Anvil Cases
Circle 65 on Reader Service Card

PHASEMETER

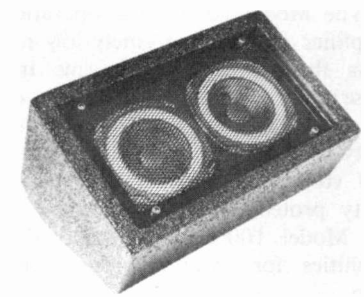
● "No hands" operation describes the Model 6200A 3½ digit phasemeter. With 0.5 degree accuracy, and 0.1 degree resolution, the operation is automatic over a frequency range of 10 Hz to 1 mHz, with input voltages from 0.1V to 120V rms. It accepts sine and square waveforms, and has an additional analog output.

Mfr: Krohn-Hite Corporation
Price: $795.00
Circle 67 on Reader Service Card

MONITOR

● Incorporating two 4½-inch pad precision transducers driven by large 34 ounce magnet structures, the Minimonitor SP-245 claims a frequency response that is flat within 2½ dB from 80 Hz to 13.5 kHz. Power handling is rated at 120 watts rms program at 16 OHM impedance. Overall dimensions for this mini-floor monitor are: 12" wide, 7½" high, 6½" deep.

Mfr: Concertaudio Manufacturing Research Corporation
Price: $99.00
Circle 68 on Reader Service Card



TIME CODE READER

● Reading and display of Time and User Data at search speeds from hand turn reel speeds to more than 40 times play speed, drop frame indicator, and detection and bypass of Code errors are among the standard features included in the Model TCR-80 Time Code Reader. The reader selects and holds in memory an edit point code without stopping the machine, and holds in memory the last address when the machine stops. Options, such as a Data Output Module and a Video Character Generator, can be installed on a plug in basis.

Mfr: Skotel Corp.
Price: $2100 to $2985 depending on options
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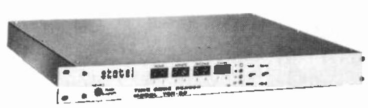
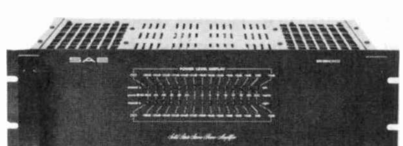


POWER AMPLIFIER

● With full complementary circuits to ensure maximum power output, the Model 2300 power amplifier delivers 150 watts per channel minimum RMS, into 8 ohms, from 20 Hz to 20 kHz, with no more than 0.05% THD. The Model 2300 includes full thermal protection, signal relays to protect speakers from low frequency information which might damage them and electronic protection for transient overloads and short circuits.

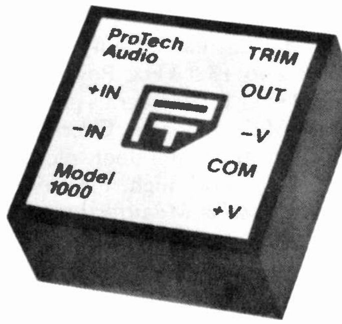
Mfr: Scientific Audio Electronics, Inc.
Price: $700.00
Circle 70 on Reader Service Card



LOUDSPEAKER ENCLOSURES

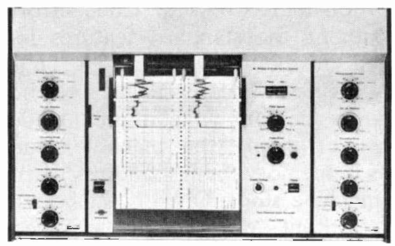
● Compact, protective loudspeaker enclosures, incorporating E-V transducers and a variety of high-end components, combine the functions of both wooden enclosures and foam-lined carrying cases. The enclosures are constructed from rugged, low-density material with low resonance characteristics. The panel holding the speakers is suspended, to protect the speakers from direct contact with damaging outside impact.
Mfr: Amanita Sound Inc.
Price: $263.75
Circle 76 on Reader Service Card

OPERATIONAL AMPLIFIER



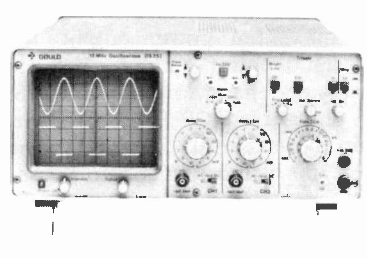
● The Model 1000 audio operational amplifier features extremely low noise (less than 0.5 microvolt rms input noise) and distortion (0.1 per cent thd @ +20 dBm), high output capability, and fast slewing characteristics (13 volts/microsecond). Reverse polarity protection is incorporated into the Model 1000, with additional capabilities for external offset voltage trim.
Mfr: ProTech Audio Co.
Price: $24.00
Circle 77 on Reader Service Card

LEVEL RECORDER



● Designed for accurate field and laboratory recording, the 2309 portable two-channel graphic level recorder offers logarithmic a.c. recording in the frequency range 1.6 Hz to 20 kHz; four switch selected recording modes; two dynamic ranges, 25 and 50 dB; eight fixed paper speeds from 0.01 to 30 mm/sec.; and a built-in calibration voltage. Other recording features include: reversible paper drive, automatic stop after one chart length, and automatic pen lift.
Mfr: B & K Instruments, Inc.
Circle 78 on Reader Service Card

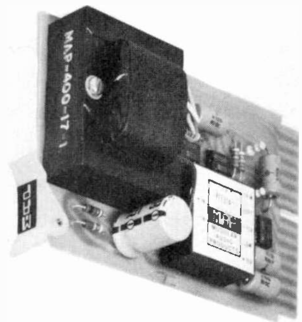
DUAL-TRACE OSCILLOSCOPE



● With dual-trace and X-Y display capability, the Model OS253 12 MHz oscilloscope features 2mV/cm vertical sensitivity with a.c., ground and d.c. coupling; channel sum and difference with channel 2 inversion; bright-line operation; d.c. coupled Z-modulation input; calibrator output and a front-panel trace-rotate control. Horizontal sweep rates are continuously variable over 18 ranges from 500 ns/cm to 0.2 s/cm, with a maximum effective sweep rate of 100 ns/cm at X5 expand. The unit employs an 8 x 10 cm CRT for display.
Mfr: Gould Inc.
Price: $695.00
Circle 79 on Reader Service Card

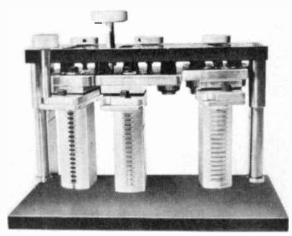
OSCILLATOR CARD

● Serving as a convenient source of test, alignment, cue, or slating tones, the IMPAC Series Model 4012 Oscillator has a built-in output transformer which provides up to +27 dBm into a 600 ohm load. The output frequency of the 4012 may be externally set or remotely varied from 40 Hz to 20 kHz.
Mfr: Modular Audio Products
Price: $105.00
Circle 71 on Reader Service Card



HEAD MOUNTING ASSEMBLY

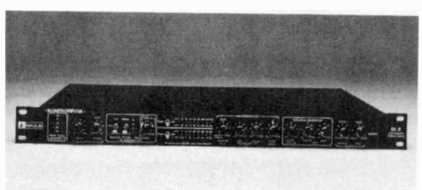
● Designed to reduce alignment time as well as simplify magnetic head maintenance, the Promix I multitrack head mounting assembly provides complete control over all aspects of head alignment. A built-in head subplate facilitates removal of an individual head to change its configuration, relap, or replace it without seriously affecting the alignment of the head. The Promix I is designed to fit most studio recorders currently in use, and can be customized for special applications.
Mfr: Grandy, Inc.
Circle 72 on Reader Service Card



● Available with fixed or detachable cable/fan, the Hydra line of standard and custom mike snakes utilizes Switchcraft connectors for all the individual mike lines and sends. AMP CPC, M, or G series connector shells with gold plated pins and sockets are used with the multi-pair cable interconnections, depending on requirements.
Mfr: Dimension Five Sound
Circle 75 on Reader Service Card

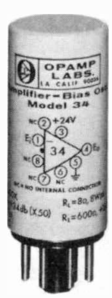
SIGNAL PROCESSOR

● A combination digital-delay/special effects processor, the DL-2 Acousticomputer provides echo, ADT, chorusing, vibrato, flanging and reverb functions. Housing a built-in VCO, the DL-2 features a 20 Hz to 15 kHz bandwidth at all delay settings, 90 dB dynamic range, delays up to 240 ms in serial (mono) mode and two independent channels.
Mfr: DeltaLab Research, Inc.
Price: $1500.00
Circle 73 on Reader Service Card

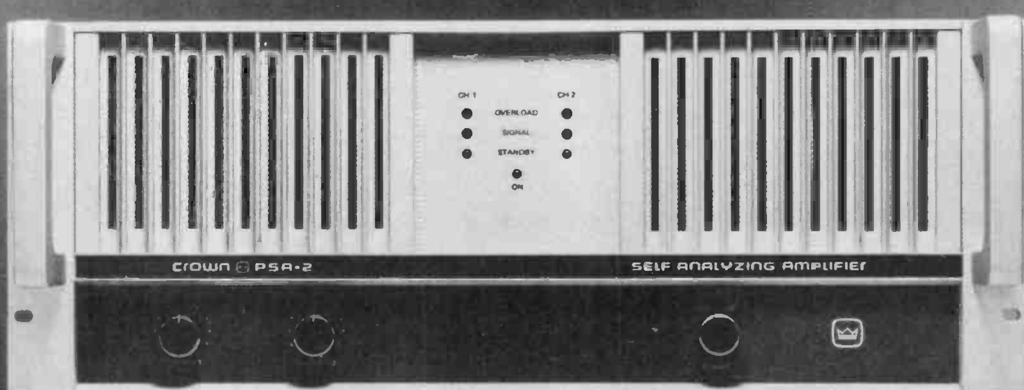


AMPLIFIER/BIAS OSCILLATOR

● A dual purpose module, the Model 34 Audio Amplifier-100 kHz Magnetic Tape Bias Oscillator-Buffer, can be employed for general purpose signal processing, earphone-speaker power amplification, distribution, combine, or microphone amplifier and magnetic tape erase-record service. The module, as a 100KC oscillator, has a 20V rms output at 1K ohm load; and a frequency response of 20 Hz to 20 kHz (±2 dB), 0.25 per cent thd (+18 dBm) as an amplifier.
Mfr: Opamp Labs Inc.
Price: $25.00
Circle 74 on Reader Service Card



Five years from now,



The Crown PSA-2 Professional Power Amplifier*

220 watts per channel minimum RMS (both channels operating) into an 8 ohm load, 20 Hz-20 KHz at a rated RMS sum total harmonic distortion of 0.05% of the fundamental output voltage. (tested per FTC specifications)

250 watts ±1dB per channel, 20 Hz to 20 KHz into 8 ohms with no more than 1.0% THD (EIA Std. SE-101-A).

400 watts ±1dB per channel, 20 Hz to 20 KHz into 4 ohms with no more than 1.0% THD (EIA Std. SE-101-A).

685 watts ±1dB at 1K per channel into 2 ohms, with no more than 1.0% THD.

*Designed for installation and use in professional sound systems

Editorial

THESE PARAGRAPHS ARE BEING TYPED as we walk out the door (well, almost) on the way to JFK airport, to catch the big bird to Brussels, Belgium. Our excuse is the March convention of the Audio Engineering Society—which will of course be long-since over by the time this little editorial catches up with its readers.

In fact—given the vagaries of the postal service—you're probably reading this as we sort out our notes from the March convention, and get ourselves ready for the next one (May 15-18, Los Angeles).

Needless to say, there were (or will be) lots of new audio goodies on display at both conventions. And that's about as good a way as any to bring up the subject of test equipment. For with all that fancy new recording hardware coming our way, how do we really know we're getting the most out of it? Obviously, something more than an oscillator and a VTVM is needed. (Also, it's just not "in" anymore to tap an i.c. with a screwdriver to see if it's working properly.)

There's a new generation of test equipment out there, capable of keeping tabs on the new generation of pro' audio hardware, or for that matter, of helping you to diagnose the ills of the old generation.

Not so long ago, test equipment was regarded almost with some sort of suspicion. Engineers boasted of being able to hear (and quickly fix) just about anything that went wrong. Fair enough—sometimes. Even now, there's nothing terribly subtle about some of the grosser types of distortion. But, as audio gear gets more and more complex, it's not so easy any more to keep everything ship shape with just your golden ears and the butt end of a screwdriver.

No; all that new test equipment is there for a purpose, and a good one, too—to help *you* get, and maintain, better audio performance. Naturally, some of the more-sophisticated equipment and procedures are geared towards the manufacturer. With better test equipment, he designs a better product (we hope!). For routine test and maintenance in the studio, you may not have the budget—or the need—for such complexity.

But, as technology progresses, a lot of formerly out-of-reach equipment may now be reconsidered by the man behind the console who's wondering why things don't always sound quite right. In short, there's no longer much reason to deprive yourself of the test gear you need to maintain the equipment in your studio.

Our feature articles this month highlight some of the latest developments in this critical area. Whether you're ready to buy, "just browsing," or not even thinking about test equipment, we hope you'll find something of interest here. And if you are not even thinking about test equipment, we offer this suggestion: think again!

And while you're thinking about it, check out Sidney Silver's feature story on delta modulation. It's just one more contender in the expanding world of digital audio.

Speaking of expanding (nice segue, don't you think?), that's what we've been doing here at **db**. We hope you've noticed that each issue has focused more-or-less exclusively on a single topic. And, we hope you like the idea. Frankly, we sometimes have second thoughts about it ourselves. Like, when that promised article on the new whatchamacallit doesn't show up on time, leaving us with a gaping hole that must be filled—and quickly, too! One of the principle tasks of our associate editors is to get out there and terrorize authors into meeting deadlines.

Although it's a bit too early to get cocky about it, we seem to be settling into a routine, and now even have some idea of what the next two issues will be about (I don't believe *that* for a minute—Publ.)

But, as we grow, we'd like to hear more from you. What would you like to see more (or less) of? Are the features too complicated?—not complicated enough?—just right?—'way out in left field? Tell us what you think.

We're intrigued (or is it terrified?) by the computerization of pro' audio. We plan to spend a good deal of time trying to sort out what's happening with this new technology, and keeping you informed about it. If you're looking for this sort of information, stick around.

Like everything else in electronics, prices come down as mass production goes up, and the power of the computer is fast coming within reach of even the most budget-conscious studio operation. We're interested—are you?

J.M.W.

WAYNE JONES

Audio Tests and Measurements—Part I

New equipment and new techniques to keep pace with the state-of-the-art in recording hardware.

IT'S NO SECRET that the audio industry in general has undergone a phenomenal growth in the last two decades. The growth has been in two dimensions; quantity and quality. The increase in quantity has been for two reasons: more people are using more equipment, and, contemporary recording techniques require more channels (and therefore, more equipment). The increase in quality has followed the growth of technology itself. As the state-of-the-art in electronics has advanced, so too has the audio industry, and its requirements for ever-greater performance.

The evolution in recording hardware has placed somewhat-conflicting requirements on the parallel evolution of test and measurement equipment. On the one hand, the increase in quantity requires faster- and therefore simpler-methods of testing. On the other hand, the increase in quality requires more demanding- and therefore more complex-methods of testing. These seemingly-contradictory demands on test equipment have produced a new generation of instruments and procedures.

The industry constantly demands improved audio performance. The development engineer advances the state-of-the-art each time he develops a new piece of equipment. The manufacturing facility tries to maintain these standards with more-rigorous production test facilities. The end-user must perform more sophisticated tests, on more signal paths, more often than ever before, to verify the standards of the equipment that were set in the development lab. The common denominator for all these situations is; more data, faster.

Wayne Jones is president of Amber Electro Design, Ltd., Montreal, Canada.

