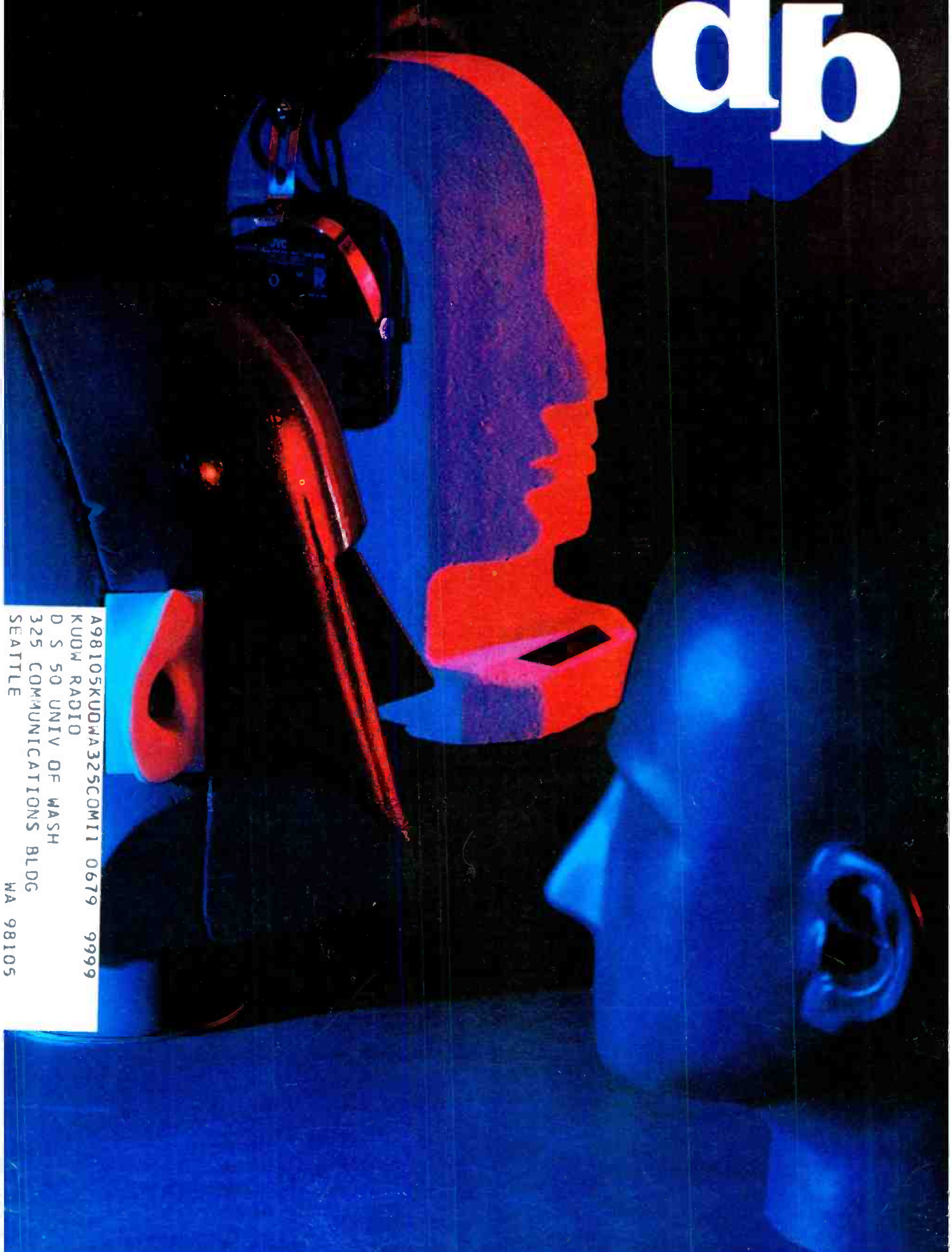
ISSN0011-7145

THE SOUND ENGINEERING MAGAZINE

JUNE 1979
$1.95

db



Coming Next Month

• Semi-pro recording — yesterday, there was no such thing. "Professionals" dominated the field, while amateurs stayed at home and made tape copies.

Today, semi-pro is very big business. In July, we'll take a closer look at this aspect of the industry, to see what effects—if any—has the semi-pro had on recorded sound.

And by the way, what *is* a semi-pro? Tune in next month, as we offer several possible answers to the question.

About The Cover

• Don't let Robert Wolsch's cover photo mislead you—psycho-acoustics is not a subject for dummies. For a little more on the subject, see this month's table of contents.

db

THE SOUND ENGINEERING MAGAZINE

JUNE 1979 VOLUME 13, NUMBER 6

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

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db Letters

To The Editor:

I have read, with more than just passing interest, the recent "Letters" column where the broadcast engineer is blasting the recording engineer and vice versa. Some letters made reference to the December 1978 copy of **db**; so I flipped through last year's copies to find out what started all the debate. After reading every letter; every Chief Engineer's opinion, and every sound engineer's rebuttals—I thought I'd write, too. You see, I AM on both sides of the controversy. I co-own a 16-track recording studio, and am a working Disc Jockey/engineer. Daily, I have to make 'sound' decisions as to how I should record it in the studio—and what will happen to my 'creation' once it hits the 'processed' airwaves. I could take issue with engineer Mr. Dunn. I could also take issue with the acoustic and scientific data that fills the pages. But I won't. It appears that the bottom line is there will always be a battle between the two factions. THEY ARE TWO DIFFERENT MEDIUMS. The recording engineer has no transmitter, and the radio chief doesn't work with a Producer in the studio THAT WANTS IT THAT WAY. May I remind the feuding parties of an old addage that seems to fit the situation: "Opinions are like rear ends—everybody has one!" Therefore, everyone is entitled to his opinion. So, let's get back to work fellas.

Martin Ashley
Vice President/Chief Engineer
Heavenly Recording Studios, Inc.

To the Editor:

As a **db** subscriber I would appreciate some information or reference material on a very early amplification device called the "Singing Flame" or "Talking Flame." This device utilized a gas flame or possibly an electric flaming arc. Considering its crude nature it must have preceeded, by quite some time, the DeForest audion tube.

If any of your staff can provide any information pertinent to this request, your assistance will be appreciated.

Barney B. Rogan
Merritt Island, Fl.

db replies:

How about it, readers? Can anyone help Mr. Rogan by supplying some information?

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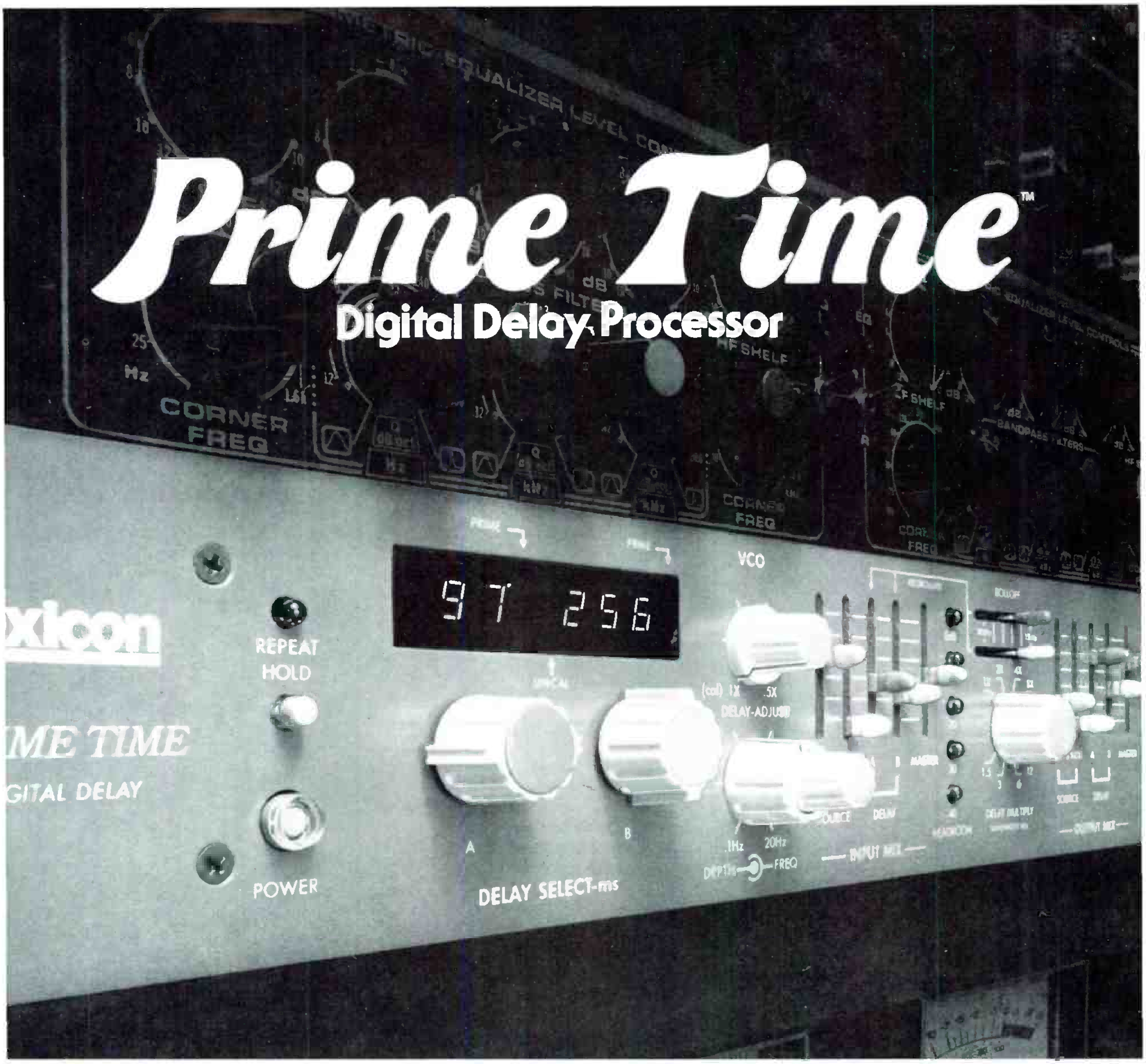

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PATRICK S. FINNEGAN

F.M. Proof-Of-Performance

• All f.m. broadcast stations are required by the FCC Rules to make an audio proof-of-performance of its broadcast system at least once each calendar year. This is a specific requirement of the Rules, and what must be measured is not left up to the discretion of the station. Part 73.254 (commercial stations) or Part 73.554 (educational stations) details what measurements are to be made, while Part 73.317 defines the minimum technical tolerances which must be met. Our discussions this month will center on this annual activity the f.m. broadcast station must perform.

WHAT IT IS

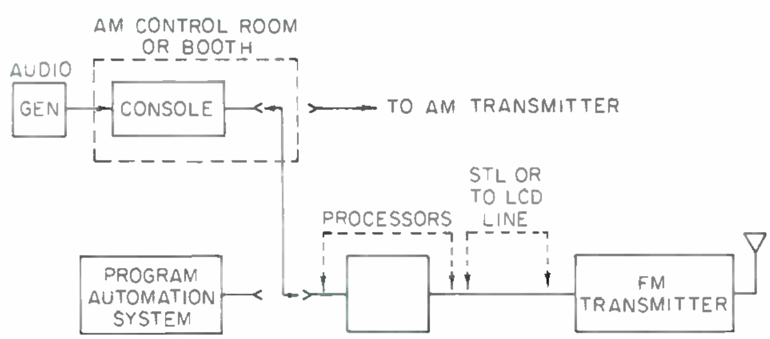
The Rules predate the advent of program automation systems, and envision the usual "live" type of operation. Many f.m. stations today do have a "live" type of operation. The proof measures this "core" or main chain of the broadcast system—the main microphone preamplifier input, the console, processors, telco line or STL, and the transmitter loaded with its antenna system. The station which makes use of an automatic programmer 100 per cent of the time will find it a little more difficult to meet this "core" requirement. But such stations ordinarily provide some method of going on the air "live" if the programming should fail. This may be a production booth, a simulcast arrangement with a sister a.m. station, and so forth. Such a main alternate route should be used for the measurements in those instances. As far as the proof is concerned, all the peripheral equipment and automatic programmers are not a part of this "core" or basic chain and are not measured at this time. These other pieces of equipment certainly require maintenance so that they provide good technical quality audio programming to the station, but are not considered for the proof itself.

The proof is a specific set of measurements to be made to the basic chain to demonstrate that it meets the minimum technical tolerances of the Rules. The station must meet these minimum tolerances *all the time*, not just one night a year. And a station cannot "fail" a proof. Anything operating out of tolerances must be corrected and measured to demonstrate that the system does meet the minimum standards. We should point out here that the proof is a *mono* proof. These requirements of the Rules far predated latter day refinements such as stereo, SCA, and so forth.

RESPONSE

The first parameter to be measured is the response of the system across

Figure 1. For those stations 100 per cent automated, use the alternate or stand-by arrangement normally provided.



"This Auditronics 501 was one of TM Productions' first

boards six years ago and it still runs a tightly packed schedule of original vocal session recording and mixdowns", says Ken Justiss, Operations Manager of TM Productions in Dallas. "Since we do more commercials and station ID's than anybody else in the world, we produce literally thousands each year, and at some point they've all gone through this Son-Of-36-Grand (serial number 011)."

"There's not a faster board to work with than the Auditronics 501 whether we use it for building demos or complex production tasks. It's compact; all its controls are so very accessible even trainees become proficient on it quickly."

"Its reliability is outstanding. We've literally worn out the faders once, and we've changed a switch or two, but the things I've seen go wrong with this board in six years are so minor, it's a waste of time to even talk about. It's an excellent creative tool and I'd find it hard to fault our Auditronics 501 in any area. In fact, our success with this board was largely responsible for our buying three more Auditronics consoles."

If you'd like to know more about what Ken Justiss at TM Productions and over 300 other satisfied users think about Auditronics consoles, circle reader service number or contact:

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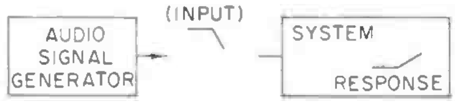


INPUT		PLOT	
100 Hz	= +1 dB	100 Hz	= −1 dB
1 kHz	= 0	1 kHz	= 0
5 kHz	= −7 dB	5 kHz	= +7 dB
10 kHz	= −13 dB	10 kHz	= +13 dB
15 kHz	= −16 dB	15 kHz	= +16 dB

Figure 2. The measured input signal voltage is the inverse of the system response. To plot the curve, change the sign.

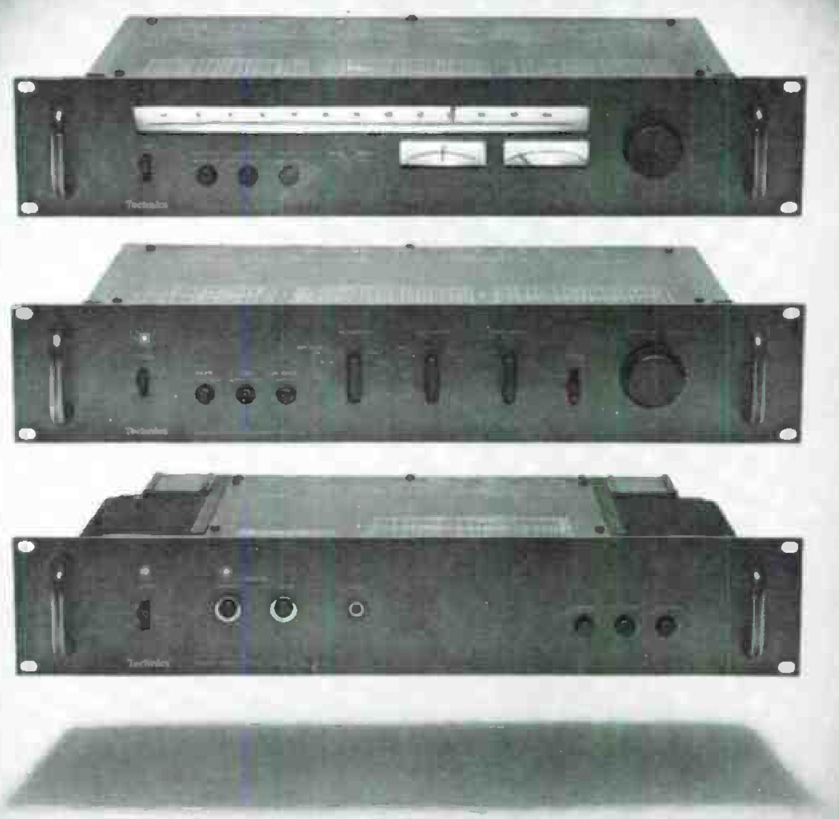
the audio band-pass. Three sets of measurements are required. One set at each of these levels of modulation: 100, 50 and 25 per cent. The basic audio tones for modulation which are required are: 50, 100, 400 Hz, 1, 5, 10, 15 kHz. These measurements are to be made with the standard 75 μsec. pre-emphasis network in the system. The resulting audio response curve of the system will describe this pre-emphasis curve, and it must fit within the tolerances of the curve as shown in FIGURE 2, Part 73.333 of the Rules.

Before making the measurements, the modulation monitor itself should have been checked for proper calibration. If the monitor is properly calibrated, then feed a low frequency, audio tone—1 kHz or lower—into the mike pre-amp of the console. Go through and disable the action of all AGC, limiting, and similar response adjusting amplifiers, thereby running them as straight amplifiers. Modulate the transmitter to 100 per cent on the reference and measure the signal level of this audio input signal. Make no change to gain controls of the system after this calibration; any change required should be done at the audio signal generator. Proceed to measure the input signal level for each tone, maintaining the same modulation level. These input levels must be measured accurately and recorded. They will be the inverse of the actual system response, so the sign must be changed when plotting these on a graph. To prevent seriously overmodulating the transmitter because of the pre-emphasis, always reduce the signal generator level to the system before changing to a higher frequency audio tone.

DISTORTION

Harmonic distortion in the audio band-pass is the next important measurement required. A series of distortion measurements, at the same audio tones and modulation levels as for response measurements, are required. What is measured, here, is the total harmonic distortion of the audio

You're looking at three ways Technics pursues the one ideal. Waveform fidelity.



Waveform fidelity. It should be the objective of any professional component. Because perfect waveform fidelity would mean an output signal that's a mirror image of the input signal.

How do our engineers pursue this elusive goal? To begin with, they use two automatically switchable IF bands in the ST-9030 FM tuner. A narrow band for extra-sharp selectivity. And a wide band for extra-high S/N and extra-low distortion. But just as significant is a pilot-cancel circuit which Technics developed for high-frequency waveform fidelity. Even the basic tuning function in the ST-9030 is unique. Like an 8-ganged tuning capacitor for outstanding reception.