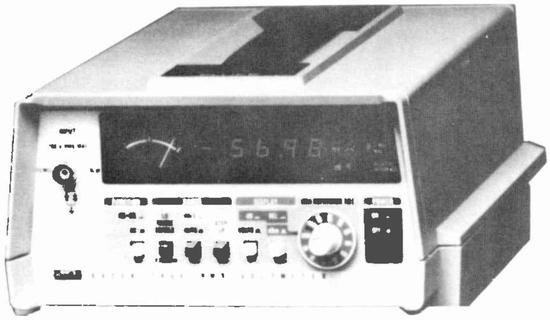
THE "NEW BREED" OF INSTRUMENTATION

Test equipment manufacturers have met the demand with a "new breed" of sophisticated instrumentation, providing higher performance and faster operation. At the same time, the new test-gear is simpler to operate and yet it presents the data in a more concise fashion. The simplicity is in the use and presentation of data; the complexity is in the internal structure of the instrument. Let's take a look at some of this new equipment, and the new techniques, as they apply to audio measurements.

AMPLITUDE MEASUREMENTS

Amplitude, or level, measurement is perhaps the most basic of audio test procedures. The changes seen here are in a number of areas. Read-outs have become digital—they're faster to see, more accurate, and not subject to "interpretation." Another area of improvement has been in detector characteristics. With previous techniques, these were dependent upon a combination of inaccurate rectifiers and unpredictable mechanical ballistics of the measuring meter. With the newer instruments these characteristics are precisely defined by the electronic circuit parameters. Many provide true rms detection, which gives more-meaningful results to measurements of noise and of complex waveforms. Many of the new instruments provide auto-ranging; a tremendous time-saver, and a perfect example of instrument complexity providing simplicity of operation.

Usually, level meters measure amplitude by measuring signal voltage. The results are displayed in volts, dBm,



Fluke 8920A true rms voltmeter. A broadband, true rms digital voltmeter with readout in volts, dBm or dB. A display resolution of 0.01 dB and auto-ranging over 132 dB provide a highly accurate, automatic method of measuring audio signal levels. dBm readings can be referenced to 0.775 volts across 600 ohms or 11 other user selectable impedances. A "relative dB" mode lets the user select any signal level as the 0 dB reference for future measurements. Wide bandwidth, high accuracy true rms readings are made possible by a Fluke designed hybrid rms converter. An available option provides a logarithmic d.c. output to drive XY recorders.

dBv, or some other unit. Some instruments can display the results as power in watts. Using Ohms Law, they take the measured voltage, square it, and divide it by a user-defined load impedance. In addition to dBm readings across a standard impedance of 600 ohms, some instruments provide the ability to measure in dBm across other not-so-standard impedances, such as 150 ohms. Another feature is the ability to set an arbitrary zero reference level. All future amplitudes read plus, or minus, so many dB of this new reference level.

The decibel relationship between average, peak and rms values of a pure sine wave.

	Avg.	Peak	Pk-to-Pk	rms
Average	0 dB	+3.93 dB	+9.95 dB	+0.91 dB
Peak	—3.93 dB	0 dB	—6.02 dB	—3.01 dB
Pk-to-Pk	—9.95 dB	—6.02 dB	0 dB	—9.04 dB
rms	—0.92 dB	+3.01 dB	+9.03 dB	0 dB

Example: If an average responding, average calibrated meter indicates a signal amplitude of +8 dBm, a peak-responding, peak-calibrated meter would indicate +11.93 dB, while a true rms meter +8.91 dBm, etc. For waveforms other than sine, the differences can be greater.

NOISE MEASUREMENTS

As a particuar case of ampiltude measurements, noise measurements have always been a source of controversy. There are numerous methods of measuring noise. Variables include; detector response characteristics, measurement bandwidth, weighting, termination characteristics and perhaps several adjustable parameters of the device under test. There are a number of conventions that have been adopted for noise measurements. Consoles and similar equipment are usually measured with a flat audio bandwidth weighting; that is, a filter with a flat response over the bandwidth of 20 Hz to 20 kHz, and rapid attenuation outside these areas. Tape recorders are often measured using an ANSI/IEC "A" standard weighting network. Equally important is the detector time constant and characteristic. The detector can be true rms, average- or peak-responding. It can have fast or slow time constants. A weighting network that has become popular recently is the CCIR network. Dolby Laboratories has suggested the use of the term CCIR-ARM, referring to the use of CCIR weighting with an average-responding meter.

Noise measurement techniques have been established for a number of reasons. Some of them are technical—others are practical. Until recently, true rms-type meters were expensive and hard to find. Therefore, practical considerations dictated the use of average-responding meters, something easier to achieve. However, the recent introduction of several low cost true rms meters will go a long way to facilitate noise measurements using this improved kind of detection characteristic.

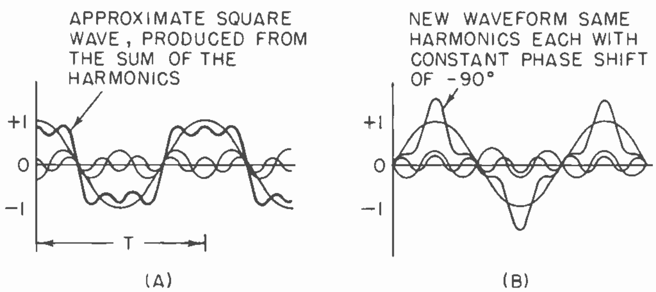
There are a variety of arguments supporting the requirements for making noise measurements using particular types of detection characteristics, weighting and so on. The most important thing however, is that the final measurement be qualified as to how it was made. The bandwidth of the measuring device, the detector characteristics and the nature of any weighting filters used must be specified, along with the actual noise figure. In addition, all of the relative parameters of the device under test must be specified, such as the termination impedance of its input and output, the gain, and any other parameters that would affect the noise measurement. If a user is attempting to duplicate noise measurements made by someone else, his meter must have identical parameters. This suggests that, the more flexible the noise meter is in terms of selection of bandwidth, weighting and detector characteristics, the easier it will be to duplicate noise measurements.

NOISE MEASUREMENT AND SPECTRUM ANALYSIS

An extension of noise measurement technique involves spectrum analysis. Rather than giving the single number representing the total energy in the broad bandwidth, a spectrum analyzer gives a graph of noise-versus-frequency. This is a more complicated presentation of data, but is a great deal more useful in some circumstances. For example, it helps to locate the source of noise, and will readily identify crosstalk, hum, or oscillation as noise sources. If the broadband noise figure is too high, the next question is usually, why? Spectral analysis of noise energy over the frequency band of interest greatly facilitates finding the noise source and answering this question. However, a difficulty with spectrum analysis of this nature is that it is almost impossible to derive simple comparative results between two spectrum analysis graphs. One type of noise may sound worse than another type of noise, although both have the same broadband noise figure. Spectrum analysis of the noise floor will indicate the differences in characteristics.

FREQUENCY RESPONSE

Frequency response adds a second dimension to the basic measurement of amplitude. It involves the measure of gain or loss of the device under test at several discrete points across the bandwidth of interest. The more points measured, the better the resolution of the result. The ultimate is an infinity of points which results in a sweep, and



How linear distortion or random phase shift in a transmission medium can modify a waveform.

(A)

Original waveform at the input of the transmission medium. The heavy trace is an approximation of a square wave, composed of a fundamental, 3rd harmonic and 5th harmonic with the illustrated phase and amplitude relationships.

(B)

New waveform at the output of the transmission medium. Note the significant change of shape as a result of a constant phase shift of —90° of the fundamental and its harmonics. No amplitude change has occurred.

herein lies one of the more significant improvements in contemporary measurement technique. The ability of a test instrument to sweep—and thus acquire a large amount of data in a short time—allows the presentation of data to the user in a concise and complete format. To paraphase a cliché, one graph is worth a thousand measurements.

DATA STORAGE—HARD COPY OR CRT DISPLAY

But generating a frequency response graph requires more than just a sweep generator. Since low frequency sweeps require several seconds to acquire the complete picture, some means of storing the information is needed, so the whole graph can be seen at one time. There are several techniques available to do this. One is to use a strip-chart or XY recorder and plot the information on graph paper. Another method uses digital storage, with presentation on a conventional CRT or oscilloscope. The first method has recently seen a remarkable reduction in price. The newer devices are inexpensive, easy to operate and portable. The digital storage method is operationally simple and allows convenient manipulation of data after acquisition. Presentation may either be directly on an oscilloscope or, on some units, may be later transferred to paper using an XY recorder. Hard copy can also be obtained by using a Polaroid or other type of photograph of the CRT screen. This applies to the analog storage, the long persistance CRT, and the digital storage method.

PHASE RESPONSE

In addition to the amplitude frequency response graph, another graph that is particularly interesting is phase response. In this case the phase shift between input and output of a device under test or the phase shift between two signals is plotted against frequency. Often this information is more useful than the amplitude/frequency response, especially in multi-channel systems. Phase roll-off can show up much sooner than amplitude roll-off and, in multi-channel systems, it can have a more significant effect on the final result. The combined effect can produce severe roll-off, affect transient behavior, and cause other abnormalities more significant than expected from observing the few-dB roll-off characteristic of the system under test.

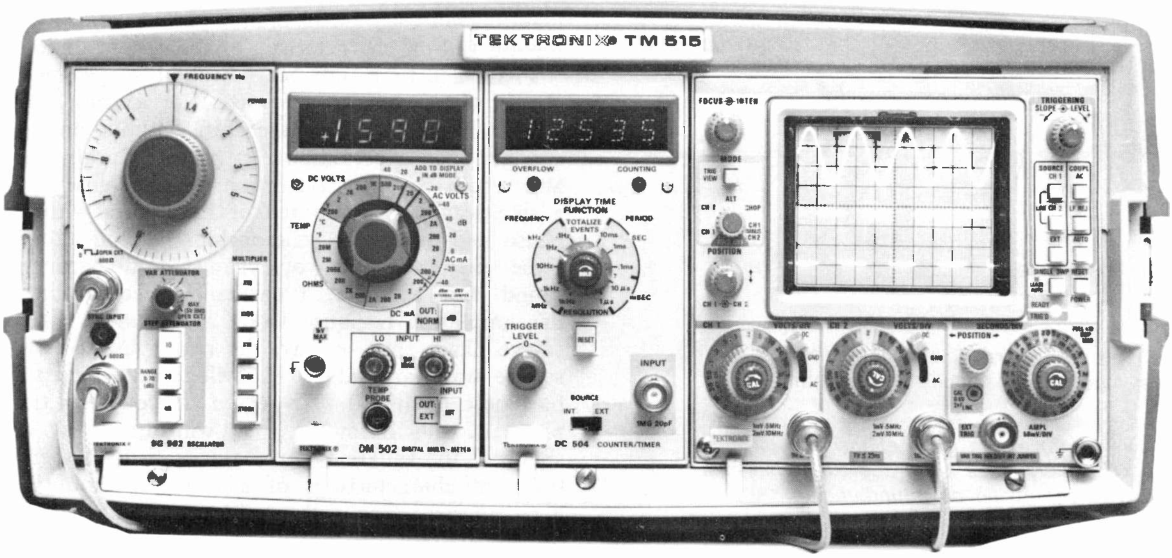
Phase and time shift can be found in systems that have a very flat frequency response. For exampe, an all-pass filter can produce a phase shift without any amplitude change. Such a condition can occur naturally in systems, and results in linear distortion. Every signal (other than a sine wave) is composed of harmonics with particular phase and amplitude relationships. If you alter either of these characteristics, you change the character of the sound. For example, a particular medium may have a flat amplitude response from 20 Hz to 20 kHz, but a severe phase shift in, say, the region around 10 kHz. A square wave around 3 kHz will have a high third-order harmonic in the 10 kHz region. If you alter the relative phase of this harmonic, you will drastically affect the shape of the square wave. Or, to put it another way, the transient characteristics of a fast rise time signal have been altered. This will produce a significant change in the subjective audio quality.

All systems have a finite amplitude bandwidth. At the upper and lower bandwidth limit, phase lag and phase lead will become apparent. A plot of phase shift-versus-frequency will exhibit phase changes long before amplitude changes. The change in slope of such a phase response defines the group delay of the system. A plot of group delay-versus-frequency will usually show two peaks; one in the lower region and another in the higher region of the bandwidth. Such conditions have a pronounced affect on the transient behavior of the system.

Attempts to attain a flat amplitude response are often made with little attention to the effected phase response and time shifts. The flat amplitude response may come at the expense of significant transient response degradation. Filters or equalizers that correct amplitude roll-offs also inject significant phase shifts, and their effects should be investigated, perhaps even more so than amplitude responses. A few dB variation in amplitude/frequency response may cause less linear distortion of complex waveforms than the significant phase shifts introduced by the filters that correct the amplitude response.

Fixed time delay in a multi-channel system causes a continuous sloped phase response. An example of this could be gap scatter in tape heads. The azimuth alignment of a tape head can affect this gap scatter condition, in addition to its normal affect on high frequency response. For this reason it is important to investigate both the amplitude and phase response of a tape recorder when adjusting azimuth alignment. When measuring two channels of a multi-track tape machine, a plot of phase and amplitude differences-versus-frequency will greatly facilitate optimization of the azimuth for the best overall phase and amplitude conditions. The ideal compromise can only be achieved with these conditions known.

Instrumentation for measuring phase and time shifts is becoming more available. Previously, such measurements were somewhat cumbersome to achieve. New instruments permit phase response plots to be generated as easily as amplitude/frequency response plots. By using a fixed time-delay line in the reference path, phase response plots can even be generated of transducers such as speakers. Phase response plots of multi-way speakers generated in the actual listening environment are particularly useful. Multi-way speaker systems, cross-over networks and room equalization systems have a significant effect on the time domain relationships across the audio band. Knowledge of this information in the form of a phase response plot can be very useful as a tool to correct monitor deficiencies.



Tektronix TM-500 series. A family of over 40 modular instruments and six mainframes. Included in the series are digital counters, digital multimeters, pulse generators, function generators, signal processors, oscillators, power supplies, oscilloscopes and digital instruments. Mainframes come in 1, 3, 5, and 6 unit widths including a "carry-on flight case" style 5-unit frame. Of particular interest to the audio fraternity is the DM502 digital multimeter with dB reading capability from —60 dB to +56 dB, the AF501 bandpass filter/amplifier with a range from 3 Hz to 35 kHz and a B.P.Q. of 5 or 15 (selectable), the SG502 oscillator covering 5 Hz to 500 Hz with less than 0.035 per cent harmonic distortion mid-band and the SC502 15 MHz dual trace oscilloscope with sensitivity to 1 mV/division.

OTHER RESPONSE PLOTS

A variation of the amplitude/frequency response plot is the large-signal frequency response. Useful in characterization of power amplifiers, such a plot shows the power output under load-versus-frequency of a power amplifier for a constant input level. The frequency response of such an amplifier at low signal levels, or without a load, may be essentially flat. But the story is often different at full output level. It may have full power at mid-band frequencies, but as high frequencies are approached, and slew rate limiting is encountered, the power response will fall off. This will result in slew-induced distortion or transient intermodulation distortion.

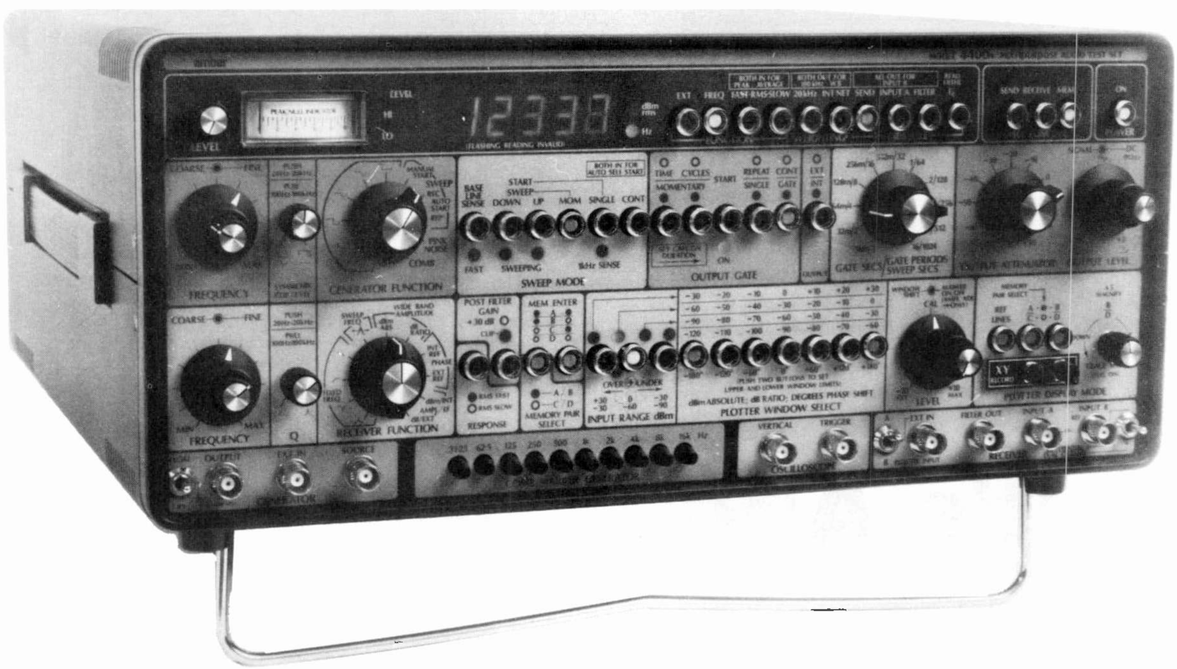
A variation on this theme is the power/frequency response. This is a plot of the power output-versus-frequency, with a constant percentage distortion. Like the large-signal frequency response, it will typically show significant roll-offs at the higher frequencies, due to slew rate limiting of the amplifier. To achieve this curve, the input level to the device under test must be constantly adjusted, to maintain the same total harmonic distortion at the output as the input signal is swept through the frequency band of interest. At the present time, there is no convenient way to do this with a single instrument, but with the increasing availability of automatic equipment it's only a matter of time before an instrument is available that can do this plot with the same ease as a normal frequency response plot.

Power bandwidth can be derived from the power response plot. It is defined as the bandwidth over which the distortion at half power is the same as, or better than, the distortion at mid-band and full power.

ADDITIONAL FEATURES

Other applications for frequency response plots require different methods to achieve the results. In places where the generator is separated from the receiver or plotter by either space or time, some method of synchronizing the two must be established. This is the case in plotting the frequency response of, for example, a tape recorder or a transmission line. Normally the control circuits of the test instrument cause the generator to sweep at the same time as the storage medium fills. In this case however, the generator may be several miles away from the plotter, or the sweep may have been generated several hours (maybe even years) before the plotter was able to receive the sweep. If the final plot is to have a known frequency axis, the plotter must have some means of recognizing the incoming sweep. There are a number of ways to do this. Some instruments actually measure the incoming frequency, using either a frequency counter or some other electronic means, and position the pen or the loading of the digital memory accordingly. Another method is to use a cue tone, typically 1 kHz, at the beginning of the sweep to control the plotting instrument. The former method is useful when there is no control over the sweep speed of the original test sweep. It is also more universal in that it will respond to any sweep over any range within the capabilities of the instrument. However, at times it can have difficulty following a sweep, if the signal is low in level or contains significant noise, such as would be the case with a crosstalk measurement. One solution to this problem is to use a reference sweep at full amplitude to operate the control circuits of the plotter. Another method is to use the second synchronizing scheme described above, where a 1 kHz tone begins the loading of the plotter. Usually, the control circuits can reliably recognize a 1 kHz signal and accurately start the plotter at the beginning of the received sweep.

Markers are a very useful feature on frequency response plots. These allow the inclusion of marks or "glitches" in the plot to identify specific frequencies. This facilitates alignment of the plot on the frequency axis of the graph



Amber 4400A multipurpose audio test set. A comprehensive audio test system containing several generators, analysis and measuring facilities and plotting capabilities. Provides a function, sweep, noise, tone burst, low distortion sine and comb generator with a high power (over +30 dBm) output stage and a 150 dB range, auto-ranging digital dBm meter and frequency counter. Measures true rms, average and peak. Includes an acoustic spectrum analyzer and a digital storage response plotting capability. Up to four plots of amplitude or phase versus time or frequency may be generated and displayed on any standard oscilloscope or transferred to paper with an XY recorder. A variety of unique features and flexible parameter variation permits comprehensive, convenient and fast generation of test results.

paper, and confirms the accuracy of the plotting instrument. Markers may be generated two ways: either in the generated sweep (that is, at the transmit end), or by the plotter itself, on the receive end. In the first method, the sweep generator causes a disturbance in its output signal at the particular marker frequency. This disturbance may be an amplitude change or a momentary pause in the sweep or some other similar effect which will cause a recognizable glitch in the plot. In the second method, the frequency of the receive signal is identified by the plotter, using a circuit such as a frequency counter. A suitable mark is then made on the frequency response plot. The former method is as reliable as the circuit itself, which is usually quite good. The disadvantage however, is that it requires a specialized generator. It is not possible to use an existing test record or test tape. The second method can suffer from some un-reliability, due to the difficulty of accurately measuring the frequency of a signal during a sweep, especially in the presence of noise. If the frequency-measuring circuit is to have a wide dynamic range, then by definition it is also sensitive to noise. But the technique is universal, in that it will respond to a sweep from *any* generator, including those not under the direct control of the user, such as a test record or a test tape. The choice of the method for the generation of markers is a function of the application.

ACOUSTIC FREQUENCY RESPONSE PLOTS

Response plots of acoustic transducers, such as speakers and microphones, are usually handled quite a bit differently than response plots of the electronic devices previously mentioned. It is far more difficult to achieve accurate (or at times even meaningful) results, due to the numerous uncontrollable influences within the acoustic environment. The chief difficulty is to isolate the transducer characteristics from the influencing effects of the environment in which the test is performed. The characteristics and abnormalities of this environment can often have a significant effect on the test results. Sometimes this is not a problem, as in the case of measuring a monitor system where both the environment and the transducer are being tested as a system. In another case, such as the evaluation of a specific microphone or loudspeaker, the effect of the environment must be excluded from the measurement to achieve useful results.

All of the frequency response plots mentioned earlier are generated using a swept sine wave on the generator side and wide-band amplitude measurement on the receive side. Such a method can also be used for measuring a transducer, but is usually applied only when a controlled environment (such as an anechoic chamber) is available. When swept sine waves are used in an uncontrolled environment standing waves are generated, and these significantly affect the response plot. However, an anechoic chamber minimizes these abnormalities, and gives a reasonably-true presentation of the loudspeaker or microphone itself. The effectiveness of the anechoic chamber usually diminishes at lower frequencies, however this difficulty of measuring low frequency response is also prevalent in all other methods of testing transducers. ■

(To be continued)

ALMON H. CLEGG

Three-dimensional Analysis: It's About Space

In which Confucius learns that one more dimension is worth a few thousand more words.

FOR YEARS, audio professionals have tried to present more-and-more data in a more-simplified format. Graphs of amplitude-*vs.*-frequency, distortion-*vs.*-frequency, noise-*vs.*-frequency, or distortion-*vs*-power are some of the more-commonly presented "pictures" used by engineers to conveniently display a collection of data points. For, as Confucius pointed out, "One picture is worth a thousand words." Measured and plotted by hand, or perhaps automatically by test instrumentation especially designed for such use, even Confucius might agree that some of these graphs are worth much more than many thousand words of description.

Over the past few years, it has been more-or-less traditional to show thd-*vs.*-power at a single frequency, such as 1 kHz, on a single graph. Sometimes, other frequencies, such as 20 Hz and/or 20 kHz, are added, to give even more meaning to the picture. But, if more-and-more frequencies are added, the picture becomes less-and-less clear. The data lines begin to cluster too close together, and it becomes difficult to distinguish one frequency line from another. Thus, the two-dimensional graph, as useful as it is, becomes limited in its full expression. Or, as Confucius

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Almon H. Clegg is the manager of the Audio Engineering Department, Product Engineering Department at Technics, and a personal friend of Confucius.

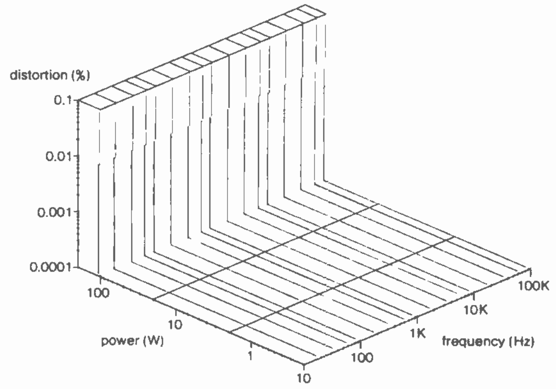


Figure 1. The 3DA characteristics of an ideal amplifier.

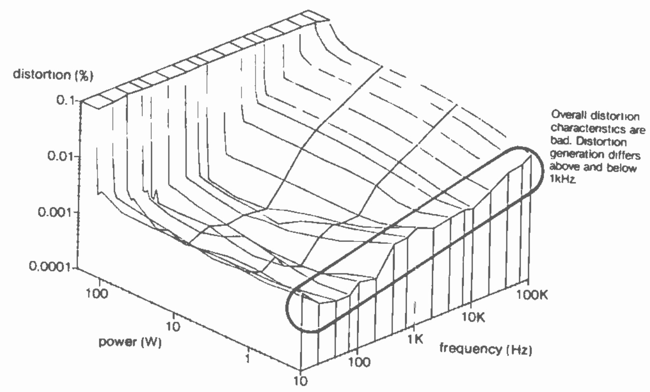


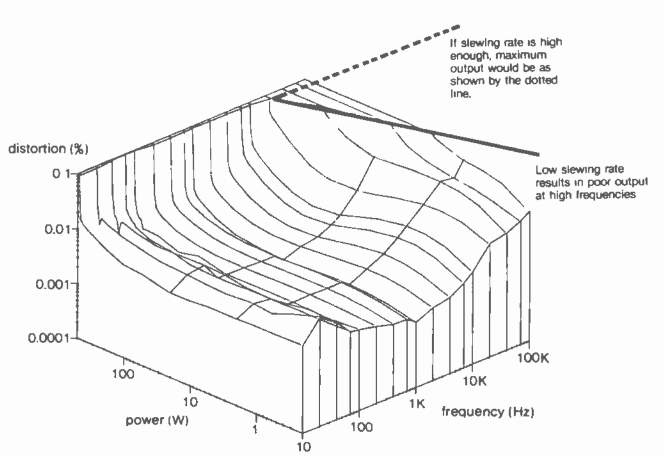
Figure 3. An amplifier with a high overall distortion level.

might have added as an afterthought, "More than one thousand data points probably requires more than one picture."

Another graphical representation which is not-so-common (mainly due to the difficulty in taking the measurements), is a plot of thd-*vs.*-frequency at some fixed power level. In any case, the number of fixed power curves is again limited to a few, since—as before—too many lines make the picture too confusing.

Nevertheless, graphical presentations of data are very meaningful, and provide a synergistic way of looking at the distortion of an audio system. A table of distortion measurements, taken at 1 kHz, as power is increased from a very low level—say, 0.01 watt to full-rated power (or more)—takes on added meaning when the data is plotted and presented as a graph of distortion-*vs.*-power level. If the distortion increases at lower levels (compared with the mid-power points), it indicates a poor signal-to-noise ratio. This assumes the distortion analyzer used for the measurements reads total harmonic distortion and noise (most do). (For the formula for thd-plus-noise, see Larry Maguire's article in this issue of **db**—Ed.) It may also indicate high crossover distortion in the output stages (due to poor biasing), or both. How the distortion behaves as full power is approached can suggest certain things about the drive circuit to the output devices, and, it can say something about the amount of feedback employed.

Figure 2. 3DA characteristics of a low-slewing-rate amplifier.



Similarly, the plot of distortion-*vs.*-frequency, at a given power level, can portray trends and conditions of the system design. For instance, rising distortion at lower frequencies may be a symptom of an inadequate power supply, a coupling capacitor which is too small, or too much ripple in one of the voltage supplies. On the other hand, rising distortion at higher frequencies can indicate inadequate output devices, incorrect phasing of feedback, presence of transient intermodulation distortion (TIM) and other slewing-induced distortions (SID), as well as other conditions.

The point to be made, however, is that a graphical presentation can be worth more than a thousand words; or, as Clegg (not Confucius) would say, "A graph is worth more than a thousand data points!"

This being the case, it would seem that adding more points, and thus giving more detail and purpose to the picture, would reveal even more about the system under test. And so it does! Taking their cue from our friend Confucius, Japanese engineers have put another thousand words into the picture. But in order to keep the picture clear, another dimension has also been added.

THREE-DIMENSIONAL ANALYSIS

The engineers at the Technics Stereo Department of Matsushita Electric Industries have developed a graphical presentation technique, which combines the information of the two afore-mentioned graphs into one three-dimensional graph. They have named it 3DA, which stands for Three-Dimensional Analysis. The method plots frequency along the traditional X axis, distortion up the Y axis, and power back along the Z axis. Thus, distortion-*vs.*-frequency-*vs.*-power are all displayed in one "picture," and as many as 4,000 pieces of data can be meaningfully shown.

In order to take such a large sampling of data and hand-plot it, point-by-point, would be an impractical task, so the entire process has been computerized. A minicomputer manipulates the distortion-measuring equipment, stores the

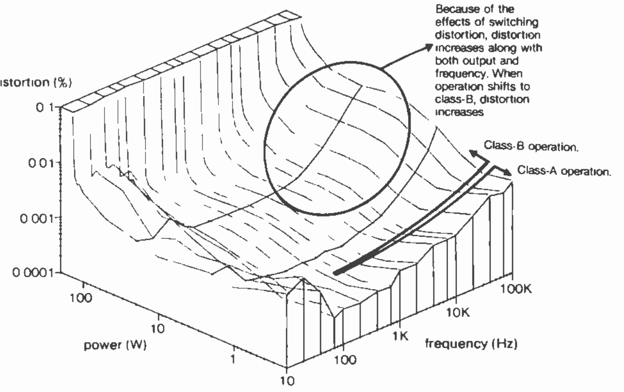


Figure 4. An amplifier with a high output idling current.

data collected in memory, and finally plots it out automatically on an XY plotter.

Specifically, 4,000 data points are taken with digital techniques employed to analyze distortion, up to the tenth harmonic. Distortion measurements are taken out to 100 kHz—much higher in frequency than is usually found in conventional measurements.

By using various "flags" in the computer program, a distortion threshold point can be set, at which the XY plotter is commanded to change to say, red-colored ink for traces above the threshold value. With the threshold value set at rated distortion, one can merely examine the contour of the lines at a glance to see if distortion is exceeded at any power level or frequency within its rated area.

As was suggested above, there is something synergistic about being able to make an expanded pictorial representation. Since the 3DA system allows presentation of 10 to 100 times the data of a two-dimensional plot, it should be able to present a comparably-larger amount of information on the graph. And a careful look will show that such is indeed the case.

BANDWIDTH, RISE TIME AND SLEW RATE

Consider for a moment three basic parameters of amplifiers: bandwidth, rise time, and slew rate. Bandwidth is a function of; the high-frequency capabilities of a device, how they are employed in the circuit, and, how much overall negative feedback is applied. Rise time, which we will consider a small-signal quality, (that is, the speed of the amplifier at low power levels before clipping, high-frequency power limitation, or other non-linearities set in), is directly related to bandwidth. Wider bandwidths mean faster rise times. It can be shown by some straightforward analysis that rise time is approximately 0.3 to 0.5, divided by the high-frequency cut-off. The exact value depends upon the circuits employed in the amplifier. Typical rise times are 3 to 5 microseconds for an amplifier with a 100 kHz bandwidth and 0.3 to 0.5 microseconds for a 1 MHz bandwidth. Thus, fast rise times merely indicate frequency response, and say little about the performance of the amplifier. On the other hand, slew rate is a measurement of the amplifier's speed (in volts-per-microsecond) in the non-linear mode, or under large-signal conditions; That is, when the feedback system has been disabled because of saturation or cut-off of some internal part of the circuit.