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Finally there's Technics SE-9060 amp. It's DC like our preamp. Has a frequency response of 0-100 kHz (+0, −1 dB). And a "strapped" circuit for more than double the power in a multi-amp system. Compare specifications and prices. We think you'll agree. There's no comparison for these Technics components.

ST-9030. THD (stereo, 1 kHz): Wide—0.08%. Narrow—0.3%. S/N (mono): 80 dB. S/N (stereo): 73 dB. FREQUENCY RESPONSE: 20 Hz—18 kHz +0.1, −0.5 dB. SELECTIVITY: Narrow—90 dB. CAPTURE RATIO: Wide—0.8 dB. IF, IMAGE and SPURIOUS RESPONSE REJECTIONS (98 mHz): 135 dB. STEREO SEPARATION (1 kHz): Wide—50 dB.

SU-9070. PHONO MAX. INPUT VOLTAGE (1 kHz RMS): MM—380 mV. MC—9 mV. S/N MM—100 dB (10 mV input). MC—72 dB (60 μV). FREQUENCY RESPONSE: Phono 20 Hz—20 kHz (RIAA $\pm$0.2 dB).

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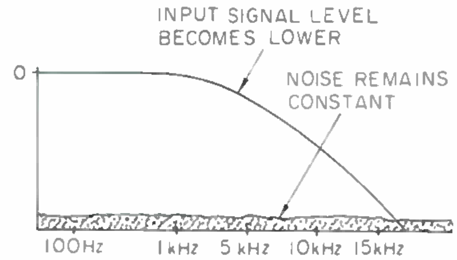


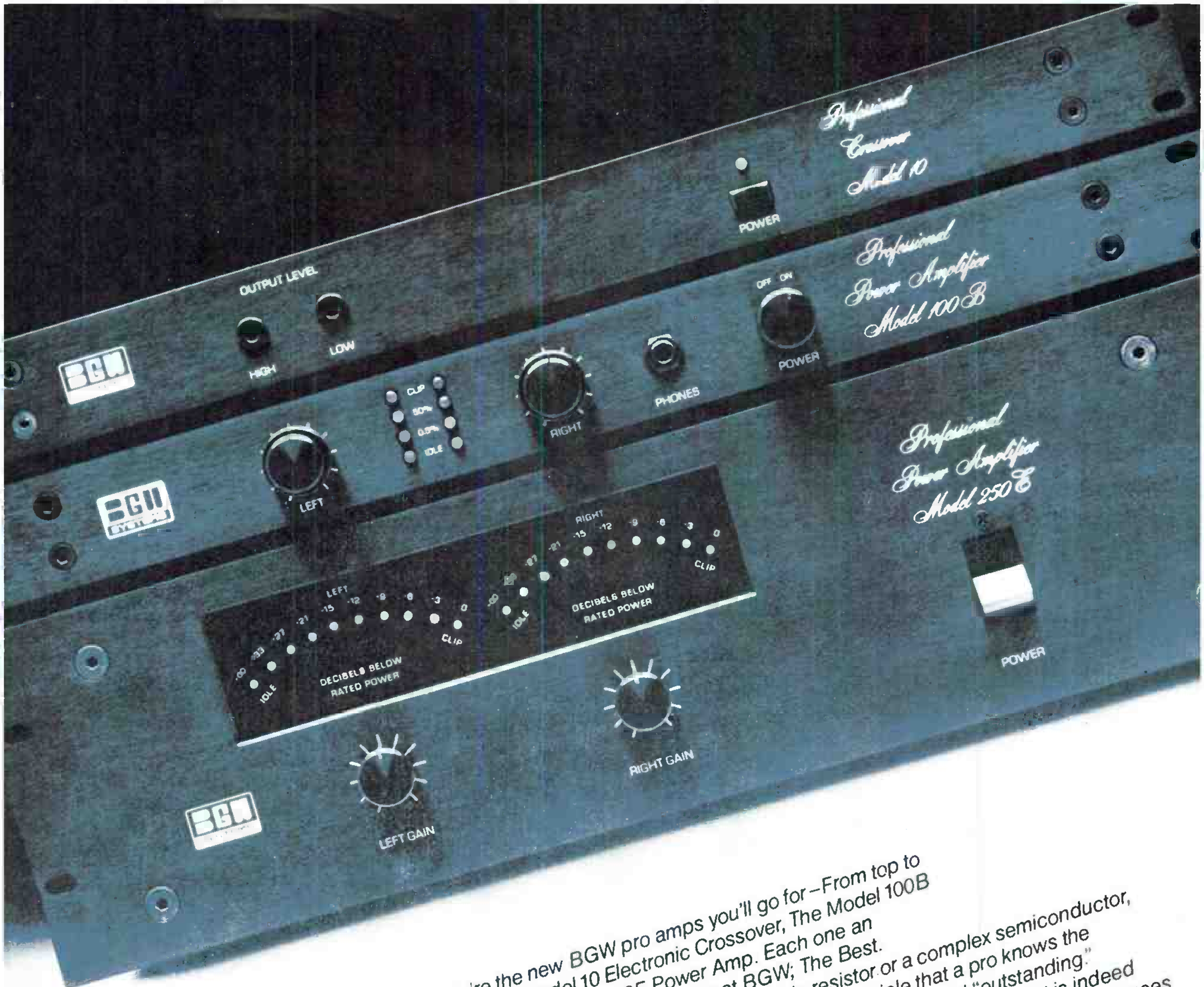
Figure 3. Because of transmitter pre-emphasis, noise can become a significant part of the measurements.

band-pass. Most distortion analyzers today make use of a null network to eliminate the fundamental tone fed to the analyzer, and measure everything else as distortion. Noise can become a significant part of this measurement. This is more true of the f.m. system than the a.m. system because of the tighter tolerances for distortion, and because of the 75 μsec. pre-emphasis in the system. This noise element is the reason distortion measurements are not required at the higher frequency audio tones and lower levels of modulation. A set of distortion measurements are required at each of these levels of modulation: 100, 50, and 25 per cent. Each set to be made at these tones: 50, 100, 400 Hz, 1, 5 kHz. Two additional measurements are required at 100 per cent modulation at these tones: 10, 15 kHz. Minimum tolerances the system must meet are: 3.5 per cent at 50 to 100 Hz; 2.5 per cent at 100 Hz to 7.5 kHz; 3 per cent at 7.5 kHz to 15 kHz. Although the system itself must have the 75 μsec pre-emphasis in it, the measuring equipment must have 75μsec. de-emphasis in its path. The distortion output of the modulation monitor normally provides this de-emphasis. But, if it is switchable, make sure that it is switched in.

In order to make the measurements, after the level at a particular tone has been set for response measurements and recorded, calibrate the distortion analyzer to 100 per cent. Then null out the fundamental tone and measure the remaining distortion according to the particular instrument. One precaution here—always switch the distortion analyzer metering circuit to a less sensitive position before switching the audio generator to a different tone.

NOISE

There are two noise parameters to be measured—f.m. noise, and a.m. noise. F.m. noise is the normal system noise that modulates the carrier in the same manner as does the audio programming. The noise can originate



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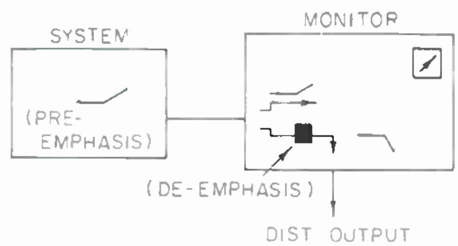


Figure 4. Noise and distortion measurements require 75 μsec. de-emphasis in the measurement equipment or circuit.

anywhere along the audio chain, as well as at the transmitter modulator itself. The voltage variable capacitor diodes often used today, as the reactive element of the modulator, are very sensitive to mechanical vibrations of blowers and similar devices. This can result in a high f.m. noise measurement. To make this noise measurement, modulate the carrier to 100 per cent with a 400 Hz audio tone. Calibrate the internal noise meter of the modulation monitor or the external noise meter. Remove the tone modulation to the system and terminate the system input with a resistor. Then, simply measure the noise according to the instructions of the particular instrument. The minimum tolerance which must be met here is —60 dB below 100 per cent modulation of the carrier. As with distortion measurements, these noise measurements must also incorporate the 75 μsec. de-emphasis.

The other noise parameter is a.m. noise. This is noise which amplitude modulates the carrier. Most generally this comes from power supplies in the transmitter, or from power tubes themselves. Present day monitors have the means within them to measure a.m. noise, but older models do not. If the means are provided, there is ordinarily a switch position marked carrier level or similar terminology. It requires setting the rf input level to the monitor accurately as a reference. Then proceed to make the measurement as specified for the particular monitor. If no means are provided in the monitor, then an external diode rectifier, an audio generator and noise meter are necessary. Space does not permit going into those older methods of measuring a.m. noise. The minimum tolerance for this type noise is —50 dB below 100 per cent amplitude modulation of the carrier.

EQUIPMENT NEEDED

If meaningful measurements are to be made, test equipment of suitable accuracy is needed, as well as suitable procedures used, to insure the accuracy of the measurements. The most basic item is a properly calibrated modulation monitor. The monitor in question is a mono monitor or the base-band monitor of a stereo monitor arrangement. This measures the total modulation of the carrier, and provides the audio output for distortion measurements which incorporates the proper 75 μsec. de-emphasis. Most modern day monitors also provide the means within them to accurately make the f.m. and a.m. noise measurements.

An audio signal generator, which itself has low noise and distortion, is required to supply the audio input test signals to the system. The amplitude of these input signals must be accurately measured either by an integral voltmeter/pad arrangement, or by an external audio voltmeter. These measurements are plotted as the system response.

A distortion analyzer which has a wide-band input is necessary to measure the harmonic distortion of the system. These are generally the null type and also provide an oscilloscope output for viewing the distortion that is being measured.

OTHER MEASUREMENTS

While the proof itself only requires the minimum in tolerances and measurements which must be made, there is nothing to prevent you from making any additional measurements you desire. For example, you may run response measurements at additional audio frequencies to provide a more accurate picture of the response curve and also make it easier to draw the curve. And if your station operates by direct power measurement, this would also be an appropriate time to accurately calibrate the power output meter and check the efficiency of the transmitter power tubes. Lastly, you may desire to measure the carrier frequency with a frequency counter.

WIND UP

Restore all the equipment for normal programming by enabling all the processors that were disabled for the tests, removing patch cords used for the tests, and so forth. Make sure the place works before you leave. In a day or so, after you have recovered from the loss of sleep, draw up the proof into a presentable package. Draw the appropriate graphs for response and distortion, use block diagrams and describe the procedures used for making the measurements, as well as other pertinent information. Sign and date the proof and keep it at the station for two years so that an FCC Inspector may look it over if he desires. ■

NORMAN S. CROWHURST

db Theory & Practice

About Signal and Noise

• In our art—or is it a science—we have a lot to say about dynamic range, signal-to-noise ratio, noise reduction and so forth. And a lot has been written about these subjects, but still a lot of people seem confused about at least parts of it. So let us start at the beginning and try to straighten out some of the confusion.

Has anyone ever told you that, by putting your ear to a sea shell, you can hear the sound of the sea? If you take one of the larger, helical sea shells and put it over your ear, you certainly can hear a sort of rushing sound, similar to that made by the waves as they beat on the shore. And people with a superstitious turn of mind believe that a sea shell has some mystical, permanent connection with the sea, whereby these sounds are brought to you in some telephonic fashion.

If you have a more scientific turn of mind, you reason that such sound is not conveyed by means of wires, nor does the sea shell contain some tiny radio receiver. What the sea shell does, is to behave as a broad-band acoustic resonator—rather like the cochlear of the human ear—to emphasize, or amplify low-level sounds already present, but inaudible—below the threshold of audibility—so you can hear them.

If you cup your hands over your ears, about the same distance away as you hold the sea shell to get that effect, you get a similar effect, but much less. The sea shell is a much more efficient resonator than your hand. Notice that the sea shell has the same form as the cochlear, but while an empty sea shell contains only air, the cochlear, much smaller, contains the fluid that transmits sound waves to the hair-cell sensors inside it.

You may ask yourself why the sea shell does not amplify all the sounds you hear, including those you hear without putting the shell over your ear. Actually it does. But you do not notice that, because of the way your hearing responds to changes in loudness. Suppose the sea shell acts like a broad-band amplifier that gives 10 dB emphasis to frequencies within its band. Near the threshold of audibility, that can make previously inaudible sounds quite definitely audible. But sounds that were previously audible will be increased in intensity only by 10 dB, which is not very much, when compared to hearing something you could not hear at all before. But try the sea shell in an anechoic room, or somewhere where there is complete silence. You will find it does not "work." This proves that the sea shell relies on residual, ambient sound that is normally below your threshold of audibility.

Whether you realize it or not, if you cannot hear sound, you subconsciously believe that no sound is present. Amplifying sound that was below your threshold of audibility, gives the illusion of producing sound out of nowhere; and such sounds are normally around us all the time. People in our profession call that "noise."

When a performer gets too far from a microphone, and the sound man turns up the gain, the microphone picks up more and more background sound. If there is no specific, recognizable form of background, such as voices chattering, the resultant sound will take the form of noise. If there are not sufficient acoustic vibrations in the air to produce such noise, the increased amplification will "find" electrical random "events" which it reproduces as a very similar form of noise.

NOISE VERSUS SIGNAL

This brings us to what distinguishes noise, which may be defined as sound we do not want to hear, from "signal" or sound we do want to hear. We use the word signal, to cover a variety of "wanted" sounds. It may be human voice, talking for us to listen to, which rather obviously contains "intelligence." Or it may be music, whether or not others would agree with what we call it.

What is the difference between any of the forms of signal, and noise? Actually, there can be a variety of differences, which is what can make this subject confusing. It also gives rise to different methods of improving signal-to-noise ratio, or dynamic range. But our ears are not always aware of these distinctions, so we are apt to jump to some false conclusions.

Early in our audio lessons, we learn that audible sound consists of acoustic vibrations at frequencies within a range, that varies from one individual to another, but in most instances is somewhere between 20 Hertz and 20

BRH90 RADIAL HORNS

This is the BRH90, a ninety degree radial horn for two inch and one and three-eighth inch compression drivers. Its an easy name to remember —Big Radial Horn, 90 degrees— very simple. But there's nothing simple about our design or construction.

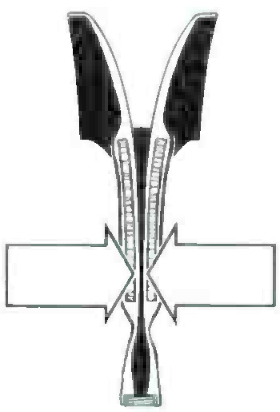
Community horns are made of fiberglass. Not sprayed-up cheaply-made fiberglass, but fiberglass hand-laminated by experts and constructed to our exact requirements for maximum acoustical accuracy, resonance-free rigidity and unparalleled strength. Our horns are absolutely weatherproof. They will *never* corrode or rust out. Nor are they ever likely to break. Not now and not forty years from now.

But the real mark of Community radial isn't apparent unless you cut it in half as we have done here. If you look closely at the cutaway, you will see that the corridor in the horn just past the throat gets extremely narrow before it flares out. That pinch in the horn is absolutely necessary to any mathmatically and acoustically correct radial horn, whether it is metal or fiberglass, and only Community does it the way it should be done. And not only do we do it right, we do it cheaper.

We also make two 90° radials for use with one inch and screw-on drivers —the RH90 and the SRH90. Both have the same flare rate, but the SRH90 (Small Radial Horn) has a smaller mouth, and is usually used as the high end in three way systems.

Talk to your Community distributor. Even though he also sells our competitor's products, he'll probably tell you that there is nothing available today that equals a Community horn.

He's right.



	BRH90	RH90	SRH90
Flare Rate	240Hz	345Hz	345Hz
Operating Range	500Hz up	600Hz up	1,000Hz up
Size: H.	11 1/8"	12¼"	6½"
W.	33⅝"	30⅜"	24¾"
D.	21"	20½"	18¾"
Weight	25 LB.	20 LB.	12 LB.
Finish	Black, High Gloss	"	"

	BRH90 KHz	BRH90 -3dB	BRH90 -6dB	RH90 KHz	RH90 -3dB	RH90 -6dB	SRH90 KHz	SRH90 -3dB	SRH90 -6dB
Horizontal Dispersion	6	85	95	6	80	90	1.2	95	100
	2	90	90	2	90	100	3	90	95
	10	80	90	10	85	90	10	85	100
Vertical Dispersion	6	50	90	6	55	100	1.2	50	70
	2	35	50	2	35	50	3	40	65
	10	20	35	10	20	35	10	20	30

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Kilohertz—a range of a thousand to one, though most of us are receptive to somewhat less than that range. Both signal and noise consist of a mixture of such frequencies, otherwise we could not hear them at all.

EARLY RESEARCH

A lot of research has gone into this subject. Dr. Harry Olson, of RCA Labs, was one of the first in the field, way back when. He built the granddaddy of all synthesizers, that occupied at least one whole room (according to where you considered the synthesizer to leave off—remember they had no solid state then, only tubes). And some of us still remember his early demonstrations which, like the early demonstrations with the phonograph, were remarkable, but a long way short of what synthesizers in everyday use can achieve today.

We mention that, because Harry pioneered a lot of the original theory work, analyzing how sounds are made up. Musical sounds, at any rate, are made up of frequencies, usually a fundamental and overtones, or harmonics, for each note identified. Then each note can have variation in the way it grows, is sustained and decays, over time.

But these pioneers in synthesis found that, even with banks and banks of oscillators, and wave-shapers, and envelope shapers (to control growth, sustain and decay) a lot was left to be desired, as regards "copying" everyday sounds, like the piano, violin, guitar, etc. And voice?! Even without trying voice, percussive sounds proved difficult, in those days.