Power transistors are inherently nonlinear devices. As both power level and frequency increase, the current gain is reduced, causing a larger drive current from the previous stages. Depending on the design of the circuit, there becomes a point at high frequencies where the circuit saturates before the output devices swing to the maximum limits of the d.c. power supply. When this occurs, the amplifier is stripped of negative feedback and the "speed" is a function of the raw circuit and the individual devices. The slew rate is a measurement of this characteristic.

A CLOSER LOOK AT SLEW RATE

The question is often asked: Does a higher slew rate mean a better amplifier? In the world of commercial advertising and misinformed salesmen, the cry is "yes, yes." However, as the requirement for high slew rates is considered, one must reserve his feelings. Basic to high slew rates are circuits and devices that have good high-frequency linearity and drive characteristics. However, to achieve this, the geometry and real estate of the power output devices become sensitive to accidental short-circuit currents, because the safe operating area is reduced. This in turn requires fast-acting short-circuit protection systems, which are capable of introducing performance limitations. Another danger is that of radio interference brought about by such wide-bandwidth capabilities.

Thus, the truth of the matter about slew rate is that, so long as it is adequate for the power rating of the amplifier and the frequency content of the musical signal, increasing to higher levels introduces undesirable characteristics which could be dangerous to the amplifier.

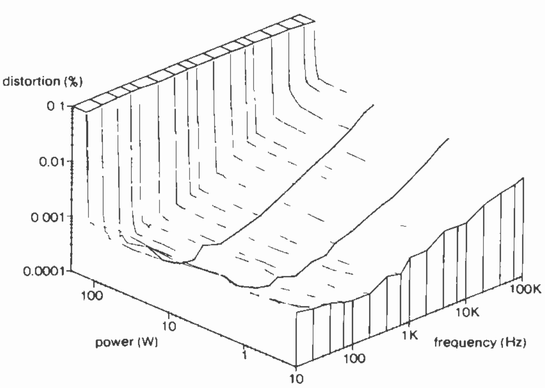
In conclusion, bandwidth and rise time are inter-related descriptive terms which characterize low-level linear performance, while slew rate is equivalent to frequency response at high power, non-linear levels. Slew, therefore, can be seen under steady-state conditions, and is not necessarily observed with transient signals only.

Transient intermodulation distortion (TIM) is evident when an amplifier's non-linear, or slew-rate limit, characteristics occur while attempting to reproduce a signal. The presence of TIM can be determined by observing the high-power, high-frequency, steady-state characteristics.

HAS ANYONE SEEN TIM OR SID?

Now back to 3DA. Since the 3DA system shows distortion out to 100 kHz, and up to full power, it is a simple

Figure 5. The parameters of an amplifier designed in conjunction with 3DA measurement techniques.



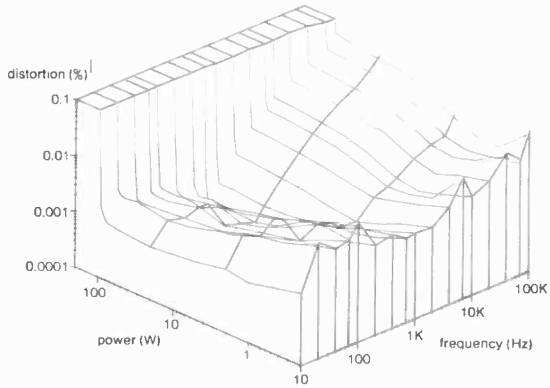


Figure 6. The 3DA characteristics of a well known 400 watt amplifier.

matter to see if TIM is present. Many other qualities and characteristics can be surmised by merely looking at the overall picture. To illustrate this, refer to FIGURES 1 thru 5. FIGURE 2 shows an amplifier which has slewing-induced distortions (SID); that is, high distortion at high frequencies and nearing full power. FIGURE 3 shows an amplifier which is bad overall. It probably has inadequate drive circuits and insufficient feedback to lower distortion. FIGURE 4 shows an amplifier with a high crossover non-linearities, which result in generally-increasing distortion as power output is increased and the drive circuits cannot supply enough current at high frequencies. This amplifier, however, might be a very "safe" design, insofar as failures caused by accidental short circuits and low-impedance loads.

Utilizing 3DA in the design process gives the engineer a way of quickly determining any design problems involving the above-mentioned problems. FIGURE 5 plots the parameters of an amplifier that was designed utilizing 3DA as a tool to perfect its distortion/power characteristics. Rated power is 120 W into 8 ohms, at less than 0.007 per cent distortion. The threshold setting for the XY plotter is 0.01 per cent. FIGURE 6 shows a well-known and highly respected amplifier which is rated at 400 watts into 8 ohms at 0.05 per cent distortion. Here, the threshold level is also set at 0.01 per cent.

To paraphrase a famous person (we're not sure whether the author is referring to Confucious or Clegg—Ed.) "4000 data points, up to the 10th harmonic, makes a mighty-fine picture."

In the search for state-of-the-art techniques, 3DA offers another step forward in objective characterization of amplifier performance. But one may ask whether this method results in a subjectively-improved sound quality. If the editor can tolerate it, a future discussion will be presented on a new distortion method called I/O analysis. This new method not only confirms 3DA techniques but offers new objective as well as subjective information about amplifier performance. ■

LARRY MAGUIRE

Distortion and How It Is Measured

Today's high-technology recording and broadcast hardware demands high-performance test equipment to accurately measure distortion.

THE DRAMATIC IMPROVEMENTS that we have recently seen in test equipment are making it possible to achieve even better-sounding recordings in the modern studio. For example, the latest generation of distortion-measurement devices permit us to measure and localize distortion-producing system components with a degree of precision that was, until quite recently, beyond our capabilities.

CLEANING UP THE SIGNAL PATH

Are you sure your audio signal chain is *really* clean? Or, is there possibly a distortion-producer hidden somewhere between input and output that could stand a little clean-up, or perhaps replacement? Instead of guessing, a few routine measurements will help you verify the specifications of your system.

This article will focus on total harmonic distortion and intermodulation distortion, describe some techniques for measuring them, and compare the results of the two types of distortion measurement. We'll also point out some of the pitfalls of *not* taking advantage of the latest technological improvements in distortion-measurement devices.

Larry Maguire is vice president of Sound Technology, Campbell, California.

TOTAL HARMONIC DISTORTION

Total harmonic distortion (thd) is defined as the ratio of the rms voltage for all *harmonics* present in the output of an audio system, to the total rms voltage at the output. Using a pure sine wave input, the ratio is expressed in per cent.

Let's consider an output waveform consisting of the following components;

V_A = the rms amplitude of the fundamental frequency.
V_B = the rms amplitude of the harmonic content.
V_N = the rms amplitude of the noise voltage.
The formula for total harmonic distortion is;

$$\text{thd (plus noise)} = \left(\frac{V_B + V_N}{V_A + V_B + V_N}\right) 100\%$$

To determine the thd, the measuring instrument must remove the fundamental frequency, measure what is left (V_B and V_N), and express this as a percentage of the total signal. FIGURE 1 shows the test set-up for measuring thd. An audio signal generator provides the test signal, which is fed to the unit under test. The degree to which the test signal is pure will limit the accuracy of the measurement. Ideally, the test signal should have at least five-times-lower distortion than what we are trying to measure. For example, if the thd of the test signal is 0.001 per cent, an accurate measurement of distortion down to 0.005 per cent can be obtained. As we shall see later, the degree to which the measuring instrument does not further distort the output of the unit under test will also limit the accuracy of the measurement.

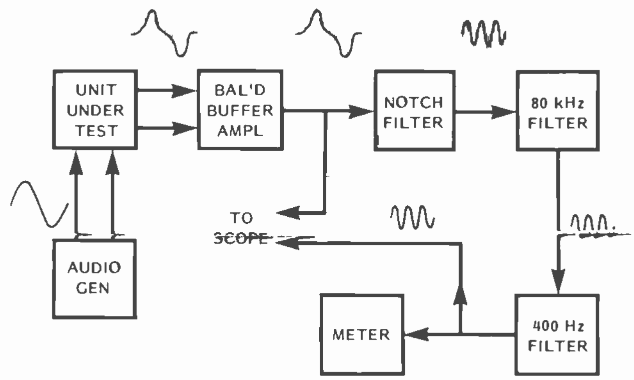


Figure 1. Total harmonic distortion measurement.

The unit under test adds distortion and noise to the test signal, as shown in FIGURE 1. All the blocks following the unit under test represent components of the thd analyzer. Several functions are performed by the first block, the buffer amplifier. First, it scales the output of the unit under test so the following measurement circuits have to cope with only a narrow range of signals. If the buffer amplifier is differential or balanced to ground, several advantages are provided. The most obvious benefit of a balanced input is the ability to measure a floating source. Many audio amplifiers are strapped or bridged and can only be measured by a balanced instrument. The second benefit is the abatement of one of the classic problems in distortion measurements; ground loops.

COMMON-MODE VOLTAGE REJECTION

FIGURE 2 shows a balanced measuring instrument can eliminate errors caused by this troublesome source. The output of the balanced buffer amplifier, V_0, is proportional to the difference between the high and low sides, V_1, & V_2, of the output of the unit under test. However, let's assume there is a ground loop. This is seen in FIGURE 2 as the common-mode voltage, V_{cm}. The balanced buffer amplifier's inputs are now $V_1 + V_{cm}$ and $V_2 + V_{cm}$ with respect to ground. The differential amplifier will reject V_{cm}, because it appears equally at both input terminals, and only differences are amplified. This is referred to as the common-mode rejection of the amplifier, and it will reduce the hum picked up in ground loops by a factor of at least 100. On the other hand, if the buffer amplifier were single-ended, V_{cm} would appear amplified in the output, causing a direct measurement error, and countless unhappy hours spent ground-loop chasing.

NOTCH FILTER REQUIREMENTS

Returning to FIGURE 1, the buffer amplifier is followed by a notch filter. This filter is extremely complex, because it must attenuate the test signal, or fundamental, to a high degree but cannot attenuate the harmonics at all. For example, to measure thd of a 1 kHz signal down to 0.001 per cent, the 1 kHz fundamental must be attenuated at least 100 dB. However, if the second harmonic (2 kHz) is attenuated at all, a measurement error is introduced, and distortion readings will appear to be artificially low. The result of these requirements is a filter with an extremely-narrow rejection band and very steep skirts, which in turn causes another measurement problem.

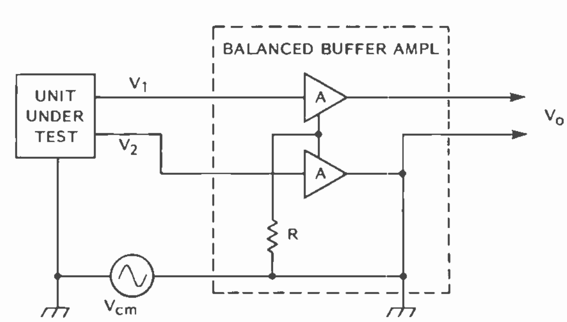
If the notch filter is not precisely tuned to the test frequency, so that the fundamental is completely attenuated or nulled out, a very large measurement error will result. But because the notch filter must therefore have such sharp skirts, even a slight mistuning will cause a large portion of the fundamental to pass through the filter, and be added to the distortion reading. Measurements with older distortion analyzers were very difficult, and slow-to-use below about 1 per cent distortion, because the null adjustments had to be done manually. To try to measure less than 0.01 per cent distortion, as is required by today's audio systems, would be totally impractical with a manual-nulling analyzer. Modern analyzers have electronic automatic tuning of the nulling circuits, which keep the notch filter tuned to the exact frequency of the test signal. These circuits have about a ± 3 per cent capture range, and completely eliminate manual nulling, which can take several frustrating minutes. A distortion reading can be obtained in 5 seconds with an automatic nulling instrument, and the readings will maintain their accuracy even if the frequency of the test signal drifts.

While we are on the subject of measurement speed, it is worth mentioning that organization of the system can reduce measurement time significantly. If the audio generator and notch filter are in the same package, their frequencies can be selected with the same set of switches. When a measurement is to be made, one setting selects both the generator and filter frequencies and—while signal levels are being set to the desired point—the automatic nulling circuits are doing their job. The result is that the nulling time appears to be zero, and the distortion measurement system is as fast as the operator.

At the output of the notch filter, the signal consists of the distortion components and noise, as also shown in FIGURE 1. These are routed through a low-pass filter to reduce the noise component. The filter cut-off frequency is 80 kHz, so the system can be used when the test frequency is as high as 20 kHz, without attenuating the second or third harmonics; 40 kHz and 60 kHz. If the distortion components are small, the noise will have a masking effect and the 80 kHz filter will reduce this effect, facilitating analysis of the distortion components. As a side benefit, the filter will also attenuate any signals that may have been picked up from a.m. radio stations—a problem in distortion measurements that is nearly as troublesome as ground loops.

The 80 kHz filter is followed by a 400 Hz high-pass filter. This filter is intended to reduce hum introduced by the unit under test, rather than by ground loops. The differential amplifier cannot reject this hum because, it is a legitimate part of the output signal of the unit under test; a normal-mode rather than common-mode signal. The 400 Hz cut-off frequency is chosen to be as low as possible while attenuating the hum by a reasonable amount, so the filter can be used for as many measurements as possible. (The filter in Sound Technology systems is a 3-pole Butterworth, and attenuates 60 Hz by 40 dB.)

Figure 2. Common-mode voltage rejection.



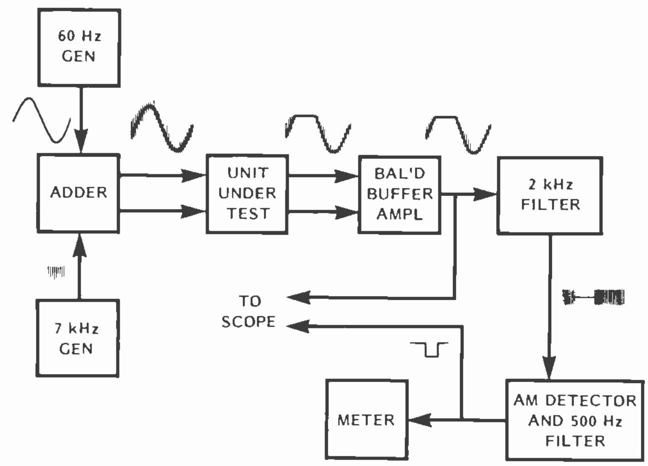


Figure 3. SMPTE intermodulation distortion measurement.

The output of the 400 Hz filter is converted to d.c. and fed to a meter to give a distortion reading. Although all meters are calibrated to read the rms value of an a.c. waveform, meter circuits can be designed to respond to the average, rms, or peak value of the waveform. Most measurement standards call for an average-responding meter circuit, although an rms-responding circuit is occasionally specified. A peak-responding circuit is valuable for measuring distortion products with high-order harmonic content —for example crossover distortion—to which the average and rms circuits are not very sensitive.

FIGURE 1 shows two outputs from the thd analyzer that can be used for oscilloscope viewing. One is the output of the buffer amplifier, which is a scaled reproduction of the output of the unit under test, and the other is the distortion products and noise present in the signal.

Viewing both signals simultaneously facilitates analysis of the distortion, and simplifies the troubleshooting process.

INTERMODULATION DISTORTION

Years ago, the Society of Motion Picture and Television Engineers (SMPTE) established what has become the most-widely-accepted method of intermodulation distortion measurement. According to the SMPTE standard, the unit under test is to be driven simultaneously with 60 Hz and 7 kHz sine waves. The 60 Hz amplitude is to be four times that of the 7 kHz. If the unit under test is perfectly linear, only 60 Hz and 7 kHz would be present at its output. However, if the unit distorts, the 60 Hz signal will modulate the 7 kHz. The SMPTE test is a measure of these intermodulation products. If their amplitude is V_2, and the amplitude of the 7 kHz signal is V_1, the SMPTE intermodulation distortion is defined by the following equation:

$$IM = \left(\frac{V_2}{V_1 + V_2}\right) 100\%$$

This equation tells us that the measuring instrument must determine the amplitude of the intermodulation products in relation to the 7 kHz signal, and express this ratio as a percentage. It is worth noting that the DIN standard, which is widely used in Europe, is the same as the SMPTE standard, except that the frequencies are 250 Hz and 8 kHz.

FIGURE 3 shows the test set-up for measuring IM distortion. A 60 Hz generator and a 7 kHz generator are added together to provide the test signal, which is routed to the unit under test. As in the case of thd, the output of the unit under test is connected to the buffer amplifier, which performs the same function described earlier. The output of the buffer amplifier is fed to a 2 kHz high-pass filter, which removes the 60 Hz while retaining the 7 kHz and intermodulation products. This filter must be fairly sophisticated, to provide 100 dB of rejection of 60 Hz and its harmonics, while passing 7 kHz and its sidebands. (A 7-pole Butterworth filter is used in Sound Technology IM analyzers.)

The intermodulation distortion products are present in the signal as amplitude-modulation sidebands of the 7 kHz signal. Therefore an a.m. detector and 500 Hz low-pass filter are used to isolate the modulation envelope. The modulation products are then routed to the meter circuit, where they are converted to d.c. and displayed.

The waveforms in FIGURE 3 are intended to show the two test frequencies, how they are added, and the signals that would be present in the measuring instrument if the unit under test is clipping on one peak of the 60 Hz signal. As in the case of thd, outputs are available for oscilloscope viewing.

Definition of signal levels when the complex intermodulation test signal is used is not straightforward. FIGURE 4 shows how the SMPTE specifies that it be done. The sine wave to the right has the same peak-to-peak amplitude as the intermodulation waveform on the left, which is comprised of 60 Hz and 7 kHz in a four-to-one ratio. The voltage level of the intermodulation waveform is defined by the SMPTE as being equal to the voltage level of the sine wave. In other words, when an instrument measures the level of an SMPTE signal, it measures the peak equivalent voltage, which is the rms voltage that a sine wave with the same peak-to-peak amplitude would measure. The true rms voltage of the intermodulation waveform is actually 82 per cent of the rms voltage of the sine wave. The definition may seem arbitrary, but it does provide a method of comparing signal levels of measurements made at different times using the complex SMPTE intermodulation waveform, and is therefore of value.

TOTAL HARMONIC VERSUS INTERMODULATION DISTORTION

It is logical to ask why we measure different kinds of distortion. On the surface you might think that either total harmonic *or* intermodulation distortion would tell the story. In practice, however, each one can fill-in, where the other has a weakness. The first thing to remember is; any method of measuring distortion is nothing more than a method of measuring the non-linearity of a device. If a device has no non-linearities, there will be no distortion. The differences in measurement methods lie in their sensitivity to various types of non-linearity. For the same non-linearity, SMPTE intermodulation distortion readings will be higher than thd measurements that are made at mid-band audio frequencies. For example, if the distortion components are entirely second-order, the SMPTE method yields a reading 3.2 times higher than thd. If the distortion components are entirely third order, the SMPTE method yields a reading 3.8 times higher. In these cases, therefore, the SMPTE method is considerably more sensitive to the non-linearity. In addition, the SMPTE measurement is less-influenced by noise, because it is a narrow-band measurement, whereas thd is a wide-band measurement.

On the other hand, thd is a more-versatile measurement technique. It can be better used to observe a device under test at both very-low and very-high frequencies. For example, measurements of thd at high audio frequencies are used to evaluate the effect of slewing on amplifier distortion. An amplifier that is driven to a given peak voltage level with a sine wave is delivering 47 per cent more power than if it is driven to the same level with the SMPTE

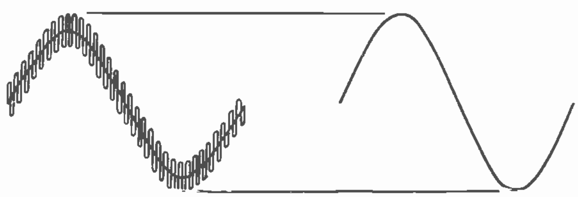


Figure 4. Peak equivalent voltage.

signal, and is therefore stressed harder. In other words, thd measurements provide a more-rigorous test of an amplifier's ability to accurately reproduce high-power signals. About the best that can be said therefore, is that each method has its advantages, and both are needed for a complete evaluation.

WHY ACCURACY IS IMPORTANT

It is generally true that test equipment specifications must be better than the specifications of the system under test, and distortion measurement equipment is no exception. For the same reason you wouldn't try to adjust a digital watch using a sundial as a standard, you shouldn't try to repair a precision audio component with imprecise test equipment. Any test equipment errors will add to the uncertainty of the measurement you are making. For example, if you measure a 100-volt signal with a meter that has ± 2 per cent accuracy, you know the voltage is somewhere between 98 and 102 volts, but that is as close as you can measure.

The same is true when measuring distortion. If your audio signal generator has a thd of 0.001 per cent, this limits the accuracy of your measurement. The residual distortion of the analyzer itself also adds to the measurement error. Let us assume for a minute that you are attempting to analyze an audio component using a signal generator that has 0.1 per cent distortion, and an analyzer with 0.1 per cent residual distortion. Let us further assume you do an outstanding job and finish up with a reading of 0.0 per cent distortion. What you have actually done is measured an audio component with 0.2 per cent distortion. But you have been mis-led by distortion cancellation. The 0.2 per cent total distortion of the test equipment has been cancelled by the 0.2 per cent distortion of the audio component, which was in opposite phase. The net result is that, in spite of your best efforts, the test equipment has let you down and prevented you from getting the job done.

Another justification for selecting test equipment with superior specifications is protection against obsolescence. Not too many years ago an audio component with 0.5 per cent distortion was considered exceptional. But today, high-power amplifiers with 0.05 per cent distortion are fairly common. The state-of-the-art will continue to improve, so buying good test equipment is like buying insurance against obsolescence.

SUMMARY

We have attempted to define both total harmonic distortion and SMPTE intermodulation distortion and to delineate some of the design parameters that are used in equipment that measures these important effects. In addition, we have discussed the significance of test equipment accuracy specifications and the effect they can have on measurements. We have tried to show that both accuracy and operating speed are important test equipment parameters. ■

ROBERT K. MORRISON

Magnetic Tape Reproducer Calibrator

A new calibration network brings added versatility to induction loop measurements.

If one were to make a recording of a flat-frequency sweep through an unequalized amplifier connected to a magnetic tape head, (a constant-current recording), reproduction of that recording through an un-equalized playback amplifier would produce an output signal rising in level as the frequency increased. Within those ranges of tape speed and frequency where playback head dimensions do not complicate matters, (such as 15 in/sec at audio frequencies), this rise would be 6 dB-per-octave. However, a reproducing amplifier is not made with a complementary response to compensate exactly for this effect (that is, with a drop of output level of 6 dB-per-octave). Since it has been the practice to pre-emphasize some frequencies in the recording process (in the interest of noise suppression), the reproducer response must have further modifications, such that the overall system response is flat. Once the decision has been made—based upon the tape speed and oxide characteristics—which ranges of pre-emphasis and corresponding reproduce equalization will provide the best compromise between signal-to-noise ratio and recording distortion, a single standard is established for the *reproducer* equalization of all recorders utilizing that recording format.

Robert K. Morrison is founder of Standard Tape Laboratory.

With so many different tape oxides on the market, one can still produce tapes with compatible playback equalization, on a number of different machines, all set to the same playback characteristic. However, the record equalization of each machine must be set according to the tape being used. (Other factors including the record head gap and bias adjustment contribute to the recorded results.) In each case, the overall system response is controlled by means of varying the record equalization as needed (once the playback characteristic is properly adjusted).

In 1953 the National Association of Broadcasters (NAB) specified a 15 in/sec tape reproduction curve in terms of an RC circuit. The figures defined the RC circuit in terms of time constants: 50 microseconds for the high end and 3180 microseconds for the low end. It was described as a curve to be superimposed upon a 6 dB-per-octave slope. The time-constant method was previously used for other broadcast standards, such as f.m. broadcast pre-emphasis, etc. Since the term is an expression of the product of a resistance and capacitance, it is a concise method of description. It is common practice to describe a tape reproducer curve, not only by the time constants involved, but by the transition frequency, which is the point at which the curve departs by 3 dB from a straight line. The easy formula to determine the transition frequency, when the time constant is known, is:

$$\text{transition frequency} = \frac{1}{2\,\pi\,\mathrm{RC}}$$

Thus, the transition frequency for a time constant of 3180 microseconds is 50 Hz, which is that point where a 3 dB deviation from flat occurs.

INDUCTION LOOP MEASUREMENTS

For the past twenty years, many of us in the audio test tape, head manufacturing, and equipment design areas have used the "hot wire" or induction loop method of calibrating a magnetic reproducer channel. (For more information on this, see the author's article, "Using Induction Loops," in the January, 1979 issue of **db**. The feature is an excerpt from Morrison's **Standard Tape Manual**—Ed.) The method has been particularly useful in determining high frequency equalization. However, low frequency characteristics of reproduce heads are complicated by core-design effects which cause the so-called "head bumps" which are not seen in induction loop measurements.

These effects become more severe as the wavelength of the signal increases. In the case of high-speed recording of the normal audio bandwidth (30 in/sec mastering for example), the design compromises of the playback head are such that a great disparity usually exists between the "ideal" low-frequency characteristic, and the actual response needed for a given head to produce a flat response. Most of the manufacturing practices for equipment have included modifying the low-end playback characteristic to provide flat reproduction of constant-current recordings of low-frequency test signals, rather than on a basis of "ideal" electrical response. In checking a system to a standard calling for a given de-emphasis at the low end (3180 microseconds for example), the playback channel would be adjusted to reproduce a constant current test tape of low frequency tones with a drop in response of 3 dB at 50 Hz.

Remember, the standards themselves specify a fixed response, to be used in terms of an "ideal" head, or describe the fluxivity of a standard tape which must be complemented by a playback channel, i.e. electronics-plus-head. An ideal head would be a head with no losses, and of course such a head doesn't exist. In all fairness, however, some of the present-day head manufacturers are producing magnetic playback heads that are so close to ideal, at mastering tape speeds, that the difference between ideal high-frequency electrical response and actual playback response from tape may be ignored. What we are saying here is that a reasonably-priced playback head may very likely fall within 0.6 dB of "ideal," as to high-frequency response, when set up and measured by means of induction loop response.

Very accurate calibration of a given system can be accomplished when the electrical response *and* the physical characteristics of the head are known. That is, if we know or determine the gap loss, spacing and contour deviations of a playback head, and then measure the electrical characteristic which will include resonance and core loss, we have an accurate, and repeatable measurement of the characteristics of the system. (These parameters are discussed in detail in the author's **Standard Tape Manual**—Ed.)

METHODS

As noted in the January issue of **db**, the hot wire or induction loop method of measurement simply requires a piece of wire located parallel to the gap of a playback head. The wire is then driven by an oscillator to provide constant current across the bandwidth of interest. The common way to do this has been to meter the voltage across a resistor in series with the loop.

Obviously, accurate instruments must be used, and the output of the playback machine should be connected to an electronic voltmeter for system calibration. With enough signal in the loop to produce a convenient reference signal at 400 Hz on the minus 20 dB scale of the voltmeter at the output of the playback amplifier, playback response is measured.

At this point, one must know the proper loop response curve for the required standard at hand. One can either calculate the curves, look them up, or make use of an accurate de-emphasis network designed for the purpose.

In the test tape business, we have used such networks for years, with different "black boxes" for each curve. (With the present array of curves, we have recently produced ONE device to accommodate the gamut.) While on the subject, this might be a good time to list the curves most frequently used at this date for magnetic reproducers.

TIME CONSTANTS AND TRANSITION FREQUENCIES

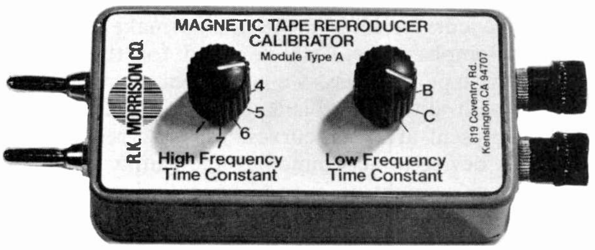
RC = Time constant (in microseconds)
F_T = Transition frequency

	Low Frequency		High Frequency	
MAGNETIC TAPE	RC	F_T	RC	F_T
1⅞ in/sec cassette	3180	50 Hz	70	2274 Hz
1⅞ in/sec cassette	3180	50 Hz	120	1326 Hz
1⅞ in/sec reel	3180	50 Hz	90	1768 Hz
3¾ in/sec reel	3180	50 Hz	90	1768 Hz
7½ in/sec cartridge	∞	—	50	3183 Hz
7½ in/sec reel (USA)	3180	50 Hz	50	3183 Hz
7½ in/sec reel (European)	∞	—	70	2274 Hz
15 in/sec reel (USA)	3180	50 Hz	50	3183 Hz
15 in/sec reel (European)	∞	—	35	4547 Hz
30 in/sec reel (USA)	∞	—	17.5	9095 Hz
30 in/sec reel (European)	∞	—	35	4547 Hz
FILM				
Super 8mm	3180	50 Hz	90	1768 Hz
16mm	∞	—	70	2274 Hz
35mm	∞	—	35	4547 Hz
VIDEO				
Two inch quad audio	2000	80 Hz	35	4547 Hz
One inch type "C"	3180	50 Hz	15	10610 Hz
¾-inch cassette	3180	50 Hz	50	3183 Hz

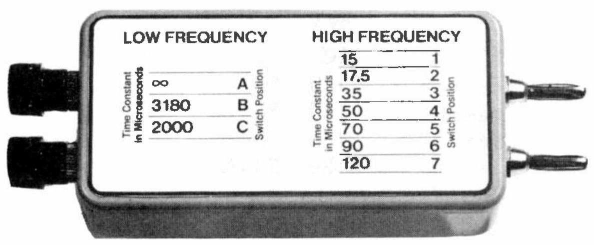
From the above list of popular magnetic curves, one can see that at the present time there are three low-end characteristics and seven high-end time constants which may be used in various combinations when setting up commercial audio reproducers. For convenience, we have produced a precision calibrator to enable an operator to quickly accomplish such measurements.

The calibrator consists of a de-emphasis network, together with needed head fixtures and cable set to provide a compensated flat output from a magnetic reproducer when set to the correct time constants in use. Such measurements can be made without calculation or reference to charts and we believe that it is usually more accurate and always more convenient to keep all readings at a single reference point near the top of one scale, thus eliminating any scale-to-scale tracking calibration errors.

In the case of head development or accurate quality-control procedures, one has the added advantage of being able to employ a chart recorder along with the calibration device, to produce an expanded scale on the chart paper,



(A) Top view



(B) Bottom view

A new magnetic tape reproducer calibrator.

indicating very small variations. This may be done without using up a large vertical segment of the chart paper merely to accommodate the normal 6 dB-per-octave rise in reproducer response when making induction measurements without a de-emphasis network.

PLAYBACK HEAD SELECTION

A number of playback heads may be connected to the same amplifier and measured with an induction loop and a de-emphasis network. Variations at this stage will indicate resonance and core loss differences. The next step is to play a standard alignment tape (with the calibration box OUT of the circuit.) Head-to-head variations will now indicate the deviations from ideal due to gap loss, and contour effects of the head. Such a procedure is often employed by head manufacturers at the time of quality control. The same technique can, of course, be employed by the user, particularly when performing acceptance tests on a quantity of heads.

The methods of determining the amount of resonant effect and contour effect for a head include: inductive feed to the head with a gliding tone to determine the resonant point, and reproduction of constant-current, long-wavelength recordings.

TEST TAPES

In a test set-up to produce or verify the performance of a test tape, the loop and calibrator provide a repeatable and unvarying means to reference old and new test tapes to a specific reproducer channel. Test tapes can and do change, particularly at the higher-frequency (short-wavelength) tones. The calibrator provides a convenient means of determining the accuracy of a new test tape, as well as noting any change with wear, de-magnetization, aging, etc. However, it must be recognized that a given playback head will also not remain unchanged physically as to spacing loss and gap condition. Also, wear patterns change with use, altering response as tape-to-gap pressure varies.