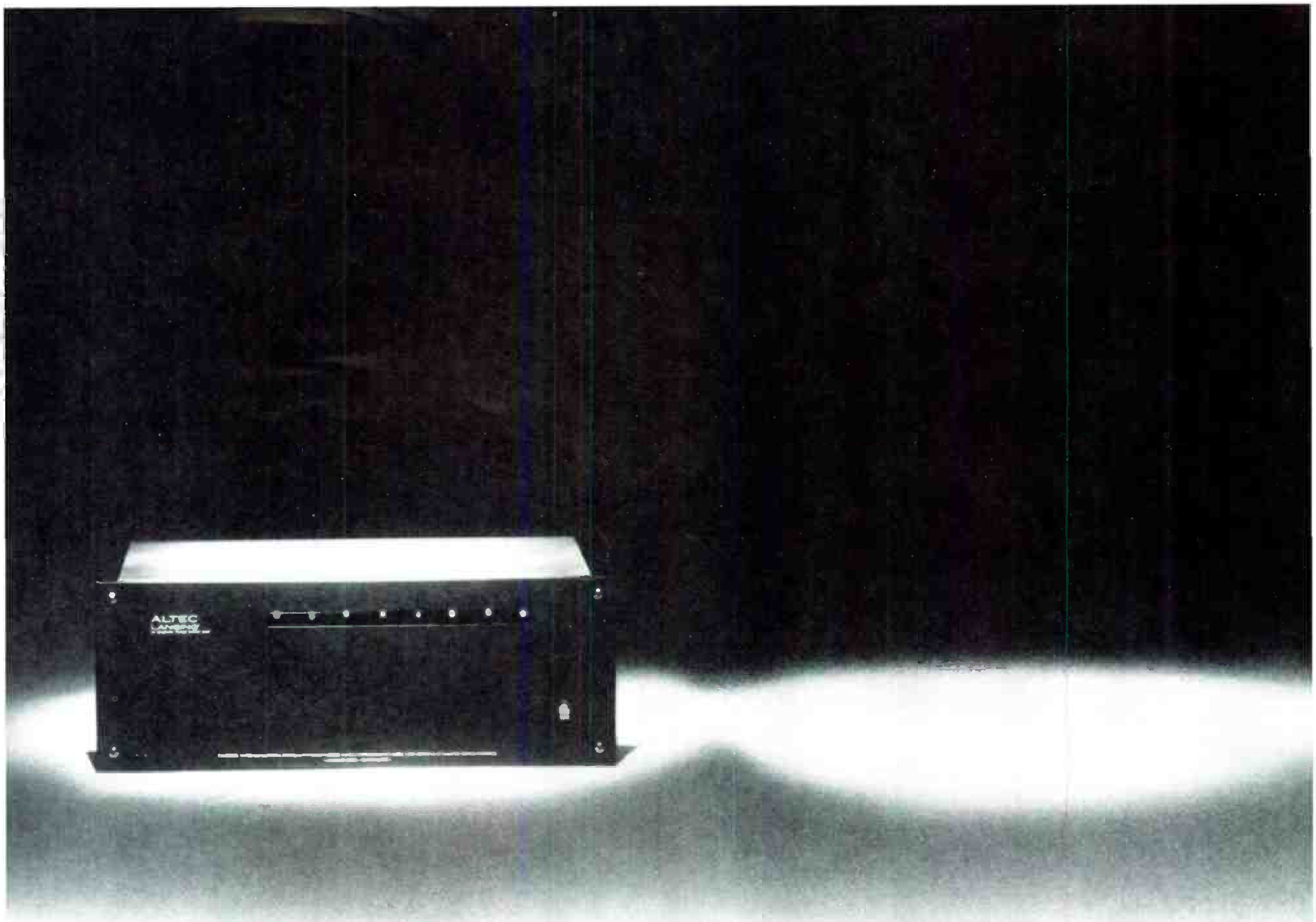
RANDOM NOISE

Multivibrators and waveform simulators, as opposed to frequency synthesis devices, introduced a lot more capability. But many sounds can best be duplicated by using random noise as a starting point. What is random noise? A roomful of people clapping, is one start toward it. The sound of rain, pounding on pavement is another. In what way do these differ? They both consist of "random" events —a lot of people clapping independently of one another, or a lot of rain drops hitting in unscheduled sequence —but they may also have "coloration."

In the case of handclapping, the coloration derives from the way individuals cup their hands. Rain drops may produce coloration dependent on the nature of the surface on which they fall, such as a tin roof. In music, a brush applied to a drum produces a similar, but somewhat more musical effect, as the wires strike the drum "randomly" each one stimulating a response from the drum, which has its own resonance.

So simulating percussion and some other instruments, can best be done by using a "noise generator" as a source, or starting point. This produces a sound just like the input stage of a high gain amplifier, consisting of random events in the form of moving electrons, to which various filtering is applied, to selectively emphasize certain frequency components in the noise. Now we are getting a better picture of the complexity of sound sources that produce the kind of sounds we want to hear.

So much for wanted sounds, which inevitably include components of a kind that theorists in the field call "noise," although that word is used in a quite different sense from the common one, meaning "unwanted sound." If wanted sounds were all periodic, meaning synthesized from however many frequencies, of sine waveform, that it takes to make them up, then one way to separate signal from noise would be to develop a complicated circuit that would re-

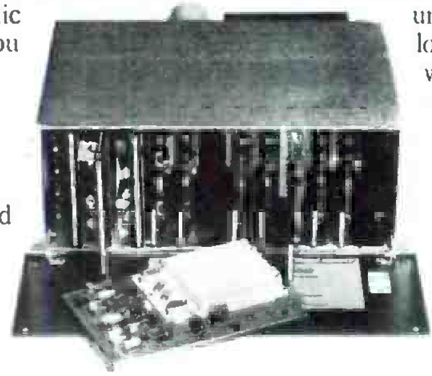


Introducing Altec Lansing's Incremental Power System. And Its Closest Competitor.

Lately a lot of the big names in professional amplification have been making head-to-head comparisons with their competition. And, understandably, the brand being featured in each ad usually comes out on top. But one product that no one is comparing themselves with is Altec Lansing's new Incremental Power System.

That's not really surprising since Incremental Power is a lot more than just an amplifier. Each main frame actually contains a flexible array of power amps, electronic crossovers, line amps and input devices. So you get a complete amplification system that's prewired and ready to use. And since it is a system, Incremental Power offers a degree of flexibility that's unmatched by any single amplifier. In fact, to match the overall performance of one Incremental Power System you'd need a rack full of traditional components.

Skeptical? To prove the point we've devised a head-to-head comparison that you can make for yourself. Below you'll find the published specifications for an Incremental Power System set up for stereo, triamplified operation. Simply select the competitive components that you'd need to match Incremental Power's performance and then judge for yourself.



There's a lot more to Incremental Power than we have room to tell you here. So if this kind of performance and package size sounds good to you, contact our Commercial Sound Sales Department for the details. Or check the Yellow Pages under Sound Systems for the name of your local Altec Lansing sound contractor. Either way you'll get the complete Incremental Power story. We think you'll agree that our short story makes the competition look a long way behind.

Altec Lansing Sound Products Division, 1515 South Manchester Avenue, Anaheim, California 92803.

ALTEC LANSING

Altec Corporation

	Power Available for L.F. @ Mfg. Rec. Load	Power Available for M.F. @ Mfg. Rec. Load	Power Available for H.F. @ Mfg. Rec. Load	Electronic X-over	Cooling	Weight	Height	Reliability
Incremental Power System	300 Watt Total 150 Watt/Ch. @ 8 ohm	150 Watt Total 75 Watt/Ch. @ 16 ohm	150 Watt Total 75 Watt/Ch. @ 16 ohm	2 or 3-way Selectable Freq.	Built-in fan blows side-to-side	70 lbs.	7"	Excellent each unit factory tested

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spond only to periodic sounds, not to noise.

AN EXPERIENCE TO REMEMBER

This theory reminds me of someone who brought us an invention—it must be more than 40 years ago now—in the realm of loudspeakers. In those days, most loudspeaker designers were aware of resonances. Not only the main resonance, but a whole sequence of them—a dozen, a hundred, who knows—distributed throughout the audio spectrum. And most designers felt the ideal was to get rid of as many of them as possible.

Some years before that, makers of loudspeakers (most of whom operated in their basement or garage) advertised their product as having "Cathedral (or some other) tone." But the fact that cathedral tone did not sound good on a jazz combo led designers to realize that the loudspeaker should add no tone of its own. But this man had the idea, since resonances are so hard to get rid of, that it might be better to make more of them, and control them the way you want.

So he put his loudspeaker unit in a box like the wind chest of an organ—the old pipe organ. And on this chest he placed enough pipes of varying length, diameter and "stuffing," to resemble a small pipe organ. He brought it to us to evaluate, in the hope we might help him put it on the market. According to him, it would be the Rolls Royce of speakers, a real super-duper.

He had a record player and a recording of an organ with which to demonstrate it. And on the organ record, it sounded terrific: more like a real organ than we had ever heard before. Fortunately for us, much of our business at that time was with installations to amplify human voice. So we insisted on trying it with a recording of human voice. The only one we could lay hands on, was of a comedian of the time, who was always good for a laugh—even if you had heard it before.

We put this on, and it wasn't even funny. A funny man talking inside an organ chest, is no longer funny, we concluded. His punch lines had lost their punch. No, this was not the way to make a loudspeaker, which disappointed our potential inventor no end. But it was an experience one doesn't forget.

NOISE BY TODAY'S STANDARDS

On today's reproduced sound, we hardly know what noise is, compared to those days. You knew when the needle touched the groove of the phonograph record, by that hiss coming from the loudspeaker. Similarly, you knew whether the radio was on in the same way, without it being tuned to a station. Everything had noise problems you wouldn't believe, by today's standards. If you could hear what you wanted to hear, at all, in the middle of the noise, you marveled at modern science. You couldn't expect to hear every word, or every instrument in the orchestra, now could you?

Nowadays, we talk about signal-to-noise ratios of 50 dB, 60 dB, and on up. In those days, we were struggling with signal-to-noise ratios where the dB figures would be negative—the noise was louder than the signal; you had to listen to the signal, down in the noise. So we have come a long way. But how? Well, we seem to have used up this month's column, just getting this far. So next month we will continue, by getting into some of the various things that have been done toward making reproduction perfect, in the sense of having no noise as a "giveaway" to the fact that it is reproduced sound. ■

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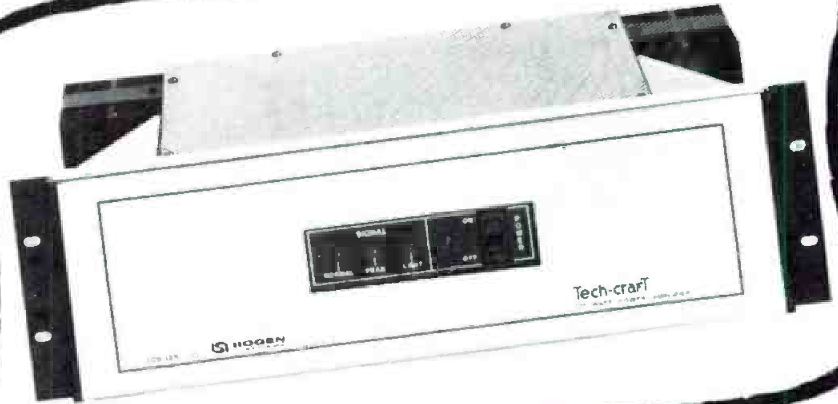

For the Critical Link:

Power Amplifiers You Can Count On

EVEN IN CONTINUOUS USE, OVERLOADED, AND AT 131°F

No sound system can be any better than its critical link, the power amplifier. Whatever inputs, mixers, and speakers you use, if the power amp lets you down, the message won't go through, the show won't go on. We've designed Bogen's Tech-crafT Professional Power Amplifiers to assure quality performance even under the most adverse conditions.

UNEQUALED PROTECTION

These amplifiers can be operated 24 hours a day at rated power, into gross overloads, at ambient temperatures up to 131°F--and continue to perform superbly. They won't be damaged even if the speaker lines are shorted or the temperature goes still higher. In other words, you can't burn them out, even when overloaded. Computer-grade electrolytics are used exclusively.

PURE POWER

The TCB-60, TCB-125, and TCB-250, rated at 60, 125, and 250 watts respectively, deliver their rated power at less than 1% total harmonic distortion, 25 to 22,000 Hz. At 20% higher power, THD is still under 2%, 30 to 18,000 Hz. Frequency response is within ±1 dB, 20 to 20,000 Hz, at rated power. Output regulation, no load to full load, is better than 1 dB. We think, and believe you'll agree when you try them, that these are the best power amplifiers you can buy for sound reinforcement.

ADDITIONAL ADVANTAGES

LED indicators show whether the amplifier is operating in normal, peak, or limiting mode. You can use these three amplifiers virtually anywhere in the world, on 120 or 240 volts, 50/60 Hz. They require less rack panel height than others and their physical balance makes them easier to handle. In short, they reflect all the experience of our nearly 50 years in amplifier design and manufacture.

14 Tech-crafT PRODUCTS

Our Tech-crafT series includes, in addition, two dual-channel power amps, two mixers and two matching mixer-extenders, two graphic equalizers, a compressor/line amplifier, and two mixer-power amplifiers. All 14 products were designed by the same team, for highest performance and utmost reliability.

This is a franchised line sold through qualified professional sound installers. For more information, please write or phone us.

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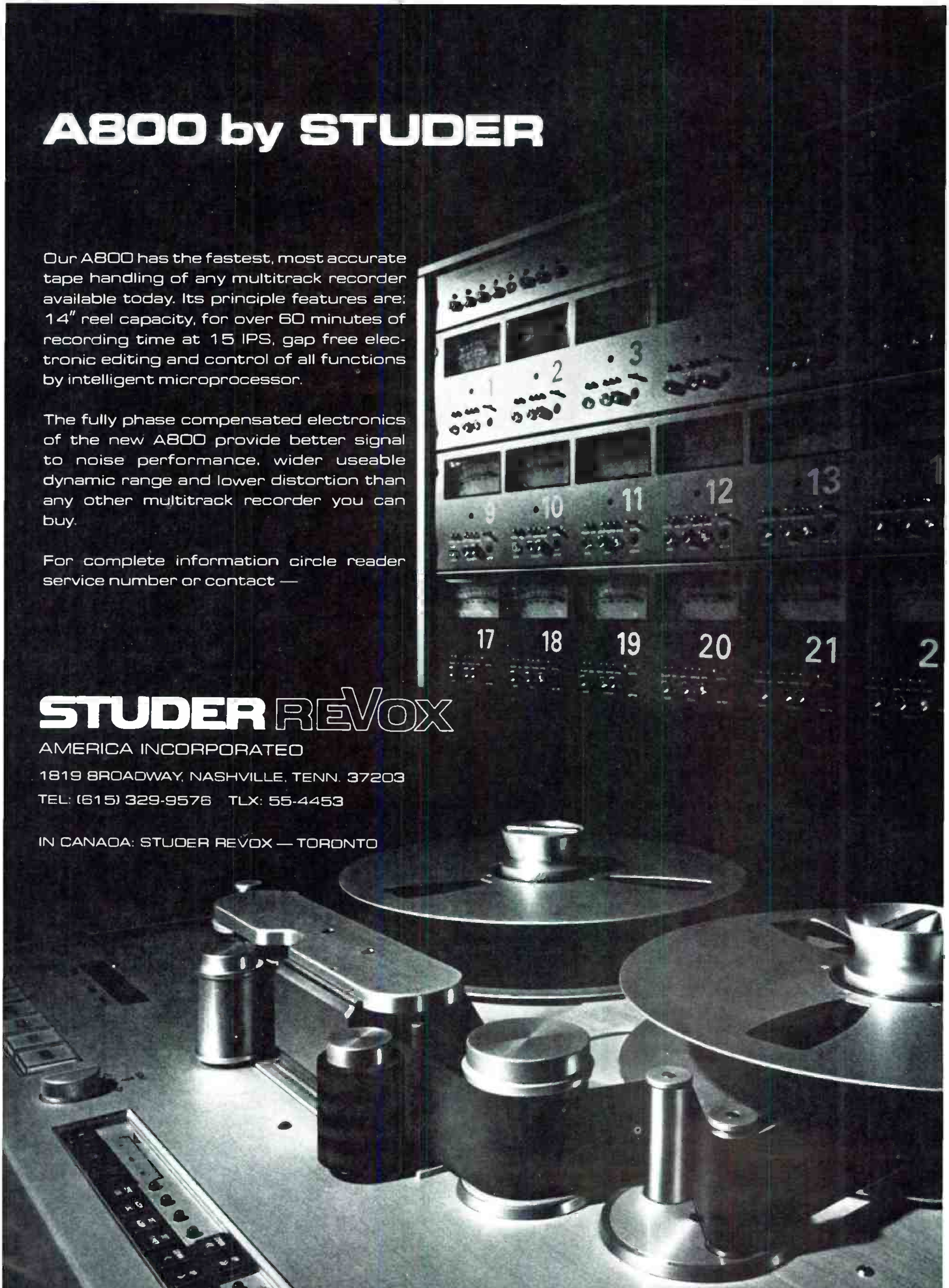
Seeing is...

● Last month this corner discussed the eye and how it operates; comparing it to a camera, in some ways. (Incidentally, as of the end of last year, a new camera is being developed which will not have any film in it. Instead, there will be a memory chip, similar to those found in computers, which will retain the image the camera sights on, and display the image on a screen. The signal can then be fed to any photo copy machine for making prints or transparencies. Sort of like an open eye of a robot—with a computer brain and a chip retina.) This time we'll take a look (if you'll pardon the pun) at what the eye sees under different circumstances.

Remember one of the earliest experiments in an early physics class when a pencil was put into a glass of water and the pencil seemed bent? The explanation was that light rays were bent when they passed through different media such as air and water. A simple optical illusion. Another is the simple reflection in a mirror where the eye seems to see the object behind the mirror at the same distance from it as the real object is. Unless the brain accounts for what the eye seems to see, the observer can be fooled.



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MIRAGES

Mirages are another eye-fooler. And there seem to be two types of these. One occurs in the hot climate of the desert. Here, heat waves, rising from the ground, cause the eye to "see" an image that is not really there. An oasis, perhaps. A similar "image" can be seen while driving on a perfectly dry road, when, suddenly, a "wet" spot appears ahead.

The other kind of mirage occurs in the Arctic regions. During a temperature inversion, where there is a mass of cold air trapped under a layer of warmer air, an image can appear above the horizon. This, however, is an image of an actual, physically-existing object such as a mountain, island, or city. The difference of temperature bends light rays from an object which may be beyond the horizon, and not visible directly, to an apparent position above the horizon. The eye is "fooled" into thinking the object is really there. It actually is, but not where the eye "sees" it.

An interesting article in the New York *Times*, a couple of months ago, discussed the possibility that the Norse voyagers "discovered" America by following Arctic images. A letter to the paper the following week said that there couldn't have been these Arctic images, since, at the time of the voyages, the Arctic was warmer than it is now—and it was "doubtful" that these mirages existed around 1,000 AD. Believe what you may, but the article did illustrate the difference between the two types of mirages with a prism effect. In the desert mirages, the prism has the apex down, causing an "image" to appear at or near the ground. In the Arctic mirage, the prism has the apex up, causing the "image" to appear in the air.

OPTICAL ILLUSIONS

Other ways to fool the eye would be with optical illusions. Everyone knows the test which consists of two parallel lines (either horizontal or vertical), one with arrows at both ends, the other with outward pointing "arrow" lines. The question is which line is longer. Of course, both are the same length, but the eye "sees" something which the brain interprets, and concludes incorrectly, that the one with the arrow lines pointing outward is longer.

A similar optical illusion occurs when two lines of the same length are drawn between receding railroad tracks. The illusion that the tracks are getting closer, the farther away they get, makes the farther line look longer, because it seems to be closer to the tracks at both ends. (An article in another New York *Times* edition discussed the theory of "size constancy" —which the brain adds to account for what the eye seems to see.)

It is believed that the same "mechanism" is at work when the moon, rising at the horizon, appears to be larger than it does when it is directly overhead. It is believed by some that it has to do with atmosphere, and by others that the moon is closer at that point. Both are wrong.

Those who agree with the theory of "size constancy" believe that it is this learned function of the eye/brain combination that explains the difference in moon size, larger at the horizon (where it is actually farther away) and smaller overhead (where it really is closer). At any rate, these examples of how the eye can be "fooled" brings us to the next subject.

USING SLIDES WISELY

Perhaps it is not generally known that the slide business is exceptionally large. In fact, one estimate puts the total figure at over a billion dollars for hardware, software, and all associated services. That's larger than either film or video for presentations. It's obvious, therefore, that the goal of the producer of the presentations should be to get the most out of the slides being used. After all, we do learn about 85 per cent of what we know through sight, and we retain about 30 per cent of what we see and about 50 per cent of what we see and hear.

The next question is: What can be done to make what we see more interesting, exciting, attention grabbing, while still providing the viewer with easy legibility and greater retentivity? It has been found that *color*, used properly in the making of slides, can offer certain advantages. The eye responds more to color than black-and-white, and the mind interprets color to mean different things. Red can mean fire, heat, danger, passion. Green can be used to mean freshness, while blue can be interpreted as coolness. But the combination of colors, especially in slide work, can either help or hinder how easily the audience can see, read, and interpret the information on the slide. The colors selected can also be made to be more effective depending on the make-up of the viewers. For example, blue is usually considered male-oriented, while pink would appeal more to women. Bright colors like red and yellow are more appealing to children or youth. But in questions of legibility of slides, proper contrast is still the essential element.

COLOR COMBINATIONS

A simple guide might be the color wheel we saw in early art classes. The primary colors, red, blue, and yellow are separated by the secondary colors violet, green, and orange. (The secondary colors, incidentally, are made up of the combination of the primary colors on either side of them.) The sharpest contrast is, of course, white or bright yellow on black, and such slides will prove impressive and be attention-grabbers, but continued viewing of this combination will cause the eyes to tire easily. Other contrasts that may also be impressive and deserve consideration are opposite colors on the wheel like red and green, or violet and yellow, or blue and orange. Such combinations will also prove interesting; and if treated properly, can make effective bar charts, for example. Using a dissolve system for two or more projection units, and the proper color choices for the build-up slides, will also provide greater interest, more awareness, and higher retentivity.

The proper use of colors can also "fool" the eye. For example, using a blue background for an object will make the object appear to be farther away from the background, or closer to the observer, than, say, the same object in front of an orange background. Perhaps, this can also be explained by knowing how the eye works. Normally, blue focuses in front of the retina. In order to see that color clearly, the eye lens becomes thinner and flatter, thus causing the color to appear farther away. With red, the opposite happens, since the color comes to a focus behind the retina and the lens becomes broader or more convex to put this color in sharp focus—bringing the color closer. (Come to think of it, if that color is brought closer, it would also cause the object to look larger. Is this what would explain why the moon, orange at the horizon, looks closer and bigger, than it does overhead? Probably not, but it's an interesting thought.)

What we set out to do was to provide some interesting ideas on what makes the eye "see" things differently. ■

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CONDENSER MICROPHONE

• Capable of withstanding extensive use and abuse, the model SM81 condenser microphone possesses a precise, cardioid polar pattern resulting in exceptional off-axis rejection of unwanted sound for improved separation and isolation. The SM81 features a switchable 10 dB attenuator built into the head of the microphone, and a three-position low-frequency response switch providing the option of a flat response, a low-frequency roll-off of 6 dB-per-octave below 100 Hz, or a low-frequency cut-off of 18 dB-per-octave below 80 Hz. The SM81 exhibits low total harmonic and intermodulation distortion below its clipping point, and operates over a wide range of simplex (phantom) powering voltages and impedances.
Mfr: Shure Brothers Incorporated
Price: $225.00
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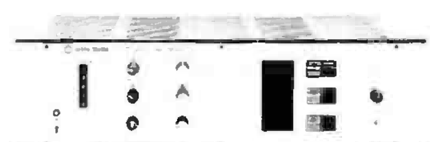
DIGITAL DELAY



• Employing a sampling frequency of 50 kHz, the DN70 digital time processor offers delay options of 163 ms, 326 ms, and 652 ms with a frequency response of 30 Hz to 15 kHz at maximum delay. The unit utilizes one input and four output channels—all standard XLR 3-pin audio connectors. Front panel controls for regeneration and direct/delayed pan are used in conjunction with the A, B and C output levels to provide a composite signal at the mixed output. Additional front panel facilities include an input level control, an led headroom indicator for constant input level monitoring, and a digital readout in milliseconds of the set time delay of each output.
Mfr: Klark-Teknik
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SHOCK-MOUNTED MICROPHONE

● Primarily intended for hand-held broadcast and sound reinforcement applications, the DO56 shock-mounted omnidirectional microphone is constructed, in such a way, as to isolate the main acoustic cavity and the diaphragm/voice coil assembly from the microphone casing. Frequency response extends to 18 kHz—with a slight emphasis in the 2 to 12 kHz range, enhancing vocal qualities. In addition, a slow roll-off below 200 Hz reduces low-frequency noise interference. "P-popping" protection is achieved via a high-density Acoustifoam blast filter.
Mfr: Electro-Voice
Price: $100.00
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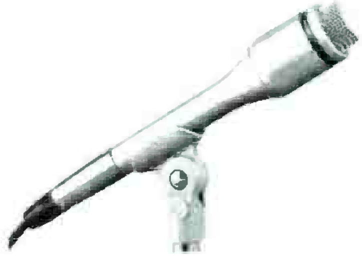

MICROPHONE SNAKES

● Available in 100 ft. lengths, each Stage snake is constructed of shielded wire, balanced low impedance mike lines. The Stage snake is fitted with Switchcraft plugs, an easy to read metal stage box and cable strain relief clamps. 6 through 24 mike lines are available.
Mfr: Musimatic
Price: $293.00 to $639.00
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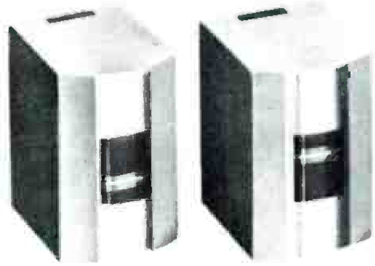
RECORD HEADS



● Direct replacements for all major types of duplicating heads, the SRF-4C series hot pressed/glass bonded ferrite record heads are designed and manufactured to insure long term operation. Mechanical construction of these heads features glass bonded, hot pressed ferrite cores enclosed in an extremely hard ceramic material designed specifically for use in magnetic heads. Two versions of this new ferrite record head are currently available: an 8-track, 2-channel head for stereo duplicating; and a 4-track head for cassette duplicating.
Mfr: Grandy, Inc.
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db Editorial

PSYCHO-ACOUSTICS—*The study of the psychological response of human beings to acoustic stimuli.* (International Dictionaries of Science and Technology—Halsted Press.) The definition has an intimidating ring to it. Obviously, psycho-acoustics is something best left in the hands of scientists, while the rest of us get on with the business of making a living in the real world of pro audio.

But then again, maybe not. Perhaps we could all benefit by a little exposure to how people respond to acoustic stimuli. If not a thorough immersion in the complexities of the ear-to-brain signal path, then at least a nodding acquaintance with the fact that "loudspeaker output" is not necessarily the end of the line.

Despite our various chores within the audio profession, we are all concerned with the creation, production and packaging of acoustic stimuli. If we do a good job of it, the listener will respond by buying our records, listening to our radio station, and attending our concerts. As our technology advances, we find ourselves with more and more tools to help us produce that "ultimate stimulus." That being so, then why aren't today's stimuli that much more ultimate than yesterday's?

Could it be we are overlooking (overhearing?) something, as we breathlessly await the arrival of that new console with 40 more inputs on it? Have we forgotten that even the most golden-earned listener is still—despite technology—just a two-input device?

Before you think, "two ears," think again. It's really "ear and eye" (hopefully, a pair of each). But since we are all so involved with the former, we tend to overlook the latter. Yet most psycho-acousticians spend a lot of time fretting about the significance of visual cues: they know so well that the eye has a profound influence on the ear. Actually, they are both influencing the brain, but for most of us audio mavens, the visual stimuli are rarely given much thought.

But we should remember that, with the visual stimulus missing (as in radio and recording), the audio stimulus has to take up the slack. So, it would seem that we should really be doubly-concerned with the audio portion of psycho-acoustics, if we are to eventually take full advantage of the technology now available to us. Even the proliferation of VTRs, and the re-emergence of large-screen television, doesn't ease the burden on audio very much: we're still a long way from surrounding ourselves in video, while doing so in audio is—or could be—relatively easy, given a little more understanding of psycho-acoustics.

And so, this month we give psycho-acoustics its due. Some cynical observers suspect that at times the subject is more psycho than acoustic. True enough—there's no shortage of quacks out there, ready to expand the brain while compressing the budget, with the latest patent-medicine magic box. The unwary had best tread lightly, and listen carefully before leaping. Happily, the lunatic fringe is still a minority, and there is no shortage of worthwhile contributions in this intriguing area of professional audio.

For example, consider that legendary laboratory which Almon Clegg conjurs up for us in **The Shape of Things to Come.** By now, an ever-growing number of observers (including your editor) will testify that ordinary stereo just doesn't hold up very well against the sonic wonders that are to be found there. Yes, they are very real. No, they are not available as production tools—not just yet, that is.

Binaural recording—once a rather limited-interest technique—is gathering more attention these days. Fred Geil describes some easily-repeatable **Experiments with Binaural Recording,** and brings us up-to-date on some of the technology that will bring the realism of binaural sound to loudspeaker reproduction. Again, its only a matter of time before some of this technology finds its way into the studio.

In the meantime, would you let an Aural Perception Heterodyne Exciter into *your* studio? Some people have, although it's not been too clear just what this mysterious black box was doing. Vladimir Nikanorov takes a look **Inside the Aphex** to give us some idea of what goes on in there. And just as his feature story arrived, so did a copy of United States Patent 4,150,253, which describes the Aphex as a "Signal Distortion Circuit and Method of Use." In other words, the Aphex is now patented. We've taken the liberty of reproducing most of the patent drawings to help the reader evaluate the merits of the system. According to the patent, it would appear that the Aphex is essentially a collection of fairly-standard signal processing devices, contained within one chassis. It would be interesting to discover whether a similar effect could be achieved with a hand-full of patch cords.

Also interesting is the fact that Aphex is not for sale. The user simply pays a fee, based on the playing time of the finished product. By contrast, the EXR Corp. is now advertising "The First Psychoacoustic Audio Processing System to be offered for sale." This is the company's model EX2 Exciter, which offers "psychoacoustic juxtapositioning" and "psychoacoustic replacement." Details on system operation are still scarce, but the company is young, and no doubt needs to strengthen its own patent position first. Hopefully, we can tell you more about it, when there's more to tell.

To conclude our foray into psycho-acoustics, Barry Hufker speculates on the significance of **Fibonacci Numbers, the "Golden Mean," and Audio Engineering.** It's been well-known for centuries that some geometric proportions seem more pleasing to the eye than others. Do the same proportions find application in music, or in audio engineering in general? Many acousticians think they do. In any case, it's food for thought. **J.M.W.**

ALMON H. CLEGG

The Shape of Things to Come: Psycho-acoustic Space Control Technology

A little background information on some audio science fiction—now well on its way to becoming a reality.

THE RAPID GROWTH OF ELECTRONICS has brought with it a proliferation of gadgetry for the manipulation and generation of audio signals. From the simple UJT siren to computer-controlled music synthesizers, there's a vast array of audio-generating and wave-shaping circuitry becoming available to the audio professional. In the recording studio, engineers and producers have been intrigued with electronic manipulation of signals, to produce certain desired (or undesired) effects. Phasers and flangers have already become almost-universal tools for the studio magician.

PSYCHO-ACOUSTIC CONTROLLERS

With the advent of computer-controlled circuit design, and the subsequent lowering of the barriers to the unknown, engineers are now exploring circuitry which can "trick" the ear. These "psycho-acoustic controllers" are analogous to trick photography and optical illusions. They can be used in one of two ways: to create a new sound image or some patterns heretofore unknown, or, to enhance the realism and presence of conventionally-recorded sound during the playback process.

The industry literature now reveals a number of commercially-available devices which claim an ability to enhance or improve the reproduction field. Some are oriented toward controlling the position or depth of the perceived playback signal, while others enlarge or spread out the sound stage. Of course, the effects are subjective, and it behooves the recording engineer to proceed cautiously, experiment carefully, and listen intently to this new generation of "black boxes."

A GLIMPSE INTO THE FUTURE

Let us explore the creative possibilities which lay ahead, as psycho-acoustic techniques are developed and refined, and perhaps made ready for eventual practical applications. For a moment, imagine that you have just received a special invitation to the legendary "Einstein/Fletcher Acoustics Research Laboratory." You have been forewarned that the experiment you will witness is "top-secret," and that the results may turn out to be one of the most spectacular psycho-acoustic accomplishments of the century. As you approach the lab, you are met by three acoustical engineers, who escort you down a long hallway, up two flights of stairs, through a heavy sound-proof outer door, and into an anechoic chamber. Suspended across the chamber is a piano-wire "floor," so that as you walk into the room, you are suspended between its six walls.

At the front of the chamber, you notice two small loudspeakers, with a collection of strange-looking electronics and a tape recorder off to one side. You are positioned at a point midway between the speakers, at the proper distance to form an equilateral triangle. A chair has been prepared for you to sit comfortably.

SOME LOCALIZATION TESTS

One of the engineers starts the tape recorder. A voice comes from the left-front speaker, with the precision of localization that can only be witnessed in a reflection-free environment. The voice says, "I am speaking to you from the left-front speaker." You close your eyes and mentally point to the location of the sound, which of course is precisely located at the left-front loudspeaker. The voice continues: "I am speaking to you from the right-front

Almon H. Clegg is assistant general manager, Product Planning and Engineering Dept., at Technics, Secaucus, New Jersey.

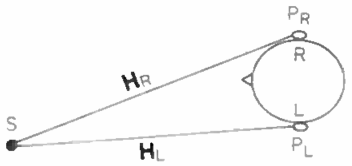


Figure 1. The paths from a sound source to the ear.

speaker." Again, the perceived position of the sound source is right on target. The voice goes on; "I am speaking to you from the left-rear speaker." And again, the localization of the voice is startlingly precise.

But, wait a minute! Didn't you remember seeing only *two* loudspeakers when you entered the room? You abruptly turn your head to the left to glance over your shoulder: there *is* no loudspeaker back there—just one of the engineers, with a sinister grin on his face. And now, the voice continues; "I am speaking to you from the right-rear speaker." Your head turns in the opposite direction—again to find nothing there. And then the voice says, "I am now speaking from a point directly over your head." This time you don't bother looking. By now, you realize there will be nothing there. As you think about what you have just heard, some music begins to play. It starts with a conventional stereo sound, and then suddenly "opens up" into a broad, deep spacious feeling, which completely encompasses you in the music. You become totally unaware of the two loudspeakers, for the sound is now coming from every direction, and you seem to be completely immersed in it. Instruments are firmly suspended at almost every conceivable position and direction.

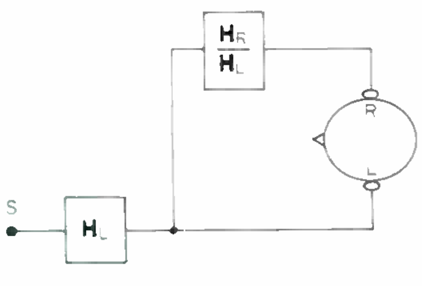
When the music stops, you can scarcely believe what you have just heard. You cling to the memory of this most-thrilling experience, while the white-frocked engineers try unsuccessfully to conceal their glee at your confusion, as they bid you a polite farewell.

FACT OR FANCY?

Although this little adventure into science fiction may seem to be based more on fantasy than fact, audio technology has already progressed to the point where such a laboratory experiment could actually take place. In fact, it is known that at least one acoustic research laboratory is conducting such experiments at this very moment.

Contemplating the physics behind such psycho-acoustic "witchcraft" is really not that complicated, at least in general terms. It is well known that a point-source of sound, from *any* arbitrary location, produces a sound wave which reaches each ear by some unique path. For example, consider a sound actually originating from the left-front. The left ear hears it directly, while the right ear hears it indirectly—modified in time (a fraction of a second later), and by frequency response (because of the masking effect of the head).

Figure 2. A circuit in which the transfer functions are electronically variable.



INTRODUCING TRANSFER FUNCTIONS

The diagram in FIGURE 1 pictorially represents the sound paths to the ear. For the purposes of our discussion, reflective paths are not considered. S is the sound source, P is the pressure at each ear, and H symbolizes the "transfer function." This is a complex mathematical equation describing the acoustic path between the sound source, S, and the point at which the sound pressure wave, P, "activates" the ear. The transfer functions contain the pertinent time and frequency information. The brain processes this data, and tells the listener the position (angular direction and distance) of the sound source.

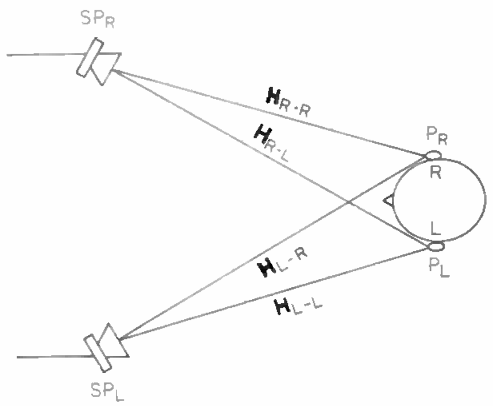
With the help of a bit of math, and just a little abstract thinking, it can be shown that the acoustic path of FIGURE 1 may be symbolically represented by the circuit of FIGURE 2. This circuit may not appear to be physically realizable, but it should help us understand how image localization is determined. By adjusting the ratio of the transfer functions, we can alter the ratio of the sound pressure common to both ears (H_L in our example), and between the ears. On paper, this allows us to manipulate the perceived location of the sound source.