The normal routine employed by those interested in making or checking test tapes is to retain a specific "testing" playback head which suffers very little wear, as it is only used for calibration purposes and not for routine production use. This head is then employed with a standardized reproduce electronics with the loop and de-emphasis network.

The gap losses for the head in use have been determined by test or computation and the tape being produced or checked is referenced to the ideal loop response, less the gap loss. (Gap losses with short-gap heads employed at 7.5 in/sec speeds and up can be on the order of a fraction of a dB).

USE WITHOUT TEST TAPES

Where it is necessary to determine or adjust electronics equalization, and when the playback head's effective gap length is known, along with the low-frequency contour characteristic, the electronics can be set to standard with the loop and de-emphasis network without the use of a test tape. In such a case, with fixed-base heads having no azimuth adjustment, the remaining factor to determine would be the operating or reference level. This could be determined with reference to the saturation level of a specific type of raw tape stock. Saturation level and distortion readings can give a repeatable measurement *if* careful vault-reference samples are maintained.

GENERAL COMMENTS

The induction loop contains a resistor with a cable connection which must be metered for constant voltage drop during the measurement. Many first-rate audio oscillators will provide flat response at all required frequencies without adjustment; however a metering facility permits verification. The calibrator is made to connect directly to the electronic voltmeter, and this method must be followed, since additional cable between them would add capacitance to the circuit, thus causing a faulty measurement.

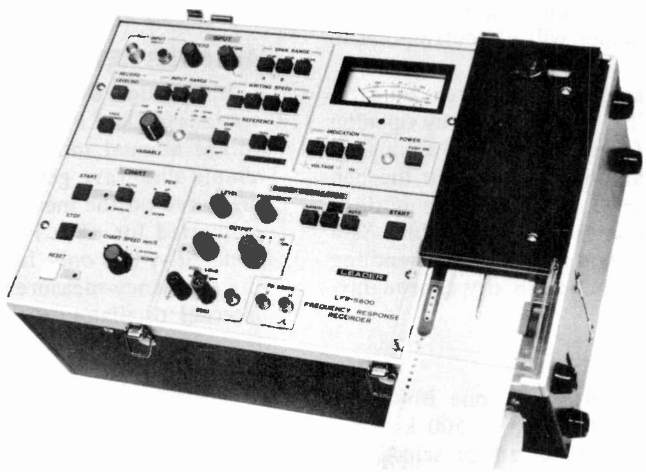
We should stress what the technique and apparatus will *not* do:

It will not provide azimuth reference.

It will also not accurately provide a flux-level reference, since the sensitivity of the set-up is dependent upon the actual positioning of the loop in front of the reproduce head. Practice shows that this may be repeated with an accuracy of about 5 or 10 per cent—that is, 0.5 to 1 dB. Even this tolerance can sometimes be used for certain rough measurements. We should state here that the frequency response of the set-up is *not* dependent on the proximity of wire-to-head.

The methods herein described are nothing new, having been used by standard labs, test tape makers, and head designers for years. They are now becoming more frequently employed by recording studios, broadcasters, and sophisticated users of recording equipment. As a result, those so-involved are one step closer to a primary method of determining where to begin in standardizing a magnetic tape reproducer. ■

db Test Report



Leader LFR-5600 Frequency Response Recorder

LEADER'S LFR-5600 is a compact multi-purpose test system, consisting of a sweep oscillator, strip-chart recorder, and an input/meter section.

SWEEP OSCILLATOR

A set of three pushbuttons selects manual, pilot signal, or automatic sweep mode. In the manual mode, a rotary potentiometer varies the output frequency between 20 Hz and approximately 52.5 kHz. A second potentiometer varies the output level, up to a maximum of 3.5 volts, or 1.8 volts with the internal 600-ohm load switched in. With this load switched out, the oscillator's source impedance was measured at 1460 ohms. Two output connectors are provided: a double-banana plug, and an RCA-type phono plug. (The load switch affects both.)

Output level, or frequency, may be read on the built-in meter, which is calibrated in volts, dBV (0 dBV = 1 volt), and frequency. The logarithmic frequency scale is reasonably accurate, when checked against an external digital frequency read-out. There seemed to be a discrepancy which varied up to about 40 Hz over the range of the meter (20 Hz to 40 kHz). However, precise resolution is obviously restricted somewhat by the necessarily-few frequency markers on the meter face.

In the pilot-signal mode, the oscillator output is switchable between 333 Hz (for cassettes) and 1 kHz. The output level remains fixed at 2.5 volts (600-ohm load switched off).

In the automatic mode, sweep speed is controlled by a five-position chart speed switch located in the chart section of the system. The switch keeps the sweep speed in sync with the chart speed. Four chart speeds are available: 0.1, 0.3, 1 and 3 mm/second. The fifth position of the switch, labelled *scope,* provides a 10-second sweep, but the chart recorder is disabled in this mode.

One of four pen-writing speeds may be selected. These are given as: 0.1, 0.2, 0.5 and 1 second. The higher the speed, the more accurately the pen tracks amplitude variations. Presumably, the 0.1 second position would be selected for accurate tracking of say, the response of a notch filter. A slower speed would be preferred for smoothing out instantaneous variations in noise level measurements.

For most charting applications, the 3mm/second position will no doubt be preferred. At this rate, the sweep takes 54 seconds to trace a 16.2 cm-long frequency response (20 Hz to 30 kHz) curve on the strip chart. (At a chart speed of 0.3 mm/second, the sweep takes 540 seconds—9 minutes!—to run its course.)

THE CHART RECORDER

When hard-copy charts of sweep runs are required, the auto/manual switch is depressed (auto), as is the pen up/down switch (pen down). However, in the auto mode, the pen still remains up, until the sweep oscillator's start button is depressed. A five-second pilot signal is heard (at 333 Hz or 1 kHz, depending on which one was preselected), after which the pen automatically drops, and the sweep run begins.

Simultaneously with the pen-drop, the chart recorder starts, *providing* the sweep oscillator's output is connected to the system's input section—presumably through the device to be tested. If the device under test is, say, an equalizer, the chart recording begins immediately, as you would expect.

However, when testing tape recorders, it may not be necessary, nor desirable, to generate hard-copy charts immediately. You may wish to record the sweep run on several tracks at once, and then look at the traces later on. In any case, the strip chart can obviously record only one track at a time anyway. Therefore, during recording, the chart recorder may be disabled (manual mode, pen up).

Later on, to produce hard copy of any track, simply connect the appropriate playback output to the system's input. Next, depress the auto/manual switch (*auto*). Nothing happens until the pilot tone (now on the tape) is detected. At the end of the tone, the pen drops, the chart begins moving, and the sweep run on the tape is traced on the strip chart. Likewise, sweep runs on test (phonograph) records may be recorded, although if the sweep time does not correspond to one of the four chart speeds, the frequencies printed on the chart paper will not correspond to the plotted curve.

With some care, multiple traces may be written, by carefully "rewinding" the chart. This may be done by opening the access cover to the roll of chart paper, and taking up the slack, while turning the thumb wheel next to the pen in the reverse direction. However, since it takes quite a while for the ink to dry, some smearing is inevitable. As a matter of fact, it seems to be a good idea to avoid handling the charts until they've had a chance to dry thoroughly. This may take an hour or more.

INPUT SECTION

The LFR-5600 has two parallel inputs; one BNC and one RCA phono plug. Input impedance is 500 k-ohms. Three pushbutton switches select input range sensitivity, which is 10, 100 or 1000 millivolts/cm. The vertical span range may be set at 25 or 50 dB (that is, 5 or 10 dB-per-centimeter). As a fail-safe measure, the strip chart will not be activated if the input range selector is not set properly. This spares the user the aggravation of collecting a series of useless off-scale traces.

MEASURING D.C.

The system may also be used to record d.c., by switching the input section from frequency response to level (d.c.). This can be useful for keeping an eye on the long-term stability of power supplies, or whatever, with or without the chart recorder in use. (In any case, the position of the pen shows the d.c. voltage—the meter is deactivated.)

When using the d.c. mode, make *sure* there is no audio signal present at the input terminals. If there is, the pen mechanism will go beserk, as it tries to keep up with the d.c. level of the a.c.-audio. (It's a nice way to spray-paint the immediate area.)

As another precaution, make sure the unit is kept in a horizontal position while the power is on. This should be no problem, as the system is intended to work in this position, and vertical placement would impede the paper feed. However, if the unit is tilted as it is being positioned in the test area, the pen has a tendency to leak, and things can get a bit messy. It doesn't happen all the time, but once is more than enough. As you may have gathered from all this, the chart recorder uses an ink reservoir system to feed the pen. This is quite-easily filled, with no disassembly required.

AN OVERVIEW

In addition to its performance as a system, the various sections of the LFR-5600 may of course be used separately. In the manual mode, the sweep oscillator functions as a simple sine-wave generator, with output level or frequency measured on the meter. (Remember, the meter scale is *not* calibrated for dBm.) The meter itself may be used independently, but only for level measurements. At all times, the frequency-measurement section remains coupled to the internal oscillator, so it cannot be used to determine the frequency of external sources.

SPECSMANSHIP

The sweep oscillator was spot-checked for harmonic distortion, with the following results:

0.85 per cent @ 20 Hz
0.50 per cent @ 1 kHz
1.10 per cent @ 20 kHz

Overall sweep was accurate, within 0.25 dB. On the 50 dB span range, a cumulative error of about 0.5 dB-in-10 dB was noted, with a —50 dB input charted at about —48 dB.

The Leader LFR-5600 Frequency Response Recorder comes complete with a red ink supply, rolls of logarithmic and linear chart paper, and connecting cables. Weighing in at 21 pounds, the system is readily portable, and comes in its own built-in metal carrying case, which appears to be rugged and quite well-constructed. The suggested list price is $3,195.00.

POSTSCRIPT

Leader Test Instruments manufactures a rather extensive line of moderately-priced test equipment. Their LMV-89 dual-channel/dual-pointer a.c. voltmeter can be found on just about everybody's test bench. Watch out for this one though; it can become addictive. For a complete catalog, contact Leader. Directions are in our directory of test equipment manufacturers in this issue. L.Z. ■

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SIDNEY L. SILVER

Digitizing Audio with Delta Modulation

Not to be confused with PCM, delta modulation is now finding its way into pro audio applications.

About thirty years ago, investigators in the telecommunications field found that, instead of using a large number of digits to encode audio information, a sequence of logical "1" and "0" pulses could be arranged in such a way that this pulse stream could be generated with a single-bit code.[1] This technique, known as delta modulation, was subsequently applied to speech transmission in telephone communication systems.[2] But, owing to its limited dynamic-range capability, and a low signal-to-noise ratio, it was considered unsuitable for wide-band audio applications. Recently, however, these difficulties have been overcome by advances in integrated-circuit technology. Sophisticated refinements in circuit design now make it possible to apply delta modulation to the digitizaton of high-fidelity music signals.

PCM VERSUS DELTA MODULATION

Unlike pulse code modulation (PCM), which uses a multi-bit code[3] to provide data on the absolute magnitude of each sample of the audio signal, delta modulation generates a single digit at each sampling time, to reflect momentary changes in the input signal. It does this by sensing the *difference* in unit steps between two successive quantities of the audio signal rather than the signal value itself. A delta modulator is thus a differential encoder, and it is this signal variation—the **delta**—that gives the technique its name. In comparing pulse code modulation with delta modulation, the latter has the advantage of simpler instrumentation (and hence, lower cost of coding equipment). There are less-severe requirements for filtering, and, because of the one-bit code employed, the need for word synchronization is eliminated. In exchange for these benefits, delta modulation imposes certain restrictions on input signal parameters. These are related to overload problems in the presence of high-amplitude, high-frequency components. Also, a larger transmission bandwidth is required, in order to achieve equivalent signal-to-quantizing noise ratios.

BASIC OPERATION

In its simplest form (Figure 1), the encoder in a linear delta modulation system consists of a differential amplifier and a D-type flip-flop in the forward path, and an RC integrating network in the feedback path, of a closed-loop system. A difference signal is obtained by comparing the audio input signal with a reconstructed feedback signal derived from the digital output of the flip-flop. The differential amplifier then decides what polarity the binary output pulse should have, in order to reduce the difference or error, between the audio input signal and the integrated output. At intervals controlled by an external clock, the signal from the differential amplifier is held in the flip-flop until updated by a subsequent signal. A comparison is then made between the present value of the input signal and its value one clock-pulse earlier. If the input signal level is equal to, or greater than, the quantized feedback signal, the flip-flop will produce a positive pulse at its output. If the input is less, a negative pulse will appear at the output. In this way, the positive and negative pulse stream is continually being modified by input signal changes, as the encoder attempts to minimize the error signal.

At the receiving end (the decoder), the binary pulse stream is fed directly to an integrating network (identical to that in the encoder), whose output is a "zig-zag" approximation of the audio input waveform. Low-pass filtering is then applied to suppress all frequencies lying outside the signal passband, including the clock frequency. Since the clock rate is generally many times the maximum frequency of interest, there is no need for very expensive filters with steep-slope characteristics, as would be the case with PCM. However, any distortion elements, lying within the audio band will pass through the system as quantizing noise. At the output of the decoder, an analog signal is recovered which is a near replica of the original audio signal.

Sidney L. Silver is on the supervisory staff of the Telecommunications Section of the United Nations, where he is in charge of sound and recording.

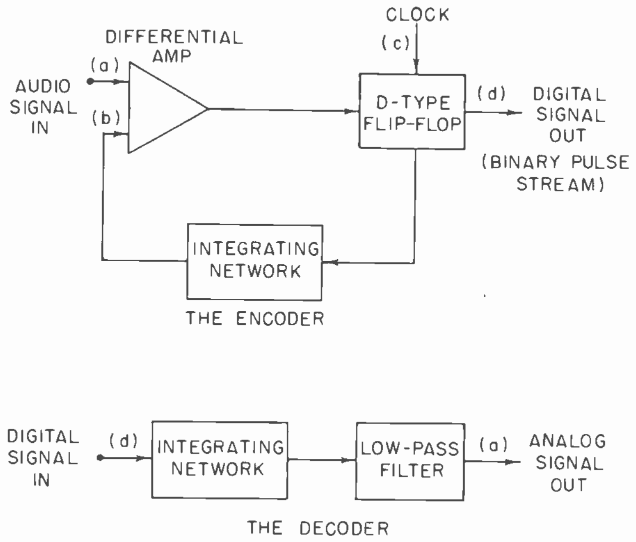


Figure 1. Basic delta modulation system. The letters in parenthesis refer to the waveforms seen in Figure 2.

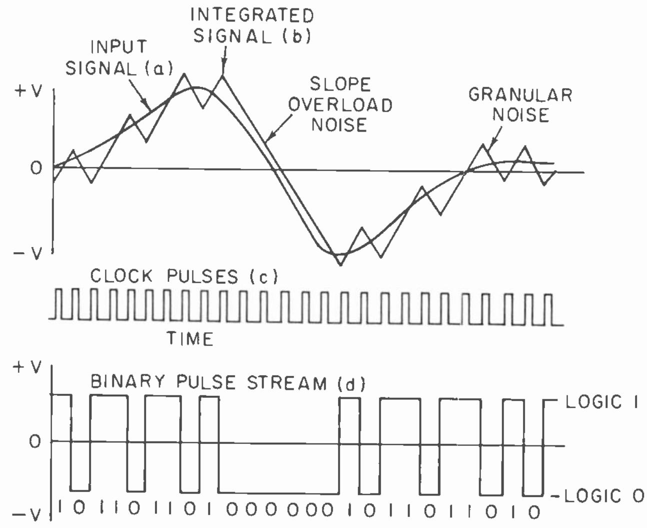


Figure 2. Waveforms illustrating the delta modulation process.

FIGURE 2 shows the waveforms generated in the operation of a linear delta modulator. Here, the output of the decoder integrator (b), is composed of a series of triangular waveforms. The slope of each segment is determined by the instantaneous value of the waveform in relation to the full voltage range of −V to +V. Note that the triangular segments rise at a constant rate during the upward-sloping portion of the audio signal, producing positive pulses at the output. During the downward slope, the triangular segments rise at a constant rate during the erated. With a level waveform (for example, d.c., or no signal at all), the pulse train would alternate evenly between positive, and negative pulses. At each clock time, the rectangular output pulse is assigned either a logical "1" or "0" value. If the sampling process is performed often enough, the digital pulse train will more accurately reflect the variations of the audio signal. Generally, the clock frequency selected is about 20 times the highest audio frequency to be processed, so that for an 8 kHz bandwidth, for example, sampling is accomplished at 5 μsec intervals.

NOISE SOURCES

There are several parameters relating to the quantizing function which must be considered because of their adverse effects on the overall operation of a linear delta modulation system. For example, when the modulator is in its quiescent state (zero input signal), the digital output of the encoder will consist of alternate positive and negative levels, i.e., a logical pattern of 101010. . . . Ideally, when these pulses are integrated at the decoder, the high-frequency triangular components are rejected by the filter. and the output will be zero. However, in practice, because of asymetries in the analog portion of the system, the idling pattern may drift slowly away from zero d.c. reference. The resulting imbalance would produce an unwanted signal at the decoder output, known as idle-channel noise, which would reduce the threshold stability of the system.

Another point to consider in a linear system is that the rate of occurrence of each quantizing step is proportional to the instantaneous slope of the input signal. If the slope increases or decreases so rapidly that the inte-

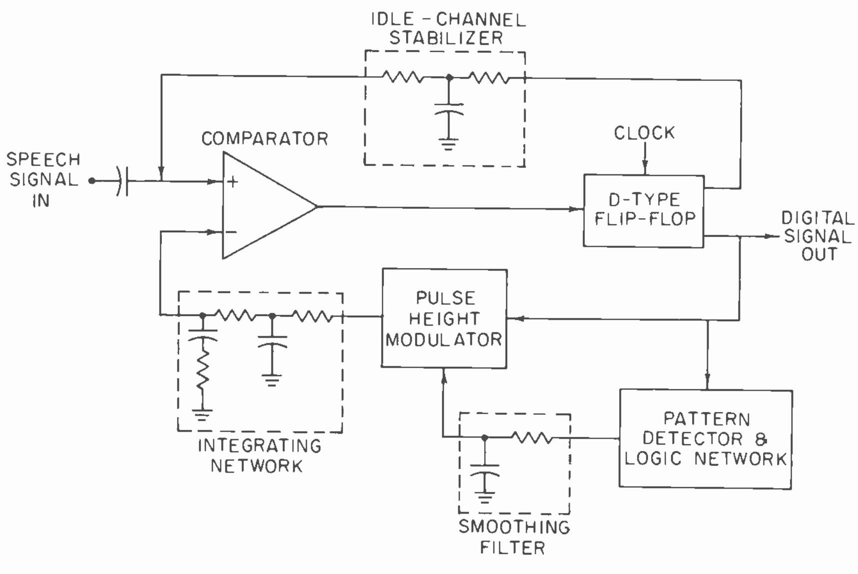


Figure 3. Adaptive delta modulator for speech signals.

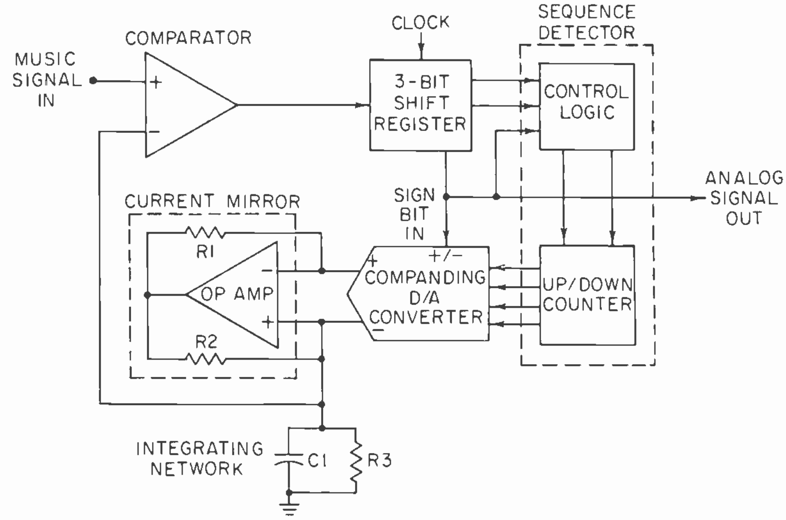


Figure 4. High-quality logarithmic delta modulator for music signals.

grated feedback signal cannot follow the input signal, a sequence of output digits of identical polarity will be generated. Under these conditions, the encoder would be in a state of overload, and the decoded output would no longer resemble the original audio signal. The difference between the rate of change of the input signal and the slope capability of the quantizing process is referred to as slope overload noise. It can be seen from FIGURE 2 that slope overload occurs when the transition from one quantizing step to another does not cross the input waveform. In this situation, the system tends to produce excessive slew rate limitations, and hence, a large amount of transient intermodulation distortion. Audio signals containing high-level, high-frequency components are particularly susceptible to slope overload, and even very small quantities are subjectively quite annoying.

Another inherent characteristic of linear delta modulators is the transmission error existing between the input signal and the decoded output signal. These small differences, called granular noise, have a random amplitude that is dependent on signal frequency and level. Granular noise arises when the quantizing step function hunts around a relatively flat portion of the input waveform; the step size being rather large compared to that segment of the input signal. Referring to FIGURE 2, it is clear that this type of noise is predominant over comparatively gradual slopes of the input signal, while slope overload noise is prevalent over steeper slopes. Let us now consider how noise reduction techniques can be used to minimize these distortion effects.

ADAPTIVE DELTA MODULATORS

It should be noted that the delta modulation scheme just described has been referred to as a linear system, because the circuitry within the feedback loop is a linear network. Obviously, such a system, having a uniform quantizing step size, is incapable of maintaining high signal-to-noise ratios over the full audio spectrum. If, however, during the encoding process the step size could be made large when the slope of the input signal is steep, and small when the slope is gradual, the integrated feedback signal would be able to track the audio signal over a wide range of input levels and frequencies. During decoding, small digital inputs would thus produce proportionately smaller analog changes at the output from one voltage to the next, while large digital inputs would create proportionately greater analog changes. This contrasts with linear delta modulation, in which the quantizing step is a constant percentage of full scale, regardless of whether the actual signal is weak or strong.

Various "adaptive" methods have been devised for dynamically changing the quantizing step size, but for the digitization of speech signals, the most commonly used is the syllabically-companded delta modulator.[4] In this type of modulator, the step size of the feedback is continuously varied in a manner dependent on the envelope of the speech signal, rather than the instantaneous value. FIGURE 3 shows how the encoder works. Initially, the digital output is used as a control signal to drive a pattern detector and logic network in the feedback loop of the

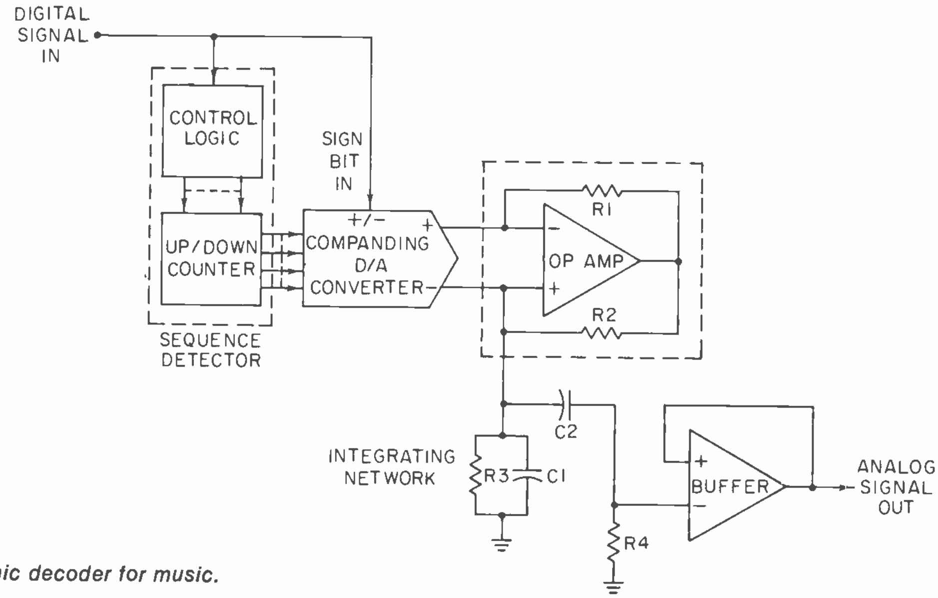


Figure 5. Logarithmic decoder for music.

modulator. Then a smoothing filter is added to obtain the required time constant, appropriate to syllabic changes in the speech signal. To sense the overload, the bit stream is examined in the logic network to detect successive strings of logical "1"s or "0"s. When this happens, the system strives to change the size of the quantizing step so that it can follow rapid fluctuations of the input signal more effectively. This is accomplished by feeding the averaged control signal to a pulse-height modulator, which multiplies this signal with the binary pulse stream taken directly from the output of the encoder. The resultant multi-level signal at the output of the pulse-height modulator is then used to modulate the size of the quantizing step applied to the integration network. The integrator is a modified double RC arrangement, which shapes the overload characteristic of the encoder to the long-term spectral density of the speech input signal. To close the feedback loop, the integrated output is applied to the inverting (−) input of the comparator, and this signal is used to closely track the audio input signal at the noninverting (+) input. In addition, a low-frequency network is used to feed back a small d.c. voltage to the noninverting input, in order to stabilize the idle-channel pattern.

Adaptive delta modulation systems using a continuously-variable slope are commercially available for application in speech communication systems. These devices are monolithic integrated circuits, which may be connected as encoders or decoders, or switched from one function to the other with simple circuitry. Moreover, they contain all the necessary active analog and digital circuitry, but also allow the user to connect the appropriate RC networks externally, according to application.

COMPANDED DELTA SYSTEM FOR MUSIC

At the present state-of-the-art, adaptive single-chip encoders for digitizing high-fidelity music have not yet been developed. However, satisfactory results have been achieved by incorporating a companding digital/analog converter in the feedback loop of an adaptive delta modulator. These modulators operate on the principle of compressing the dynamic range of the input signals according to a logarithmic characteristic, so that the quantizing step remains small compared to the audio input. In the demodulator, operating in an open-loop configuration, a complementary expansion takes place to restore the signals to their original form, with a corresponding reduction in quantizing noise. Thus, by providing an analog output that is exponentially-related to the digital input, even relatively-low clock rates can be used to process a wide range of amplitudes and frequencies, with very low distortion.

FIGURE 4 shows a block-schematic diagram of a high-quality logarithmic adaptive delta modulator[5] designed to cover the audio range from 20 Hz to 20 kHz. In this circuit, the comparator output determines the sign of the difference, or error, between the input voltage and the

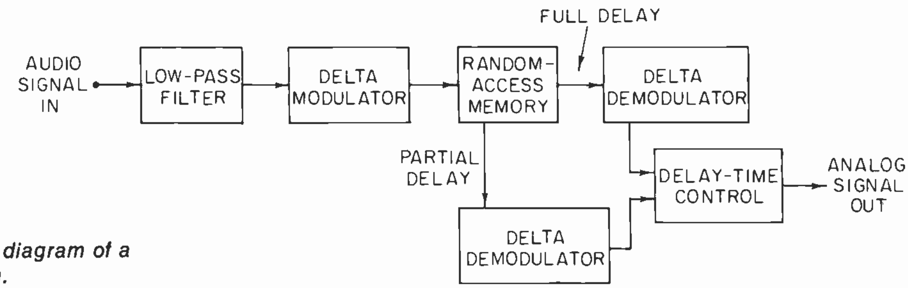


Figure 6. Simplified block diagram of a digital audio delay system.

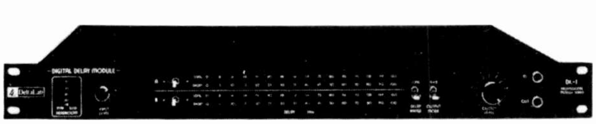
voltage developed across the integrator, C_1. This error signal is clocked into a 3-bit shift register at the sampling frequency, while a sequence detector monitors the pattern of "1"s and "0"s in the register. Here, the shift register may be regarded as a string of D-type flip-flops, arranged so that the first flip-flop can shift its data to the next one in line, just before it is required to remember new data. Information about the evolution of the input signal is thus taken from the output pulse train, so that the value of the signal at each clock instant is estimated to be a function of the past history of the quantized value. This data is then taken into account in predicting the slope overload, or slew-rate limitation of the system.

In detecting the overload state, the control logic circuit sends either an upcount or downcount to the up/down counter—the output of which drives the digital/analog converter. If, in the coding process, the shift register contains an alternate sequence of "1"s and "0"s, i.e., 101 or 010, the slew rate of the error signal is too fast, so that overtracking occurs, and the counting device counts down accordingly. If, however, there are a series of consecutive "1"s or "0"s stored in the register, the slew rate is not fast enough, and the device counts up. The counter output is then applied to the D/A converter, which transforms the number into a charging rate for the integrating capacitor, C_1. Thus, when the system is in slope limit, the quantizing step is increased, allowing the integrator to follow rapidly-changing audio signals. Conversely, when the integrator cannot follow slowly-varying audio signals, indicating a hunting behavior, the step size is reduced.

At the decoder, the digital signal is demodulated in a simple open-loop circuit, with the comparator and shift register removed from the system. As shown in FIGURE 5, the binary input pulses are fed directly to the sequence detector, and after being decoded by the A/D converter, the integrated signal is a.c.-coupled to the buffer to produce the analog output signal.

It should be pointed out that presently-available i.c. logarithmic D/A converters are characterized by current-sinking outputs. Current is thus pulled into the device by either the positive or negative output terminals, depending on the logic state existing at the sign bit input. When the sign bit is high, i.e., a logic "1", all of the output currents flow into the positive terminal, and when the sign bit is low, i.e., a logic "0", the currents flow into the negative terminal. In order to provide source current to the integrating capacitor, C_1, a fast, temperature-stable current mirror is needed to "invert" the current flow into the D/A converter's output terminal on positive-error signals. The converter must then be able to pull current against the mirror when swinging negative. To accomplish this, a precision op amp is used to monitor the input current at its inverting input, which is the A/D converter's sinking current. By using matched feedback resistors, R_1 and R_2, and trimming the offsets, the op amp will source an equal current at its noninverting input, which is loaded by the integrator. Nevertheless, should the leakage currents on the two inputs of the op amp be unequal, a d.c. component flowing into C_1 may cause saturation effects, and the integrator would not function properly. In this case, a large-valued bleeder resistor, R_3, can be placed across C_1 to eliminate the problem.

A device which utilizes the delta modulation process—the DL-1 Acousticomputer by DeltaLab.