To do this physically, let us consider a conventional two-speaker stereo system. In FIGURE 3, the various acoustic paths are shown. The transfer functions for the paths between the left speaker and the ears are: H_{L-L} and H_{L-R}. From the right speaker, the transfer functions are: H_{R-R} and H_{R-L}. Notice that if the listener is centered and equidistant from the left and right speakers—and if both the room acoustics and the shape of the head are symmetrical—we can assume that $H_{L-L} = H_{R-R}$, and $H_{L-R} = H_{R-L}$. Even if these transfer functions are not yet fully defined, this still appears to be a reasonable assumption, and is the basis of most stereo system designs. In actuality however, we would find that these transfer functions are usually dissimilar, and become more so as we discover that our lab-style assumptions about physical placement, room acoustics and head shape are usually not valid in the real world. Although physical placement of the listener may be easy enough, room acoustics are rarely symmetrical, and heads—never. But more about that some other time.

For the moment, in the general case the sound pressure level at each ear is:

$$P_L = SP_L \cdot H_{L-L} + SP_R \cdot H_{R-L}$$
$$P_R = SP_R \cdot H_{R-R} + SP_L \cdot H_{L-R}$$

Figure 3. The acoustic paths in a stereo system.



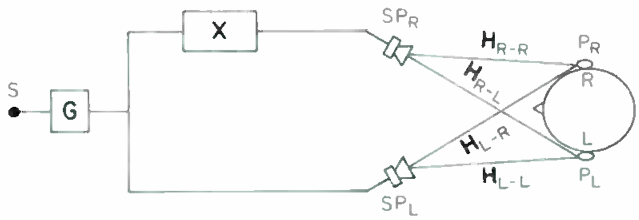


Figure 4. A model of a psycho-acoustic controller.

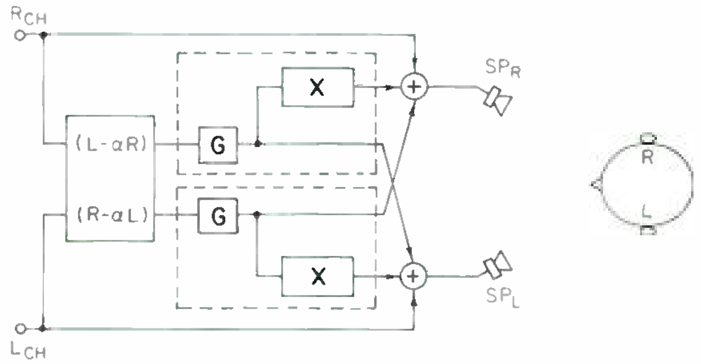


Figure 5. Block diagram of a prototype system for enhancing the stereo image.

Using an extension of the logic from FIGURE 2, a new model may be generated, and this is shown in FIGURE 4. Now, if we can design the circuits in **X** and **G** properly, it should be possible to faithfully re-create the original sound source location, regardless of where that was (front, rear, sides, above, etc.). Intuitively, we may realize that even a sound source located at say, left-rear-above, should still have just two sets of transfer functions.

BLACK-BOX DESIGN PARAMETERS

The task of the **X** and **G** "black boxes" must be to add a signal which exactly cancels the unwanted sound from the left-speaker-to-the-right-ear, and, from the right-speaker-to-the-left-ear. Then, the time and frequency response of each signal must be adjusted. When his has been done, the complex acoustic signals from each speaker will be properly tailored to achieve the desired end result—that is, to re-create the psycho-acoustic location of the original sound source(s). *When* all the conditions are symmetrical, the solution to **X** and **G** are as follows:

$$\mathbf{G} = \frac{\mathbf{H}_{L}}{\mathbf{H}_{L\text{-}L}}$$

$$\mathbf{X} = \frac{\mathbf{H}_{L\text{-}L} \cdot \mathbf{H}_{R} - \mathbf{H}_{L\text{-}R} \cdot \mathbf{H}_{L}}{\mathbf{H}_{L\text{-}L} \cdot \mathbf{H}_{L} - \mathbf{H}_{L\text{-}R} \cdot \mathbf{H}_{R}}$$

Of course, we have still not adequately defined any of the variables within these equations. In fact, the various transfer functions in **X** and **G** are quite complex, and their mathematical—and then, electronic—realization is well beyond the scope of this article. However, with today's level of audio technology, they are indeed capable of being modelled by analog and digital circuits, which have the capability of generating the time-delay and frequency response modifications dictated by the equations.

PROTOTYPE CONTROLLERS

Prototype circuits that have already been constructed show astonishing accuracy and precision of image position control. Keep in mind, however, that for the moment, this is a laboratory series of experiments, which must be carried out under carefully-controlled conditions. For example, the position of the listener's head cannot move more than two or three inches if the experiment is to have maximum effect. (Also keep in mind that stereo itself was once just a laboratory experiment—Ed.)

PRACTICAL APPLICATIONS

To bring this developing technology a little closer to practical reality, the system shown in FIGURE 5 represents a psycho-acoustic controller that will greatly enhance any conventional stereo system. Through experimentation, it has been found that this two-channel psycho-acoustic controller can produce a sound stage of about 180 degrees, under most "real-world" listening conditions. Listeners have described it by saying that, while conventional stereo places the apparent sound images somewhere between the left and right loudspeakers, this system makes the sound "jump out," and become free of the speakers. It presents a homogeneous wall of sound, forming a semicircle about the listener. Most listeners have felt strong side images, well beyond the placement of the loudspeakers themselves Although listening position is certainly an important factor, it is no more critical than with conventional stereo. In fact, the enhancement effect is usually quite obvious, even from an adjacent room.

FURTHER EXPANSION OF THE SOUND FIELD

Another interesting prototype was constructed by taking two stereo psycho-acoustic controllers and using them in a discrete quad format. The controller handling the rear channels was modified to take into account the fact that the rear side of the head is shaped differently and, the distance along the surface of the head between the two ears is also different. The result is a quadraphonic system which can clearly reproduce images from *any* location, without the need for anechoic conditions.

A well-known, and quite apparent, weakness of all quad systems (even discrete four-track tapes) is an inability to produce unambiguous side images. However, the quad controller just mentioned produces an astonishing accuracy in side-image localization, as well as the illusive center-rear position.

CATCH-22

Psycho-acoustic control technology depends, for its success, on the creative manipulation of time-delay (phase) circuits, as well as other parameters. It is but one more area in audio where a better understanding of the implications of phase integrity is required. If this technology is eventually to become a practical studio production tool, the phase integrity of the recording and transmission chain will have to be carefully maintained.

However, given a recording and broadcast system that is able to utilize psycho-acoustic localization controllers, the recording engineer and producer should be able to apply these techniques on a selective basis—perhaps during the mixdown session—and this will give even more creative and artistic freedom in the positioning of instruments and vocalists.

Editor's note—For a more-rigorous technical presentation of the technology discussed in this article, the reader is referred to a technical paper that was presented at the 60th convention of the Audio Engineering Society. The paper, entitled "On the Advanced Stereophonic Reproducing System 'Ambience Stereo'" is available as AES preprint No. 1361 (G-3). ■

FRED G. GEIL

Experiments with Binaural Recording

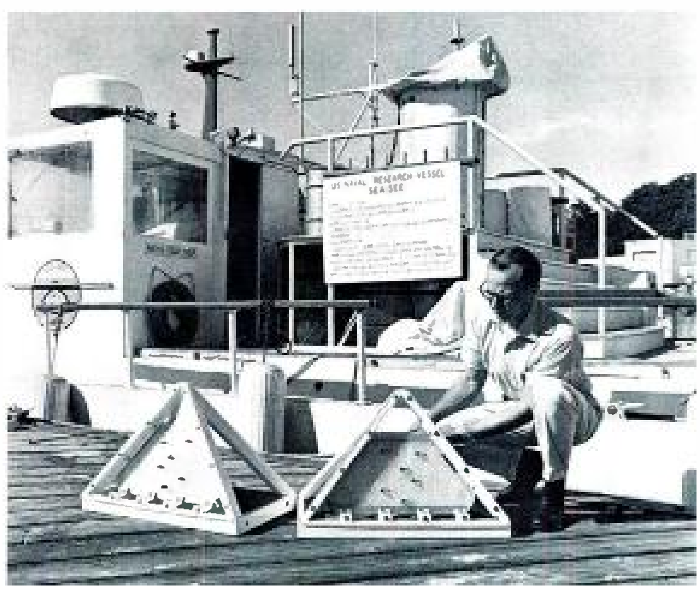
A first-hand account of some binaural recording techniques, and an attempt at binaural reproduction over loudspeakers.

DURING THE EARLY 1970's, the US Navy conducted a series of experiments in underwater localization of sound sources. At the Naval Ocean Systems Center in San Diego, the Navy sought means to improve the ability of a sonar operator to accurately pinpoint the direction of sounds in the ocean.

An early experiment incorporated two large brass "ears," spaced about four feet apart, and steerable by the operator. The brass ears achieved an impedance mismatch to the water that was similar to the mismatch between our own ears and the surrounding air. The large size and spacing of the ears was necessary, due to the speed of sound in water, which is about five times faster than in air.

Although certainly unwieldy, these brass ears worked fairly well, and helped advance the state of binaural knowledge, at least for underwater applications. The experiments also added to our understanding of the role of the outer ears, or pinnae, which produce a delay and time smear effect on signals reaching the eardrums. These effects were incorporated into an experiment which used a fixed pair of air-backed, nonsteerable, hydrophone arrays. The operator sat in a submerged large glass sphere, recording a diver who made noises while circling around the sphere. Not exactly your conventional recording session, although the resultant data underscored the significance of head motion in accurate localization.

Ron Betsworth of the Naval Ocean Systems Center displays two hydrophone arrays which were scaled and located geographically to simulate two human ears. The arrays were then mounted on an acrylic plastic observation chamber ten feet below the surface on the NOSC research vessel Sea-See. The Artificial Pinna Program sought to determine if there is any advantage in directionally detecting signals underwater, as contrasted to normal sonar signals.



Fred G. Geil is supervisor of Sonic Technology, Oceanic Division, Westinghouse Electric Corp., Annapolis, Maryland

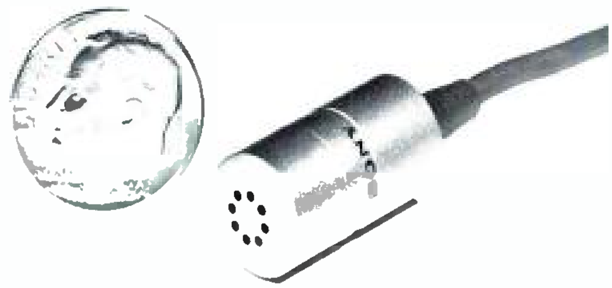


Figure 1. Sony tie-clip microphone used in the dummy head.

The Navy's research—as well as work by other experimenters—has shown that the important factors in accurate binaural localization are;

1. Visual cues.
2. Head motion.
3. Head masking.
4. The shape of the pinnae.
5. Time differences between the ears.

VISUAL CUES AND HEAD MOTION

In any experimental recording project, the significance of visual cues and head motion cannot be over-emphasized. Despite the potential effectiveness of the recording system, during subsequent playback the visual-cue factor is of course missing. And although the listener is presumably able to turn his head from side-to-side, there's nothing there to see (except a loudspeaker). That places the entire localization burden on head masking, pinnae shapes and time differences.

HEAD MASKING EFFECT

I recently steered my seventh-grader, Marty, towards a science fair project to investigate sound localization factors; specifically the importance of the head-masking effect, and the contribution of the outer ear shape. He installed Sony omni-directional tie-clip microphones (FIGURE 1) at the ear positions of a surplus (and ear-less) dummy head. He did not recess the microphones to ear-drum position, because we did not want two ear-canal resonances in the acoustic path of each ear signal.

A recent Audio Engineering Society paper[1] points out that the listener's outer ear configuration produces a resonance peak in eardrum pressure at about 2.7 kHz. Therefore, if the microphones were mounted at the ear-drum location, we would have, in effect, a double resonance: once, as the sound was recorded—and again, as we listen to the playback. The AES paper describes the electronic compensation needed to counteract this effect.

SOME LOCALIZATION EXPERIMENTS

Marty put open-air type headphones on his volunteer audience (seated in another room) while he walked around the microphone pair with a specially-procured signal generator which produced a noise source rich in harmonic content (my electric razor). At first, using simply-spaced microphones (no dummy head), he observed that localization was quite poor, since only time differences were transmitted to the listeners.

Using the head, localization was better, since amplitude differences were now provided as well. However, there was confusion between front-quadrant and rear-quadrant sounds. Listeners often referred one-o'clock sounds to the five-o'clock position.

For the next step in the experiment, add-on ears fashioned out of clay helped reduce the confusion somewhat, although forward-sounds were still not convincingly up front.

ADDING HEAD MOTION

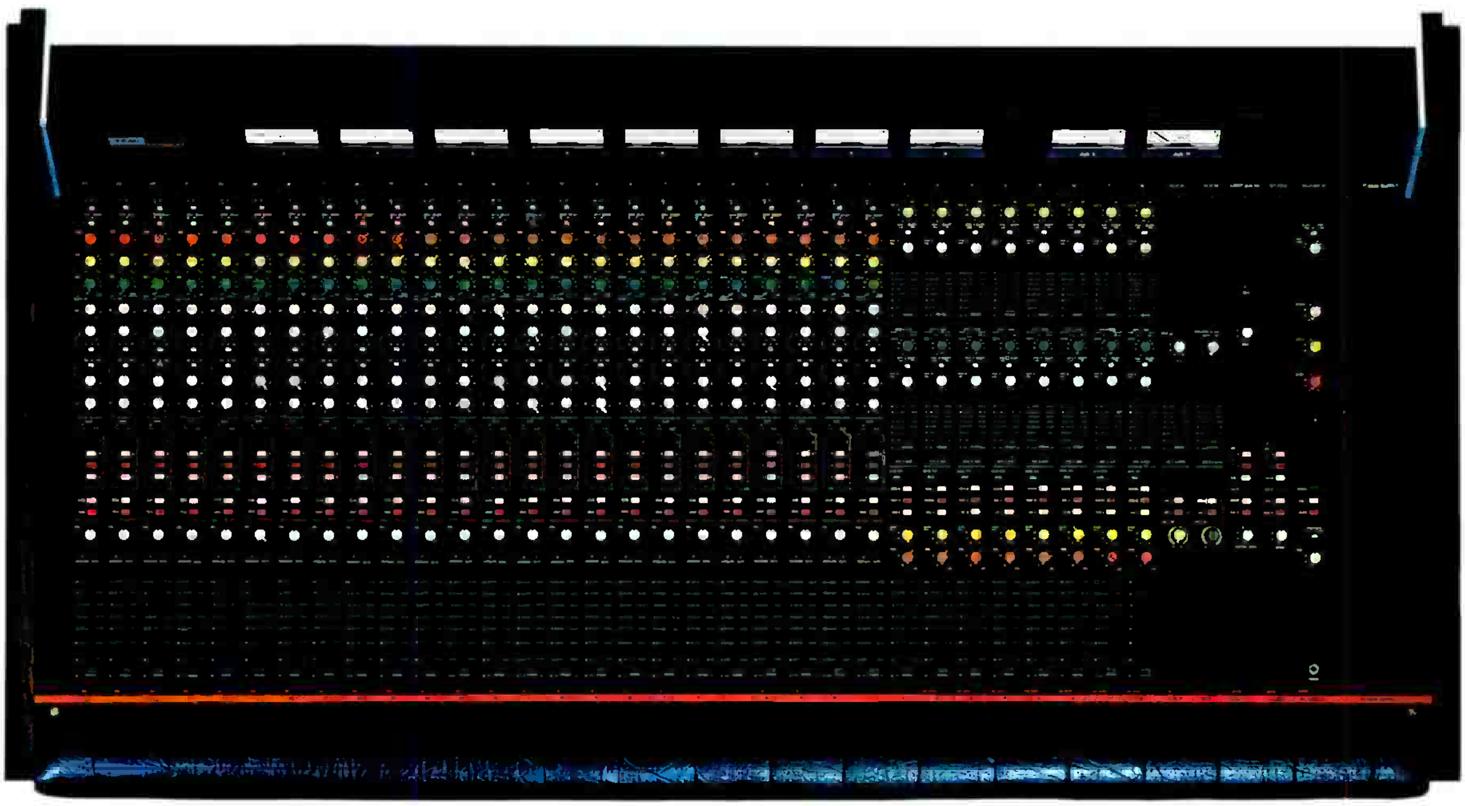
Frontal localization was enhanced with a little head motion. Here, the dummy head was aimed at a nearby television set. By rotating the dummy head from side-to-side, while the listener did the same, a definitely up-front sound was produced. However, while moving the dummy head oppositely to the listener's own head motion, the sound source appeared to originate in the rear.

In order to improve the fixed-head forward localization, we decided to experiment with more realistic-looking ears. We were fortunate enough to obtain a pair of rubber ears made by Industrial Research Products Corp., and installed them in our dummy head. These ears are cast from a soft, tear-resistant RTV silicone rubber, which provides a good simulation of pinna flexure. This produced a more-realistic overall sound, although our fixed-head, fixed-source localization tests were still only marginally better than our primitive clay ears.

These ears are a part of the KEMAR (Knowles Electronics Manikin for Acoustic Research) system, which has been described elsewhere.[2] The KEMAR ears, installed in our dummy head, are seen in FIGURE 2. The complete KEMAR system (not used in our experiments) is shown in FIGURE 3. A cut-away view of yet another dummy head system is seen in FIGURE 4. Here, Schoeps omni-directional capsules are mounted within each ear of a system designed by research engineers at the Victor Company of Japan (JVC). (The re-assembled head may be seen on this month's cover—Ed.) *(continued)*

Figure 2. The KEMAR ears installed in the dummy head.

YOU'VE GOT IT WIRED WITH A MODEL 15.



Buying a big mixer can be very deceiving. From the time of delivery to the moment your board is operational, you can run into quite a few additional costs and frustrating time delays.

But consider the Model 15. Rear panel patch points are already wired. Included in the cost. The meter bridge is already wired. Included in the cost. The separate power supply plugs right in. Also included in the cost. It's not unusual to get your board in the morning and do your first session that same night.

With the Model 15, you've got performance and flexibility wired, too. From the discrete microphone preamplifier, equivalent input noise is −126dB (weighted). With one input assigned to one output buss, signal-to-noise is 76dB (weighted).

Formats are 16- or 24-channel input/8-buss output. Fully modular.