AUDIO DELAY APPLICATION

Adaptive delta modulators are now being utilized in digital time-delay systems, to effectively re-create the echoes and reverberation produced by reflections in a concert hall. In FIGURE 6, an audio delay line of adjustable length is formed by inserting a random access memory (RAM) between the delta encoder and decoder. Before encoding, the bandwidth of the audio input signal is reduced to about 7500 Hz. This is done to simulate the sound-propagating characteristics of a typical concert hall, which loses most of its high-frequency energy through absorption. Since the signal is encoded as a stream of binary digits, it can be delayed by an amount determined by the rate at which the data is fed into the RAM. Here, the input samples are read into successive memory locations at the clock rate, and subsequently read out of the memory, in sequence as needed. Each memory element is addressable so that very-fine resolution in the amount of delay is possible. At various stages, the pulses can be retrieved and decoded to recover the audio signal. The result is an output signal containing a multiplicity of delayed signals, decaying gradually in amplitude, which can be re-cycled repeatedly through the time-delay system. ■

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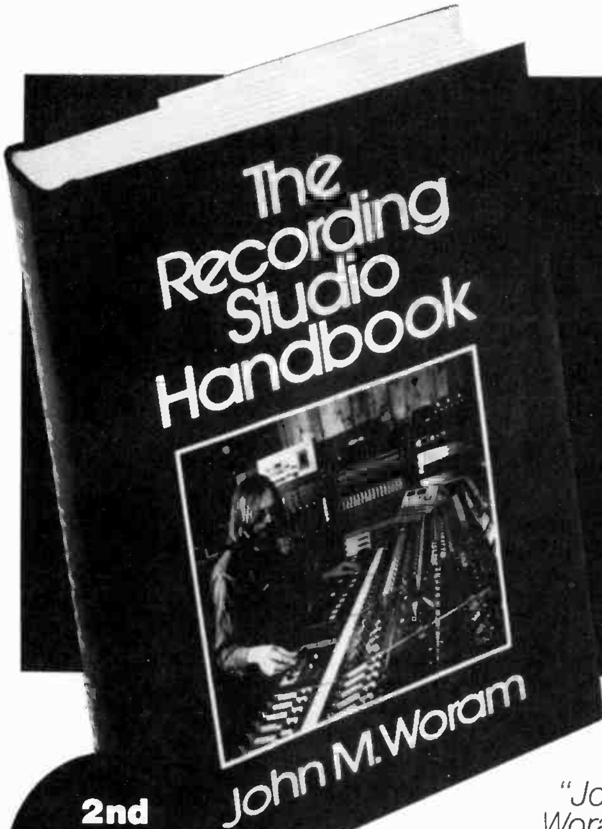

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FOR SALE: Westrex 3DIIH, $3,495.00; Haeco SC-2, $4,800.00; Haeco SC-1, $1,495.00; Grampian D, $385.00; Grampian BI/D, $325.00; Westrex 2B, $525.00; All cutterheads reconditioned and in specs. Used HAECO Cutterheads available. **International Cutterhead Repair, 194 Kings Ct., Teaneck, N.J. 07666. (201) 837-1289.**

TEST RECORD for equalizing stereo systems; Helps you sell equalizers and installation services; Pink noise in ⅓ octave bands, type QR-2011-1 @ $20. Used with various B&K Sound Level Meters. **B&K Instruments, Inc., 5111 W. 164th St., Cleveland, Ohio 44142.**

AMPEX, OTARI & SCULLY recorders in stock for immediate delivery; new and rebuilt. **RCI, 8550 2nd Ave., Silver Spring, MD 20910.** Write for complete product **list.**

THE LIBRARY . . . Sound effects recorded in STEREO using Dolby throughout. Over 350 effects on ten discs. $100.00. Write **The Library, P.O. Box 18145, Denver, Colo. 80218.**

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.**

TASCAM MODEL 10 input & output modules. Inputs $160.00, Outputs $110.00. **Jay Sound, 4300 Watertown Rd., Maple Plain, MN. (612) 475-3151.**

WANTED
RECORDING EQUIPMENT OF ALL AGES AND VARIETIES
microphones, outboard gear, consoles, tape decks, etc.
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(213) 934-3566

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

12½ in. NAB 0.050 in. ALUMINUM FLANGES. This new item in stock for immediate delivery. For prices, call or write, **RECORDS RESERVE CORP., 56 Harvester Ave., Batavia, N.Y. 14020. (716) 343-2600.**

PROFESSIONAL AUDIO COMPONENTS: AKG mics; Badap 1; Crown; dbx; Delta Lab; Eventide; Frazier; Gauss; GLI; Ivie; Malatchi; MasterRoom; Nagra; Neumann mics; Orban; Otari; Pentagon; RTR; Sennheiser mics; Switchcraft; Tascam; Uni-Sync; and UREI. These products are on demo in our showroom and in stock for immediate delivery. Our shipping is insured and prepaid. **Barclay Recording & Electronics, 233 E. Lancaster Ave., Wynnewood, Pa. 19096. (215) 667-3048 or 649-2965.**

REK-O-KUT drive belts. Specify model. $9.95 delivered. **QRK Electronic Products, 1568 N. Sierra Vista, Fresno, Ca. 93703.**

MCI JH 428 PROFESSIONAL recording console. Excellent condition, fully operational, 3 years old, 28 x 24 with quad. & stereo outputs, light meters, full patch bay and power supplies. $25,000. For more information write: **Console, P.O. Box 3015, Washington, D.C. 20010. Or call (202) 347-1420.**

SOUND LEVEL METER and analyzer, General Radio Precision Type 1933 with ½-inch microphone. **(201) 540-8811.**

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18 CHANNEL INPUT & expander "UNI-SYNC" Trouper 3. Fully assignable microphone to 4 outputs or mix busses—including road case. Demo unit—NEW. $2000.00 FOB. **American Music Network, Inc., 1031 E. Belmont Ave., Fresno, CA 93701. (209) 486-1641.**

SONY DXC-1200 color video camera. Like new with tripod. Call **Bill (419) 385-4724.**

CUSTOM DeMedio recording console; 32 inputs, 24 outputs, full patch bay. **Wally Heider Recording (213) 466-5474.**

ARP 2600 SYNTHESIZER, $1,400.00, Polyfusion series 2000 system, a complete studio system (modular) $7,500.00. Both synthesizers in like new condition; **John Bement, RFD 2, South Royalton, Vermont 05068. (802) 763-8212.**

REELS AND BOXES 5" and 7" large and small hubs; heavy duty white boxes. **W-M Sales, 1118 Dula Circle, Duncanville, Texas 75116. (214) 296-2773.**

THE EXR AUDIO exciter will do more for your audio chain or P:A. system than outboard EQ could ever do. If you don't agree, we'll buy it back. We also stock Marshall, Lexicon, Eventide, DeltaLab and Loft. Call us. **ASi Pro Audio (512) 824-8781.**

MCI 24 TRACK with 16 track heads. Well maintained work horse—reasonable **M. Guthrie (212) 581-6505.**

AMPEX ART-102, ten hours total use, $5,200; Revox A-700, twenty hours use, $1,800; Crown Model 700, ten hours use, $900; **Telecomputer, P.O. Box 2798, Opelika, AL 36801. (205) 749-8261.**

WANTED

USED RECORDING equipment for sale. **Dan (415) 232-7933.**

WANTED IMMEDIATELY, experienced maintenance man for major New York recording Studio. Please send resume to: **Dept. 42, db Magazine, 1120 Old Country Rd., Plainview, NY 11803.**

WANTED: ONE USED 16-track erase head only. Good Condition. To be used with MCI JH-16. **Al Gambino, 5972 South Haught St., Las Vegas, Nevada 89119.** Or call **(702) 736-7305.**

EMPLOYMENT

ENGINEER, EXPERIENCED with clients or salesperson in recording/advertising wanted for position at 24 track recording studio and production company on Long Island. Start on a commission basis. Call **(516) 364-8668.**

TECHNICAL MANAGEMENT AVAILABLE

STUDIO CHIEF ENGINEER with over 10 years experience in maintenance, design, and operation of tape, film, disc, digital and video systems at major N.Y.C. broadcast and recording studios. If you specialize in state-of-the-art audio and require full time technical assistance for success in this challenging field, write **Dept. 41, db Magazine, 1120 Old Country Rd., Plainview, N.Y. 11803.**

BUSINESS OPPORTUNITIES

FOR LEASE: Denver recording studio. 6,000 square feet; two control rooms . . . "live" chamber. Attractive offices . . . ample parking. Centrally located near downtown. Plus equipment for sale: Audio Designs board . . . *mint*; two ampex, 440 B mono machines; two 3M, M64 2-track machines; assorted speakers, amplifiers, etc. Write or call . . . **Fred Arthur Productions, Ltd., 1218 E. 18th Ave., Denver, CO 80218. (303) 832-2664.**

RECORDING STUDIO for rent, monthly basis. 16 track MCI studio, full noise reduction. $1500.00 monthly rate, plus terms. **Cathedral Sound, Inc., 1575 5th St., Rensselaer, NY 12144. 1-518-465-5689.**

SERVICES

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. Service available at **2201 Lunt Ave., Elk Grove Village, IL 60007; 500 Rodier Dr., Glendale, CA 91201; 75 Commerce Way, Hackensack, NJ 07601.**

ACOUSTIC CONSULTING – STUDIO ANALYSIS, ROOM EQUALIZATION. **Sugarloaf View, Inc., 31 Union Square W. New York, N.Y. 10003. (212) 675-1166.**

SPECIALISTS IN THIELE cabinet designs & conical exponential horns; tour systems, permanent installations using Electro-Voice, BGW, White, dbx, Soundcraft, Harrison, Otari, David Clark, DeltaLab, Lexicon, Sennheiser, Sony and Wireworks components. Contact: **Engineering and Sales Group, Star Tech Sound, Inc., (609) 779-7887.**

PARTS-SERVICE, Akai, Ampex, Professional & Consumer, Dokorder, Pioneer, Tandberg, Teac, etc. **Electronic Engineers, Inc., 1639 West Evergreen Ave., Chicago, IL 60622. (312) 227-2600.**

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

People/Places/Happenings

• In a suit filed against **Clair Brothers Audio Enterprises, Inc.** of Lititz, Pennsylvania, **Linda Joyce Wheatcroft**, a sound technician, is charging sex discrimination in hiring practices. The suit stems from an incident during the spring of 1976 Peter Frampton Tour.

Ms. Wheatcroft is seeking permanent injunctive relief from discrimination against her with respect to employment because of her sex, an accounting for loss of earnings incurred as a result of Clair Brothers Audio Enterprises, Inc. unlawful refusal to hire her, and money damages resulting from the unlawful employment practices.

• Named vice president of **Fantasy Studios** and engineering for **Fantasy/Prestige/Milestone/Stax, Jim Stern** is currently supervising the installation of a new state-of-the-art fourth studio at Fantasy. Mr. Stern has been with Fantasy since 1969.

• **Ron** and **Howard Albert** have been named vice presidents of **Criteria Recording Studios,** Miami, Florida. The Albert Brothers, currently co-presidents of **Good Sounds Records,** a subsidiary of Criteria, and partners in their own production company, **Fat Albert Productions,** have been with Criteria for more than a dozen years.

• **Lewis Barrett** has been named executive vice-president of **Northshore Marketing**, the professional audio marketing firm based in Redmond, Washington. Mr. Barrett was formerly the national sales manager for the **Otari Corporation.**

• **Spectra Sonics** has completed the construction of its new factory at 3750 Airport Rd., Ogden, Utah 84403. Their phone number, (801) 392-7531, remains unchanged.

• Appointed manager of engineering for the broadcast division at **Sony** Video Products Company, **Frank Brown** will assume the responsibility for the development of an engineering group for Sony Broadcast and the management of national parts distribution. Previously, Mr. Brown served as manager of sustaining engineering with **Ampex Corporation.**

• **The New England Conservatory of Music** will hold a Summer session from June 25-August 3, 1979 featuring workshops, courses, and master classes. Among the offerings of special interest is the Electronic Music Workshop with **Robert Ceely,** June 25-29. For more information on this or other offerings contact: **Robert L. Annis, Director of Summer School, New England Conservatory of Music, 290 Huntington Ave., Boston, MA 02115.**

• Overseeing the manufacturing at all **Electro-Voice** facilities, **Bernie Ullom** has been promoted to the position of vice president, manufacturing of Electro-Voice. In addition, **F. Davis Merrey** has been appointed vice president and general manager of **Technical Audio Products Company (TAPCO).** an operating unit of Electro-Voice.

• **Clarence C. Moore,** founder and president of **Crown International, Inc.,** passed away on January 24, 1979, at the age of 74.

During the late 1930's, in response to the problems of altitude on short wave broadcasting, Mr. Moore developed the non-voltage or loop antenna, and received a patent for his invention of the cubical quad antenna system.

In the early 1950's, Mr. Moore developed the first tape recorder which included a power amplifier. Demand for the recorder prompted Mr. Moore to start the Crown line of audio products, working out of a small farm building. In 1953, Mr. Moore introduced the concept of the three motor mechanism to tape recorder technology, as well as his patented electromagnetic braking system.

In the ensuing years, Mr. Moore gained further attention and respect for his work in the audio field through the creation of the first ¼-inch 4-channel recorder and the introduction of the first solid state power amplifier.

• Appointed marketing manager for broadcast products at **Orban Associates Inc., Jesse Maxenchs** will be responsible for the promotion and sale of Orban products directed to the radio and tv broadcaster. Mr. Maxenchs brings with him, over 15 years experience in broadcast equipment marketing with companies such as **ESA, Inc., Belar,** and **AEL.**

• **KSR Studios**, a new 24 track facility, is fully equipped with MCI equipment, UREI Time-Align monitors, and a complete array of outboard equipment. The studios are located at 1680 Vine Street, Suite 515, Hollywood, CA 90028 (213) 467-0768.

• Executive promotions in the sales and technical markets departments at **Shure Brothers Inc.** include: **Allen R. Groh,** formerly manager, technical markets and product management, promoted to manager, high fidelity products market group; **Lottie Morgan.** promoted to sales manager, with responsibility for management of sales to retail outlets within the United States; and **John A. Owens**, previously account manager, named manager, product management and OEM sales.

• A new rep firm organized to serve the professional audio, sound reinforcement and musical instrument trade, **Professional Audio Associates,** located at 104 Drake Road, Burlington, Massachusetts 01803, will cover dealers in the New England and upstate New York areas. Initial lines include Unisync, RSD/Studiomaster mixers, and Advanced Audio Designs amps, preamps and delay lines. Their telephone number is (617) 926-6107.

• **QUAD-EIGHT** of North Hollywood, California, has promoted **Mark Pinske** to technical sales director and appointed **Ron Bennett** director of marketing. Before coming to QUAD-EIGHT, Mr. Pinske toured as a musician and had also worked on three Broadway musicals. Mr. Bennett has served as QUAD-EIGHT's sales manager since 1974.

THE BEST VALUE IN A PROFESSIONAL TAPE RECORDER

When you evaluate a tape recorder, here are the most important areas to consider for value, quality, and sound.

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 CHOICE IS OBVIOUS

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Call Ruth Pruett on 415/593-1648 for the name of your nearest Otari professional dealer.

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In Canada: BSR (Canada, Ltd.), P.O. 7003 Station B, Rexdale, Ontario M9V 4B3 416/675-2425

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At Altec Lansing, we've been making high-quality loudspeakers for over forty years. And we're very proud of the reputation that they've earned during that time. But perhaps it's because we're so well known for our speakers that some people tend to forget that we also make a full line of professional electronics. Equipment that's built with the same quality and reliability that our speakers are famous for.

Case in point: The Altec Lansing 9440A power amplifier.

The 9440A is a dual-channel power amp that delivers the high performance standards that today's audio professional needs. It produces more than 200 watts per channel into 8 ohms of highly reactive loudspeaker load. At 4 ohms the power is typically greater than 400 watts. And even at these levels the 9440A has less than 0.25% THD or IM distortion and a frequency response that's flat ±0.25 dB from 20 Hz to 20 kHz.

Of course high performance must be matched with high reliability. The 9440A is designed to meet the most demanding conditions. Day-after-day. Year-after-year. Reliability provided by sixteen 250-watt hometaxial power transistors backed up by a massive die-cast aluminum heat sink. Reliability ensured by an efficient VI limiter, a unique 40% power-limiting circuit and an output relay that protects against dangerous turn-on/turn-off transients. Reliability good enough to earn both UL and CSA approval.

And because we think that an amplifier should do more than just amplify, we've incorporated some features in the 9440A's design that will help make life a little easier. Features like lighted VU meters, meter range switches and provisions for adding plug-in input transformers. Features like a front-panel-mounted switch that converts the 9440A into a single-channel amplifier with a true balanced output. Features that help make the 9440A a versatile addition to any sound system.

But perhaps the best feature of the 9440A is that it's from Altec Lansing—a leader in quality audio products for over four decades.

So if you're thinking about power amps, think about Altec Lansing. Check the Yellow Pages under Sound Systems for the name of your local Altec Lansing sound contractor. And meet the great power behind our speakers.

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