The Model 15 will drive any 16-track recorder and give you a vast array of mixing, monitoring and cueing capabilities. For example, the Cue mixing position can be fed by 48 sources simultaneously (all the inputs plus all 16 tape playback positions plus all eight echo receives).

Out of the crate, you'll have a lot more mixer in the Model 15 than you can get elsewhere for the money. Add your savings on installation (both parts and labor), and the Model 15 becomes even more cost-effective.

So think about the real, often hidden costs of buying a mixer. When you add it all up, we think you'll see the practical advantages of getting it wired with a Model 15.

The Model 15's functions, interior layout and complete specifications are described in our 10-page Product Information Bulletin. See your Tascam Series dealer or write us for a free copy.

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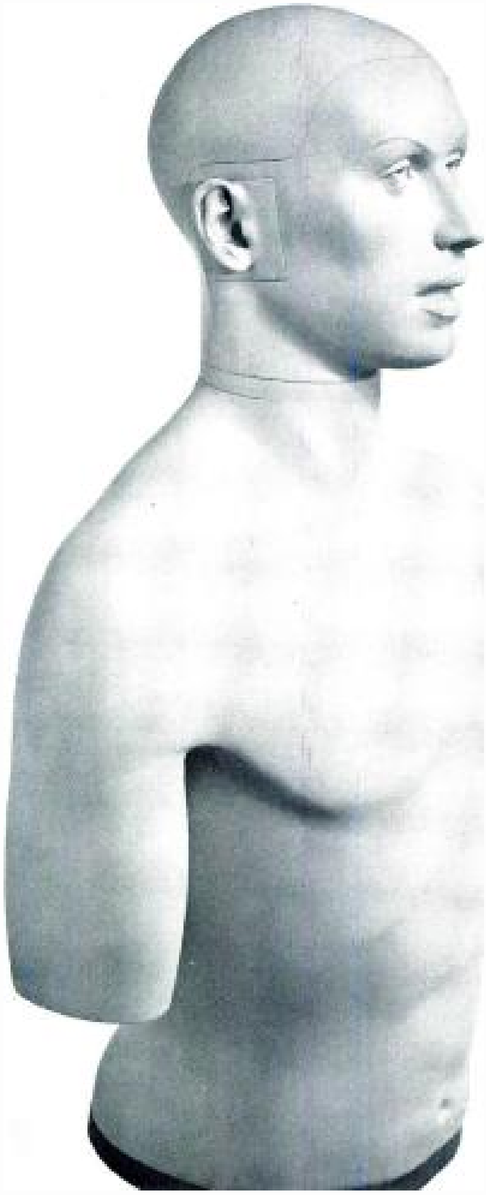


Figure 3. The complete KEMAR manikin provides life-like test conditions, primarily in research and development of hearing aid devices.

OUR DUMMY HEAD AT THE CONCERT HALL

Unfortunately, binaural sound recorded in the concert hall can't take advantage of either head motion or visual cues, and we have to rely on the time difference, head masking, and pinnae factors. Under these constraints, how successfully can we localize sounds? To find out, I took the dummy head with its brand new ears to Maryland Hall in Annapolis, to record a rehearsal of the Verdi Requiem, complete with symphony orchestra, chorus, soloists, and trumpets in the balcony. The ensemble was the Annapolis Symphony and Annapolis Chorale, under the direction of Leon Fleischer.

The dummy head was hung from the ceiling, and the experiment produced two immediate results. First—the bad news: there was considerable comment, and none of it was complimentary. To make a long story short, our dummy head would not be invited to attend the actual concert performance. However, the good news was a very realistic rehearsal playback over headphones, with the balcony trumpets sounding definitely to the rear. Frontal sounds, including comments by conductor Leon Fleischer, were fairly-well localized. In fact, the localization was better than I have experienced with headphone playback from my normal (head-less) microphone array.

LOUDSPEAKERS AND BIPHONIC PROCESSING

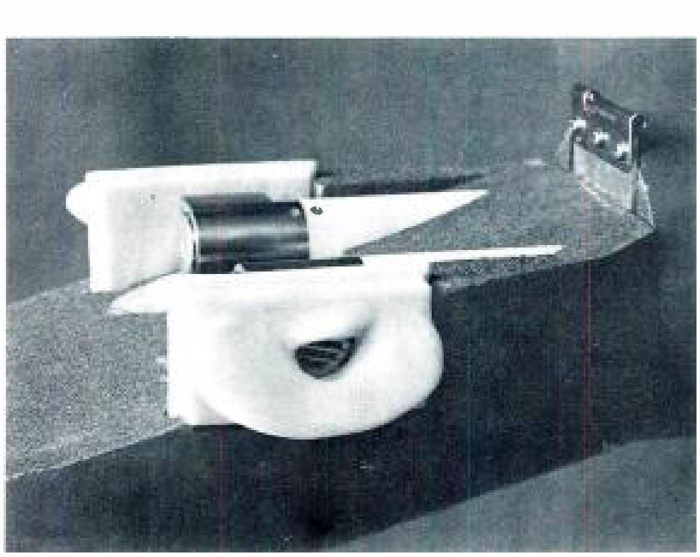
Research work by Damaske[3] investigated the concept of creating 360-degree binaural localization using only front loudspeakers. It was shown that, under the proper test conditions, localization may be nearly ideal for the entire horizontal plane, *including* front-to-back discrimination. Manufacturers are now introducing practical realizations of these concepts in commercial equipment. I was fortunate enough to borrow a JVC BN-5 Biphonic Processor for our continuing experiments. In this system, the two loudspeaker signals are modified to cancel the acoustic cross-talk that occurs in the listening area between the left speaker and the right ear, as well as between the right speaker and the left ear. (These concepts are further described in Almon Clegg's feature, "The Shape of Things to Come" in this month's issue—Ed.)

I played the dummy head recording through the Biphonic Processor into a typical living room music system, with speakers spaced about thirteen feet apart, at ear level. With the Processor switched on, the sound seemed to spread out beyond the speakers, and have some depth to it. The balcony trumpets were not clearly in the rear—rather, they were somewhat difficult to place. Localization and placement of the forward instruments and voices was good. Switching off the Processor caused the performance to shrink back to the wall between the two speakers.

I have played this dummy-head recording through the Biphonic Processor for others, getting mixed reactions. Some prefer it, some don't, and others don't notice much difference at all. I have found its effect to be well-preserved through the re-recording process, even with a cassette recorder. This seems to indicate that phase shifts introduced equally into each channel do not negate this psycho-acoustic process.

JVC engineers are taking the biphonic process one step further, with R & D proceeding on a Q-biphonic recording technique. The process permits un-ambiguous localization of sounds anywhere within the listening area. Sound sources may seem to originate overhead, or within inches of the listener, despite the fact that the speakers are actually positioned some distance away. A Q-biphonic dummy head is seen in FIGURE 5. Eventually, the Q-biphonic technology will be available as an electronic signal-processing

Figure 4. Cut-away view of a dummy head, showing omni-directional microphones within each ear.



device, permitting its application to tapes that were recorded with conventional microphones, rather than dummy heads.

ORTF MICROPHONE SYSTEM

As mentioned above, the Verdi Requiem concert performance was not to be recorded via the dummy head. However, I was able to record the performance with my regular coincident-pair set-up, using two AKG D 200E cardioid microphones, arranged as shown in FIGURE 6. The microphone capsules are spaced 17 cm apart, at an angle of 110 degrees. This configuration is frequently used by the French Broadcasting System (*Office de Radiodiffusion—Television Francaise*), and has come to be known as the ORTF system. About ten years ago, Carl Ceoen conducted a series of listening tests, in which several miking techniques were compared.[4] These included: X-Y (crossed cardioids), "stereosonic" (coincident Figure-8's), ORTF, M-S, and five pan-potted microphones. Listeners were asked to judge each system for liveness, perspective, dynamic range and other subjective qualities. Mr. Ceoen noted that the ORTF system was most-often selected as the best overall system.

Figure 6. The author's ORTF microphone assembly.

Figure 5. A prototype Q-biphonic dummy head, developed by JVC research engineers to investigate the quadraphonic sound field.

Confirming Mr. Ceoen's findings, this has been a very satisfactory microphone set-up, providing good localization and ambience, when positioned about ten feet behind and above the podium. It also provides an acceptable mono signal.

ORTF AND THE BIPHONIC PROCESSING

My next experiment was to play the recorded ORTF performance through the Biphonic Processor, comparing it to the recorded dummy-head rehearsal. It seemed that the 17 cm spacing was close enough to the dummy head's ear spacing, and the cardioid patterns might help to approximate the head-masking effect.

The results were surprisingly like the dummy head results, although I have had as many different reactions from others as I did in the previous experiments. My own preference is for playback of ORTF recordings through the Biphonic Processor, although some listeners prefer the conventional sound.

FUTURE EXPERIMENTS

Another experiment remains: to apply the Biphonic process to the Annapolis Symphony's next f.m.-stereo broadcast. Here, the processor will be used prior to transmission, with comments solicited from the radio audience. ■

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VLADIMIR NIKANOROV

Inside the Aphex

An attempt to sort out the fact and fantasy surrounding the mysterious Aural Perception Heterodyne EXciter.

SOME TWENTY YEARS AGO, Curt Knoppel constructed the first prototype of what is now known as the Aphex System. At the time, he thought he was building a Heath-kit preamplifier. And so he was, complete with a few of those inevitable mis-wired connections that go into most kits.

While trouble-shooting the finished kit, Knoppel became intrigued with the characteristics of the output signal, which could best be described as "weird." It turns out that —due to the wiring problems—an audio signal applied to the input of the preamplifier followed *two* routes to the output, and one of these passed the signal through a phono cartridge (a Shure M 75) that was plugged into the phono input.

This improbable signal path certainly needed some corrective maintenance, for the resultant output exhibited a very poor low frequency response, along with a phase shift across the 200 Hz—20 kHz bandwidth. For good measure, the output signal also contained a high level of harmonic distortion.

Before getting around to making the necessary circuit corrections, Knoppel began experimenting with this "processed" audio output, and eventually tried mixing it, in varying proportions, with a direct signal. Thus, Aphex was born.

Vladimir Nikanorov is technical director at Bonneville Broadcast Consultants, Tenafly, New Jersey. Mr. Kikanorov was formerly a recording engineer with Melodiya Records, Moscow, USSR.

ENHANCED AUDIO

Knoppel discovered that the Aphex signal seemed to function as an audio "enhancer," when mixed with the direct signal in proper proportion. Since there didn't seem to be any definitive "scientific" explanation for what was going on (or why), the early days of Aphex found the system shrouded in mystery, with an aura of "hocus-pocus" surrounding it.

Early "information" from the company didn't do much to promote confidence. One brochure stated, "The new Aphex sound is generated in the brain itself, as it would

Figure 1. A block diagram of one embodiment of the invention.

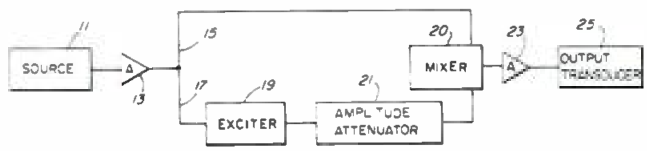
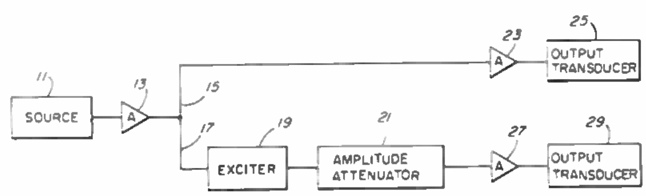


Figure 2. A block diagram of another embodiment of the invention.



U.S. Patent Apr. 17, 1979 4,150,253

The Aural Perception Heterodyne Exciter (Aphex).

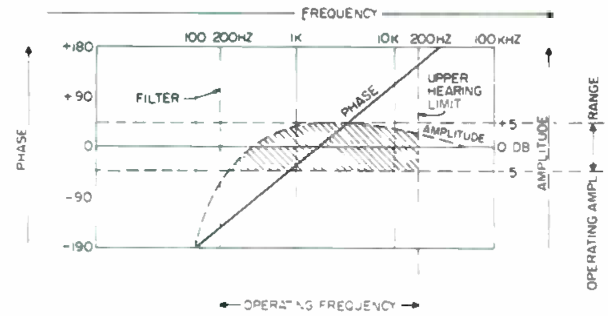
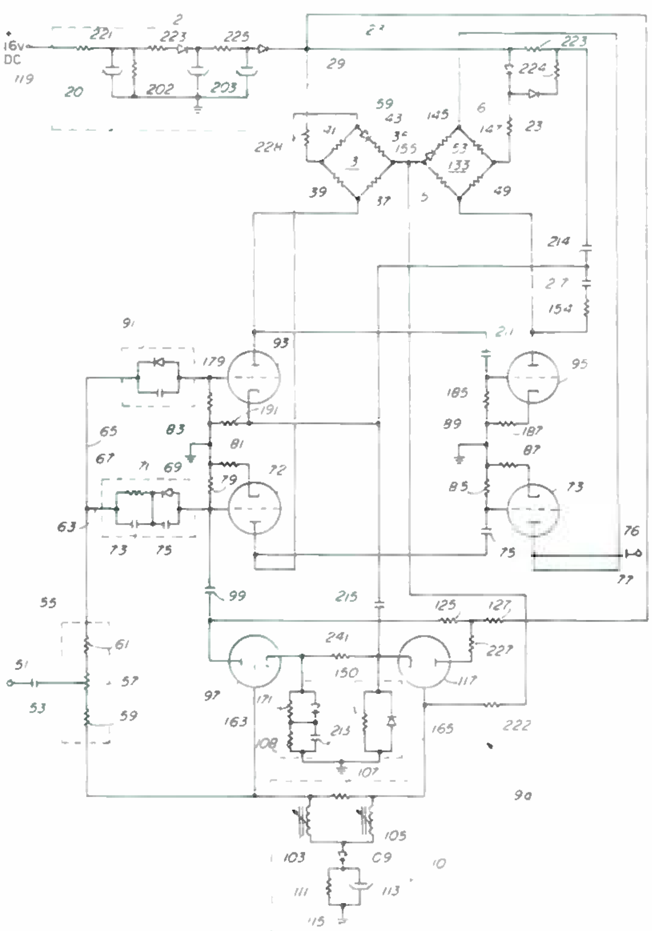


Figure 4. A graph illustrating the frequency dependent phase shifting characteristics of the exciter circuit in Figure 3.

have been originally. Natural sonic awareness is restored . . . Aphex encoding permits the brain to discriminate and recreate the original details . . . The brain-generated sound, always a part of audio perception, can now be fully developed and strengthened with Aphex." And so on.

Of course, it's always difficult—if not impossible—to accurately describe any sort of "sound effect." If you don't believe that, just try to explain flanging to someone who has never heard of it. Even now, a complete evaluation of the Aphex—or almost any other signal processing device—really requires a listening session, and the best that can be done here is to take a closer look at the electronics behind the sound, in the hopes of reducing at least some of that feeling of being present at a magic show every time the system is described.

Figure 3. A circuit diagram of a vacuum tube version of the exciter circuit shown in Figures 1 and 2.



U.S. Patent Apr. 17, 1979 4,150,253

SONIC AWARENESS PATENTED

A patent was recently issued under the title, **Signal Distortion Circuit and Method of Use** (US Patent No. 4,150,253—April 17, 1979). In an attempt to better explain Aphex, the patent illustrations and excerpts from the accompanying text will be presented here. Verbatim excerpts from the patent text are reproduced here in italics. All numbers are cross-references between the text and the illustrations. Figures 1 through 8 only are from the patent.

In FIGURE 1, *the signal travelling along path 17 is passed in succession through an exiter circuit 19, an amplitude attenuator 21 and is then combined in a mixer 20 with the signal travelling along path 15. The combined signal is passed through an amplifier 23 . . .*

In FIGURE 2, the direct and side-chain signals are *not* mixed together, but rather, feed separate transducers—for example, loudspeakers in P.A. applications. In this case, the mixing would take place acoustically.

VACUUM TUBE CIRCUIT

FIGURE 3 shows *a schematic diagram of a vacuum tube version of the exciter circuit 19* (of FIGURES 1 and 2). It seems to bear a striking resemblance to a mis-wired preamplifier, and the *inductive circuit 101 whose purpose is to establish the phase slope* (at the bottom of the figure) may be a component-replacement of that early Shure stereo cartridge.

FIGURE 4 is *a graph illustrating the frequency dependent phase shifting characteristics of the exciter circuit in Figure 3.* There seem to be a few minor errors in this figure that may be worth noting. Presumably, the phase plot begins at −180 degrees and not −190, as shown on the left-hand vertical scale. Also, the right-hand 200 Hz should read 20 kHz. The operating frequency range designated by dashed lines should be shifted slightly to the right, so that the 200 Hz line intersects the amplitude response curve at the −5 dB point.

As for that straight-line phase response, that's a puzzler. Nowadays, there are all sorts of all-pass networks and time delay systems around, capable of producing almost any phase response that may be needed. However, the patent says that *the two signals applied to the grid of tube 72 are completely out of phase* (i.e., 180°) *with each other.* But, as noted above, *connected to the grid of triode 97 is an inductive circuit 101 whose purpose is to establish the phase slope.* The text then lists the components, and notes that the circuit *is connected to the grid of triode 117.*

This inductive circuit *produces a linear frequency dependent phase shift of about 360 degrees over a frequency range from about 100 Hz to about 22 kHz,* with a zero

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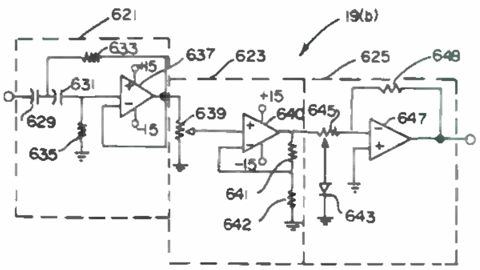


Figure 6. A circuit diagram of a solid state version of the exciter circuit shown in Figures 1 and 2.

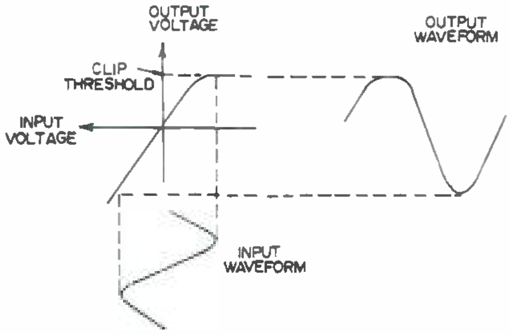


Figure 7. A plot of the transfer function of the harmonic creator in Figure 6.

phase shift at about 2 kHz (or at about 1.5 kHz, according to the graph in FIGURE 4). Presumably, this response appears at the output of triode 117, and therefore, also at the grid of tube 72. But, if it is a frequency-dependent phase shift, how can it always be completely out-of-phase with the signal arriving via path 63?

The test acknowledges that *it cannot be said with absolute certainty which specific elements in exciter circuit 19 perform which functions.* The circuit description concludes by stating that the output signal contains *low order, odd and even, phase shifted and amplitude dependent harmonics.* Probably, what is meant is that the output of triode 97 shows a phase shift of 180 degrees with respect to the signal from path 63, while the output of triode 117 is the frequency-dependent phase shift.

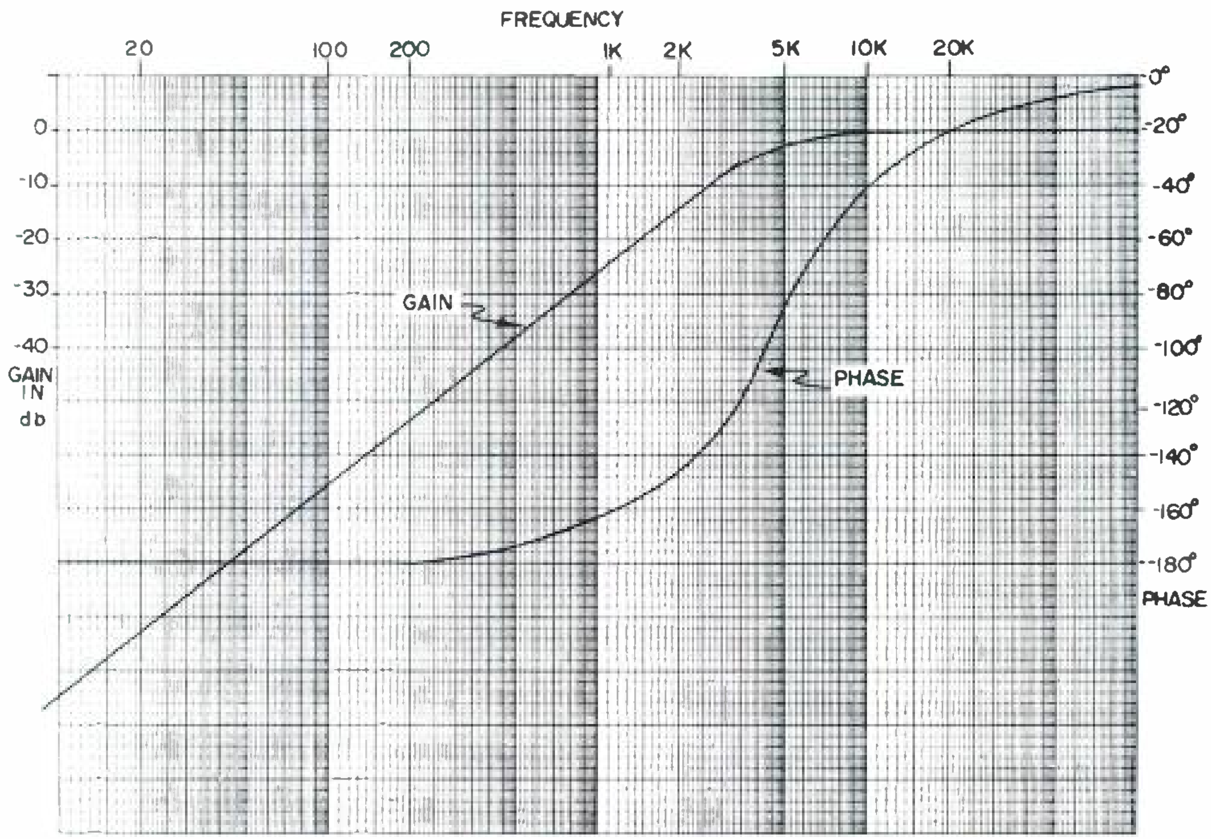
SOLID-STATE VERSION

The solid-state version of the circuit (FIGURE 6) is a lot easier to handle. Here, the circuit comprises three sections; a high-pass filter 621, a variable-gain amplifier 623, and a "harmonic creator" 625.

THE HIGH-PASS FILTER

The filter is a conventional two-pole Butterworth filter (12 dB/octave) with a roll-off frequency at 4.5 kHz, as seen in FIGURE 8. The phase response is typical of such a filter, although in the illustration, both the curve and the phase markings along the right-hand side of the graph are a top-to-bottom mirror image of the customary manner of presentation. According to the patent, *it is believed that*

Figure 8. A plot of the gain and phase of the signal emerging from the high-pass filter in Figure 6.



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the phase change in the signal resulting from passing the signal through the filter produces a directional or "three dimensional" effect in the resulting sound.

THE HARMONIC CREATOR

FIGURE 7 is a plot *of the transfer function of the harmonic creator,* showing the asymmetrical clipping of the input signal. *By clipping the peaks softly only low order harmonics are created. By selecting the proper threshold only transient portions of the signal become clipped. By clipping the signal on one side only both odd and even harmonics are created.*

The combination of the Aphex high-pass filter section and the diode create a large amount of overshoot and ringing. FIGURE 9 shows an overshoot of about 7 dB on a 5 kHz square wave passed through the side-chain. It is quite possible that a complex audio waveform—after filtering and processing—could create overshoots substantially greater than the 7 dB measured in the illustration. This means that the processed audio might overload the main channel, even if mixed in at a relatively low level.

OTHER FEATURES

The Aphex specification sheet refers to a de-esser circuit that does not seem to be mentioned in the patent. According to the spec. sheet, the de-esser is a gated notch filter with a center frequency of 5 kHz and a notch width that may be varied from 1 to 20 kHz. Notch depth is variable from 0 to 14 dB. There is also a compressor in the side chain, before the harmonic creator (the diode). According to a recent Aphex press release, the compressor is ". . . used to insure a stable amplitude level which increases control of the harmonic generator."

SIDE CHAIN OUTPUT CHARACTERISTICS

FIGURE 10 shows a real-time spectrum analysis of a processed 2 kHz sine wave. Note that the second harmonic (4 kHz) is only ten dB down from the fundamental, and that subsequent even-order harmonics (8, 12, 16 kHz) are significantly higher in level than the odd-order harmonics (6, 10, 14 kHz).

By now, it should be clear that the side-chain signal is a highly-distorted version of the input program. In fact, it produces phase, harmonic, amplitude and intermodulation distortion, but in a manner in which—when mixed in with the direct signal—creates an apparent improvement in overall sound quality.

SOME PSYCHO-ACOUSTICS

The apparent "enhancement" of an Aphex-processed signal can be at least partially examined in the light of two well-known psycho-acoustic phenomena:

1. In the presence of two equal-amplitude audio pulse streams, the listener will perceive that the longer-duration pulses are the "loudest."

2. The Fletcher-Munson effect notes that the listener with normal hearing is most sensitive to sounds originating within the 3-4 kHz range, with sensitivity decreasing at higher and lower frequencies.

These characteristics mean that the listener's subjective perception of loudness is often at variance with the sound level measured by instrumentation. The discrepancy between perceived and actual levels will vary, according to frequency, duration, and other variables. As suggested above, one of the more significant variables is the duration of impulses within the program.

THE HEARING MECHANISM

When an impulse occurs, it is transmitted through the listener's outer ear to the middle ear. Then, the impulse enters a tube in the inner ear, known as the Organ of Corti. Here, a frequency analysis is carried out, in a manner similar to the conventional frequency analyzer. Within the Organ of Corti, nerve endings are assigned to different frequencies, just like the band-pass filters in the analyzer.

The ear's oval window serves as the input terminal of the analyzer. The nerve endings then pick up their selected frequencies, and these are transmitted to the brain by parallel nerve fibers. Since the inner ear is only about 30 mm long, the frequency analysis—or in this case, frequency distribution—occurs almost instantaneously. Then, it becomes a function of the brain to "average out" the peaks. The averaging time for the brain is between 20 and 100 milliseconds, with a defined mean of 35 milliseconds.

Since the period of a 50 Hz signal is 20 milliseconds, it would appear that the comparitively slow "reaction time" of the brain itself would preclude it from differentiating peaks whose duration may be on the order of microseconds.

However, tests of apparent versus actual loudness conducted by others have shown that most listeners are indeed sensitive to minute variations in impulse duration. For example, gun shots from two different weapons register peak levels that vary by perhaps some 20 dB. Yet, depending on the impulse time of the sounds, the listener may perceive the shots to be about equal in loudness, or perhaps only a few dB apart.

Based on these observations, there appear to be two ways to effect the apparent loudness of audio transients. The conventional manner would be to vary the amplitude. But as an alternative, an artificial lengthening (or shortening) of the duration of the impulse would accomplish the same effect.

Aphex claims that its processor will alter the characteristics of a pulse in both ways, but with an emphasis on altering its duration. The reason this is necessary is that such pulses are often deteriorated by the recording or transmission medium. So, by artificially lengthening the pulse duration, the "missing ingredient" is returned to the sound, thus enhancing it, as compared to non-Aphex processing.

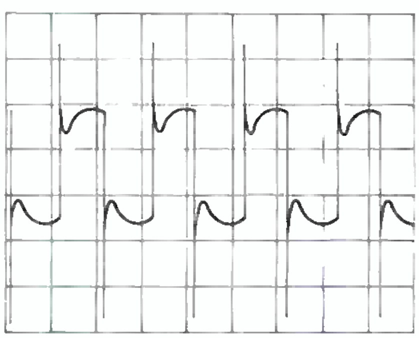
Aphex also feels that the harmonic distortion generated by the system accentuates the natural overtones of musical instruments, as well as the harmonics generated by the brain, which are primarily even-ordered and, therefore, more pleasing to the listener.

THE PATENT CLAIMS

The full text of the patent contains 35 separate claims, which are—for the most part—variations of the primary claim, which is;

1. A method of improving the quality of electronically processed sounds of music and speech comprising:

Figure 9. Scope trace of a 5 kHz square wave passed through the side chain.



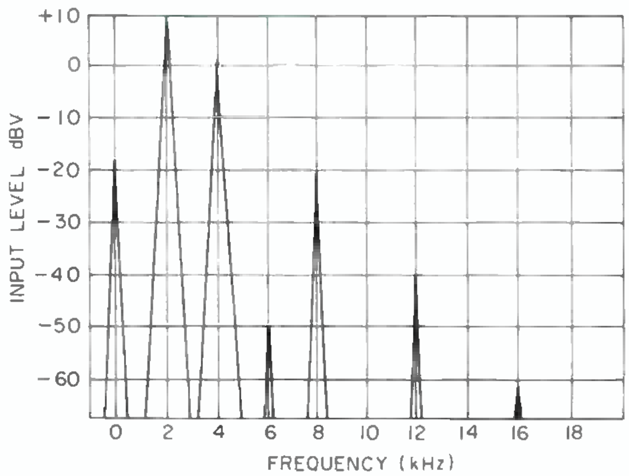


Figure 10. A real-time spectrum analysis of a processed 2 kHz sine wave.

(a) spliting the audio signal representative of the sound into two signal paths;

(b) passing the signal in one of the paths through a high pass filter designed to shift the phase of the sound passed in a frequency dependent manner;

(c) passing those frequencies in the output signal of the high pass filter whose amplitudes are above a preselected level into a harmonic creator, the level of the input to the harmonic creator varying directly with the level of the output from the high pass filter;

(d) attentuating all frequencies in the signal emerging from the harmonic creator a uniform amount to the level below the level of the signal in the outer path; and

(e) combining the attenuated signal containing the harmonics and the fundamental frequencies from which the harmonics are created with the signal in the other path.

ANALYZING THE CLAIM

In other words, the system seems to do the following;

(a) The input signal follows two signal paths—one direct and the other through a side chain of signal processing blocks.

(b) The side chain signal passes through a high-pass filter.

(c) Next, the side-chain signal is asymmetrically clipped by a diode.

(d) The filtered/clipped signal is attenuated.

(e) The filtered/clipped/attenuated signal is re-combined with the direct signal.

As already noted, the patent does not mention the notch filter or the compressor. Also, if there are some particularly unique circuit designs within the system, the patent does not seem to identify them. For, despite the abundance of "verbiage" in the claim, the system seems to be nothing more than a collection of fairly-standard audio building blocks, arranged in a unique manner.

Of course, this is not a detraction of the system. If you like what you hear, it really doesn't matter that there are, after all, no magic boxes contained therein. And apparently, a lot of people like what they hear, for the Aphex literature is liberally sprinkled with testimonials from all sorts of big names in the recording and broadcast industry. And, the Chicago Sun-Times reports that "Four out of five Grammy nominees for the best engineered record this year are Aphex improved." Even without any magic, four-out-of-five ain't bad. ■

BARRY HUFKER

Fibonacci Numbers, the "Golden Mean," and Audio Engineering

The famous "golden mean" has influenced mathematicians since the days of Pythagoras. As with Fibonacci numbers, it keeps turning up in the most unlikely places. What about in the world of audio?

IT WAS LEONARDO OF PISA—FIBONACCI, as he is more-popularly known—who gave the world the *Liber Abaci*, or, *Book of the Abacus*, which was instrumental in introducing Arabic numerals to Europe. And, it was a slightly-older Fibonacci (which translates from the Italian as "son of a simpleton") who devised the "Fibonacci Sequence"—a series of numbers which arose from a hypothetical mathematics problem based on the breeding of rabbits. His solution gave the sequence; 1, 1, 2, 3, 5, 8, 13, 21, 34, 55, 89, 144, 233, 377, 610, 987, . . ., n. Each number (except the first) is the sum of the two previous numbers.

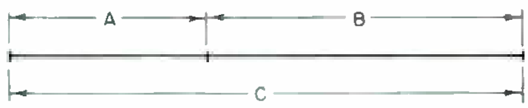
Demonstrating an interesting characteristic (at least to mathematicians), Fibonacci's sequence can be made to yield the ratio, 1.61803 (or, 0.61803), by dividing *any* number in the series, by the number adjacent to it. Although the numbers below 610 only give approximations of these ratios, from 987 onwards, the ratio is precise.

FIBONACCI, AND THE "GOLDEN MEAN"

Long before Fibonacci, this relationship was known as the "golden mean (or ratio)," or, as the "divine proportion." In fact, the work of the ancient Greek sculptor, Phidias, made use of the proportion, and the ratio has come to be known by mathematicians as "ϕ" (phi) in his honor. Perhaps not quite so famous as "π," but. . . .

Phi is the only number whose reciprocal ($1/\phi$) is equal to itself, minus one. In other words, $1/\phi = \phi - 1$. Actually, there are two values of ϕ; 1.61803 and -0.61803. Using the Fibonacci sequence, both values may be found by taking any two adjacent numbers (beginning with 987), and dividing them. Labelling the numbers A and B, we find that $A/B = 1.61803$, and $B/A = 0.61803$.

The golden ratio may also be found by dividing a line, C, into "mean," A, and "extreme," B, sections, such that the smaller section is to the greater, as the greater is to the whole. In other words, $A/B = B/C$, as shown below.



The frequency and diversity with which nature employs the Fibonacci numbers, and the golden ratio, suggests that its universality is more than "just a coincidence." And, Mother Nature has also allowed the Fibonacci proportions to serve both nature and art. Writing in the *Fibonacci*

Barry Hufker is a recording engineer/producer at KWMU, University of Missouri—St. Louis.

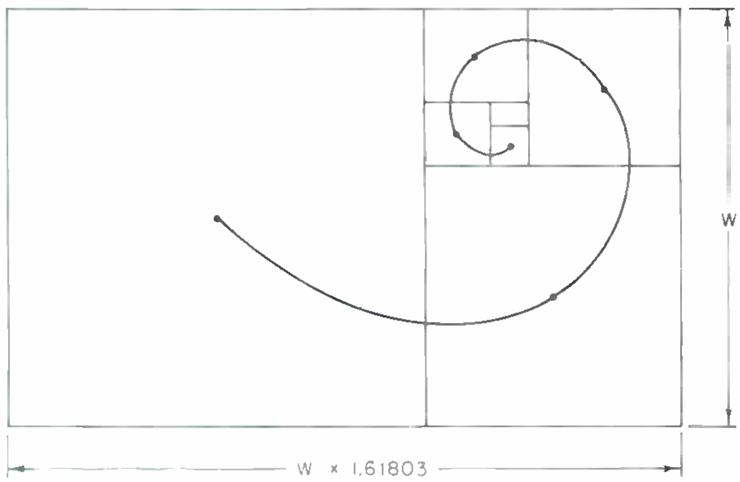


Figure 1. Constructing a "golden spiral." The spiral connects the center-points of progressively-smaller squares. Each square is a section of a golden rectangle.

Quarterly, Marjorie Bicknell and Verner E. Hoggatt, Jr. have reported that, according to German psychologists, ". . . most people do unconsciously favor 'golden proportions' when selecting pictures, cards, mirrors, wrapped parcels and other rectangular objects. In the same publication, Helen Hadian cites artists such as da Vinci, Cezanne, Seurat, Picasso, Gris and many others as users of the golden ratio in their work. Though an extremely detailed discussion would be needed to treat this topic fairly, for the purpose of this article it is sufficient to say that major elements of the paintings (body angle, body proportion and relationships between people and objects, etc.) are based on complex golden mean schemes. Perhaps it has been an intellectual, or instinctive, understanding of geometric proportion which has helped these artists earn their reputations.

THE "GOLDEN SPIRAL" AND THE EAR

An important and interesting outgrowth of the golden mean is the golden spiral, which was investigated by Jakob Bernoulli (who was so impressed, he had it engraved on his tombstone). This logarithmic spiral is derived when one draws a "golden rectangle" (length = width x 1.61803), and then sub-divides it to form a square. The remaining area is a smaller "golden rectangle," and if the process is repeated, a series of progressively-smaller squares are created. A curved line connecting the centers of these squares produces the "golden spiral"—a design that is found again and again in nature. It is the shape of rams' horns, snail shells, the form of galaxies, and as far as audio is concerned, it is the shape of the cochlea of the ear. The entire scientific significance of the cochlea's shape is not completely understood, although it is well-known that people "hear" logarithmically. And, there is strong evidence to demonstrate that, as the Fibonacci proportions appeal to the eye, they also appeal to the ear.

Again in the *Fibonacci Quarterly*, Edward L. Lowman's investigations reveal that ". . . proportion is certainly a major structural and expressive element in music," and, in twentieth-century music, two elements of temporal organization involving Fibonacci numbers stand out. One of these is the "structuring of the lengths of phrases and sections in Fibonacci proportions. The other is the use of Fibonacci numbers . . . to generate what are known as 'irrational' rhythmic values. . . . From the outset, composers found that generating such rhythms from little musical 'games' stimulated their imaginations, assured a measure of consistency, and taught them to free their minds from old and ingrained habits. A chart made from various permutations of the Fibonacci series, a great favorite with many composers, constantly reveals surprising and provocative relationships. In the composer's mind, these are often transformed immediately into musical ideas. . . . Fibonacci proportions have been among the most favored and useful tools."

Use of the Fibonacci sequence and the golden mean is not restricted to any type of music or period of music history. Lowman points out that the composer, Bela Bartók, used them extensively. Often, the golden mean is the major dividing point of a piece. The *Sonata for Two Pianos, and the Divertimento for String Orchestra* are just two such examples. Of the 443 measures of the *sonata's* first movement, Bartók chose the "golden mean" (measure 274) as the place to begin the recapitulation. A similar technique is found in the *Divertimento.*

In Bartók's *Music for Strings, Percussion and Celeste,* Fibonacci numbers are again present, though this time in a more sophisticated manner. The first movement of the piece is 88 measures long, and, according to Lowman, ". . . if we allow a measure's silence, we have 89. The *fff* climax of the movement arrives after 55 measures, of which the strings play the first 34 with mutes, removing them for the last 21. . . . The 34 bars following the climax are divided into 13 and 21 . . . and the final measures are divided again by a change of texture into groups of 13 and 8."

Lowman believes the listener will only subconsciously be aware of the proportions, and yet, "they will do their job just the same. What the listener will perceive is a sense of balance, a feel that the musical events he hears occur at the 'right' places, that they form intriguing patterns in time."

ARCHITECTURAL ACOUSTICS

Within the field of architectural acoustics, the Fibonacci sequence has met with considerable attention, in the design of listening rooms. In Michael Rettinger's book, "Acoustic Design and Noise Control—Volume I," the golden section is cited as one of six ratios that have been frequently used in room design.

In this application, the golden section is given as 1:1.62:2.62. If we take any three adjacent Fibonacci numbers, A, B, C, we will find that B/A = 1.62, and C/A = 2.62 (actually, 1.6803 and 2.61803). Interestingly enough, three of the other five ratio sets are reasonably close to the Fibonacci sequence.

Endless possibilities can be imagined, as various geometric shapes can be realized from the divine proportion. Golden ellipses, cuboids, rectangles, triangles and many more shapes could be selected to vary (and perhaps improved) studio acoustics. It is interesting to note that even the reflection of light rays within two glass plates (a studio window, perhaps) may be expressed in terms of Fibonacci numbers. Can the same be said of sound reflections? At present, the answer is unknown.

FIBONACCI AND BROADCASTING

The author has found the sequence to be of help in producing "King of Instruments"—a weekly program of live organ music and commentary, aired on KWMU-FM, in St. Louis. Golden proportions are utilized in several ways, including the balancing of audio levels.

When recording, the organ is allowed as much dynamic range as possible with the VU meter peaking at approximately 100 per cent (O VU). The voice of the program host is then set to peak at roughly 62 per cent (the golden mean of 100). In doing this a "golden" balance is theoretically established between voice and music which has worked quite well in actual practice.

As a third technique, the divine proportion can be lent to aid the pacing of a radio program, television show or film. In every production there is an attempt made to achieve the proper pace, and get the best balance between various elements. Whether these elements are scenes from a movie or the proper balance between the amount of talk and music in a radio program, there is an attempt to maintain interest for the viewer or listener. The Fibonacci proportions can lend that proper timing. Just as in music, the program can be divided and subdivided into sections whose length are dependent upon their importance and the golden ratios. These sections may be strung together or a single major division can be utilized, in a drama for instance, to realize maximum effect at that point. The possibilities are limitless.

CONCLUSION

It is extremely important to understand that a discussion of this subject is not just a fascination with "Numerology" or some cousin of Astrology. The subjective attraction the golden mean possesses has hopefully been presented as owning some merit. In no way whatsoever is it suggested the Fibonacci numbers or the divine proportion be used alone as some sort of "cure-all." No scientifically-proven techniques, research or devices should be discarded. The Fibonacci sequence and the mean are to be viewed simply as tools.

Edward Lowman, in speaking of Bela Bartók (and great composers in general), said that his use of the sequence and the mean as techniques "grew out of the shapes of the musical ideas themselves, just as have most techniques throughout history. One can imagine his realizing at some point that these proportions were what his ideas had been approaching all along. The technique was thus a means of focusing and clarifying the effect." The same can be said of their use in audio.

In music, the foundation for these techniques is already laid. A widespread understanding of such devices makes it hypothetically possible to write "hit" songs based on them or for advertising agencies to come up with more appealing jingles. The theoretical uses in audio of these proportions appear to be vast, though only time and experimentation will determine their true worth. ■

JOHN WORAM

A db Special Report On the Future of Radio and Recording

Due to recent events that may have a profound effect on all of us in the professional audio industry, we have postponed our coverage of the recent AES convention to make space for this special report.

PART OF LAST MONTH'S EDITORIAL concerned itself with a Further Notice of Inquiry (FNOI), issued by the Federal Communications Commission, in the matter of f.m. Quadraphonic Broadcasting. At the time the editorial was written, it seemed that no more needed to be said on the subject—at least in the pages of **db**. However, some recent events deserve the further attention of **db** readers—as the effects of this FNOI could very well have far-reaching implications on all of us.

A PERSONAL NOTE

These comments are being written, not as another editorial, but as one man's opinions on the subject. Since I have an "axe to grind" in the matter of quadraphonic sound, this should be made clear up front. I am by no means an impartial observer, and I don't think it would be quite right to offer these words under the format of an editorial.

As a consultant, I am involved in the advancement of discrete quadraphonic broadcasting. Others are similarly involved in the advancement of matrix broadcasting. That being the case, I will avoid using this space to proclaim the joys of any particular type of quad broadcasting system. Frankly, I think we've all heard quite enough about the superiority of this or that system by now, and we should all be content to let the marketplace decide who is building the "better mousetrap."

However, I am getting a little perplexed by the avalanche of mis-information that seems to be circulating around these days. Most of it is in the form of letters and petitions to the FCC, and now, even the mailboxes of Congressmen are filling up with letters from concerned constituents, who seem to be getting some funny ideas about what's going on in the world of broadcasting. If our glorious leaders in Washington are in any way influenced by the letters piling up on their desks, the consequences could be disastrous to all of us. In fact, it could almost be said that the quadraphonic issue has become of secondary interest, when compared to the "everything else" that is now involved in this FNOI.

THE BACKGROUND MUSIC LOBBY

Perhaps it's just one of life's little coincidences, but there is a sudden deluge of letters from Muzak licensees, and most of these claim that various quadraphonic standards will put them out of business. One Muzak operator writes that, *"So far as the question of quadraphonic broadcasting is concerned, there has been no demonstrable evidence of any quantitative public interest, and therefore no recognizable necessity for such service."*

Another letter reports that, *". . . the apparent interest in FM quad sound on the part of the general public is extremely limited, as borne out by the lack of large quantities of program material."*

Still another writer says, *"The adoption of quadraphonic broadcasting would negate all our efforts during the past years and make it an impossibility to continue to operate a profitable enterprise . . . I urge the Commissioner's disapproval of quadraphonic broadcasting."*

One more excerpt should be enough: *"I understand that the FCC will hold hearings to determine the feasibility of quadraphonic sound. As an entire industry totally de-*

pendent on SCAs, you must realize this would destroy our business."

WHAT'S GOING ON HERE?

Why on earth are the Muzak people so vehemently against letting the American public get a chance to experience quadraphonic broadcasting? Since these writers are responding to the Commission's FNOI, could it be that not one of them noticed the following remarks contained within that Notice? *"Over 2,000 comments from broadcasters, manufacturers, and the general public were received. With few exceptions, comments were in support of some form of quadraphonic broadcasting . . . The FCC has concluded that there is substantial interest in FM quadraphonic broadcasting."*

Let's run all this through a "B. S. filter" and see what comes out the other end. Most of us know that the poor broadcaster can do little more than take a coffee break without first getting permission from the FCC. If a station decided to arbitrarily abandon its SCA commitments in favor of quad, what do you suppose would happen? Obviously, there would be a breach of contract suit, and the FCC would raise hell with the station management. Licenses would be in jeopardy, and a generally-unpleasant time would be had by all.

Another question worth asking is, why would a station want to desert its SCA subscribers in the first place? Presumably, the SCA service is profitable, and both the station and the SCA service organization were not forced at gunpoint into their present contractual agreement.

But what about the radio station that does not offer an SCA service at this time? Should the FCC forbid it from broadcasting a discrete quadraphonic program, if it feels such a service would be profitable, in the public interest, or perhaps, both? Notice that I am studiously avoiding making any comments about whether such programming is an improvement over conventional stereo. That decision is best left to the marketplace. Rather, I am asking; is it in the public interest for the government to restrict quad broadcasting, in order to protect a private-interest organization from the spectre of a little healthy competition?

SERVICES FOR THE BLIND

In addition to the background music industry, there are numerous services to the blind that make use of SCA. And, it is interesting to note that while the background music people seem to be irrationally obsessed with getting the government to protect them from the hassles of free competition in the marketplace, the blind are expressing legitimate concern over what will happen to them if the FCC weakens or entirely destroys SCA service by reducing channel spacing to 150 or 100 kHz.

Judging by the letters showing up in Washington, it seems that many blind listeners are getting the impression that it is quadraphonic broadcasting that will destroy their highly-prized SCA reading services. If that were really the case, many of us would be hard-pressed to defend our position. The SCA service to the blind—as well as many other public-interest SCA programs—deserve to continue and flourish.

A LITTLE POLITICS

As usual in Washington, this entire matter has its share of political overtones, which are clearly recognized by many. One registered professional engineer sums it up nicely: *"For what are obvious political considerations the Commission has for some time considered reducing the channel spacing to 150 kHz or even 100 kHz. The objective is to make additional channels available to local broadcasters and minority interest groups. While the objective is laudable the means are ironic in that while additional broadcasters might be accommodated they would be given a degraded service."* As noted in the April editorial in **db**, the Commission acknowledges that *"100 kHz channel spacing would probably preclude . . . stereo and SCA."* That being the case, it would also preclude quad. A 150 kHz channel spacing would cause deterioration of *all* presently-known f.m. formats. Again, it is ironic that the deterioration would be most noticeable in mono, and least objectionable in quad.

SEPARATING THE ISSUES

By now, I hope it is clear that there are actually two issues confronting us: one is quad broadcasting; the other is the future good health of the entire recording and broadcasting industry—including SCA. At this moment, you may not be particularly concerned about the former, but, what about the latter? Doesn't that concern all of us just a little bit?

In order to preserve the current state-of-the-art in f.m. broadcasting (and therefore, in recording as well), no reduction in channel spacing should be tolerated. That means that some stations may be able to broadcast discrete quad if they so desire, while others may choose to offer SCA services. In fact, a station may actually offer *increased* SCA service if it chooses to do so—a point that also seems to have escaped the notice of many concerned parties.

SOME FURTHER COMMENTS

Perhaps the comments of the National Public Radio Network make the most sense: *"NPR recommends that the Commission adopt standards for quadraphonic broadcasting which allow individual broadcasters to choose the system best suited to the needs of their listeners and that there be compatibility among the systems adopted.* NPR also notes that, ". . . *in fact, the suggested spacing reductions would also impair or preclude current SCA and stereophonic broadcasts."* On the matter of channel spacing, *"Public radio has suffered the effects of congestion in the reserved portion of the band for many years, and is in full agreement, therefore, with the reasoning which motivates this proposal (but) we cannot support the adoption of a plan for reducing channel spacing which will impair the performance of existing SCA and stereo services."*

In contrast, here are some excerpts from the comments filed by Muzak: *"The discrete quadraphonic proposals offer no recognizable advantages and many disadvantages. Discrete quadraphonic would reduce FM service for all types of receivers, eliminate many valuable SCA services, and degrade sound for listeners using existing equipment . . . We submit there is no justification for further consideration of standards that would permit broadcasters to 'use up' their FM channels by broadcasting in discrete quadraphonic."*

FILING REPLY COMMENTS

If you have any interest in this matter, the Commission has extended the period for filing reply comments until July 11, 1979.

As of the moment (26 May, that is) the Commission has received little or no reaction from either the broadcast or the recording industry on the matter of reduced channel spacing. What's the matter folks, don't you give a damn?

If you do, your comments should be addressed to:
William J. Tricarico, Secretary
Federal Communications Commission
Docket 21310
Washington, D.C. 20554 ■

WAYNE JONES

Audio Tests and Measurements—Part III

As sound quality continues to improve, so too must the methods and instruments used in performance measurement.

DISTORTION AND ITS MEASUREMENT have undergone radical changes in recent years. In general, audio equipment has significantly lower distortion than a few years ago. This has resulted in even-greater demands on test equipment, which must always perform better than the device under test. In the never-ending quest for improved sound quality, distortion characteristics are being examined in more detail. Different types of distortion are being evaluated, more detailed distortion specifications are being written and, to overcome some existing technical limitations, new methods of distortion measurement are being used. Today, popular (if such a word may be used) types of distortion include total harmonic distortion, intermodulation distortion, difference-frequency distortion, and transient intermodulation distortion. There are a variety of techniques for measuring each type of distortion.

A significant breakthrough in the measurement of low levels of harmonic distortion was realized a couple of years ago, with new test equipment that enabled distortion measurement to extremely low levels, in the order of 0.002 per cent. Previously, popular instruments had been able to measure down to levels of about 0.04 per cent. As audio equipment in general improved, it became increasingly difficult to make distortion measurements with existing analyzers. One may ask, why is it necessary to measure distortion down to such incredibly low levels since the human ear cannot hear this distortion anyway?

TESTING THE TOTAL SYSTEM

As today's equipment becomes more complex, it becomes important to examine the distortion characteristics of the total system. A low-distortion system requires even-lower distortion in the individual components of the system. Hence the requirement for instruments that can measure these incredibly low levels of distortion.

HARMONIC DISTORTION

Harmonic distortion is the most-often-measured distortion parameter. A single-frequency, low-distortion sine wave is sent to the device under test, and the signal at the output of the device under test is examined for the presence of harmonics of the original test signal. There are basically two methods employed in test instruments to make this measurement. The most common technique is to use a band-reject, or notch, filter to remove the fundamental from the measured signal. Using a well-constructed notch filter, with manually or automatically adjustable parameters, a notch depth of 100 dB or more can be achieved. Having removed the fundamental in this manner, the succeeding circuits of the analyzer measure everything that is left. A ratiometric circuit or "normalizer" then presents the amplitude of these components as a proportion of the full signal as it appears at the input terminals. This number is, by definition, the percentage distortion. This technique is used in virtually all commercially-available distortion analyzers.

A newer technique that is finding interest is the use of a spectrum analyzer to measure distortion. An alternative to this approach is the use of a wave analyzer. In both cases a narrow band-pass filter is swept over a band that includes the fundamental frequency and its components. For example, if the test frequency is 1 kHz, the wave analyzer or spectrum analyzer will be used to look for the presence of the second harmonic at 2 kHz, the third harmonic at 3 kHz and so on. The root-mean-square of the amplitudes of each of these harmonic components as a percentage of the total signal is the per cent distortion.

Each technique has advantages and disadvantages. Some measurement situations can only be handled by using one particular technique, while certain measurement situations can benefit from the use of both techniques simultaneously. The notch filter technique is certainly the easiest. New distortion analyzers offer a high degree of automation, requiring very little manual manipulation of the controls to achieve a result. They provide a single number

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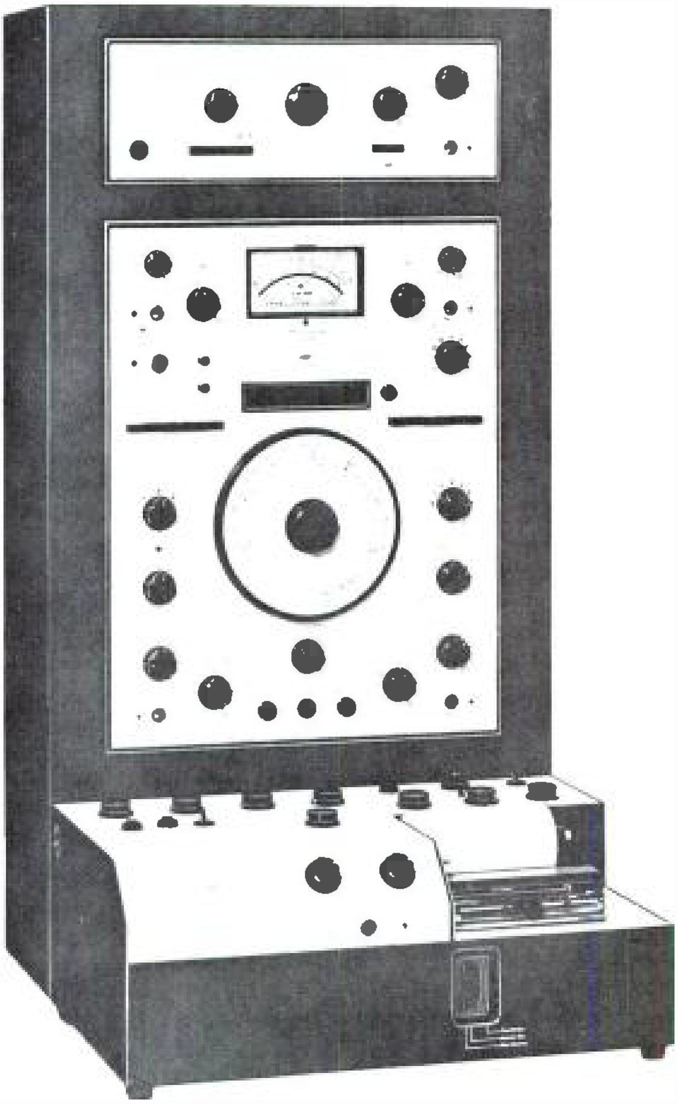


Bruel & Kaejr 2010 Heterodyne Analyzer, 2307 Level Recorder and 1902 Distortion Control Unit. One of the most comprehensive distortion measurement systems around. Plots harmonic distortion, intermodulation distortion and difference-frequency distortion, versus-frequency from 2 Hz to 200 kHz. The distortion control unit allows selection of the second through the fifth, even or odd order components to generate a powerful set of graphs of system performance. Can measure components down to —80 dB (0.01 per cent). The system can also provide frequency response plots, spectrum analysis plots and, with additional components from B&K, the system can be expanded to plot other parameters such as phase-versus-frequency.

that is the direct reading of percentage distortion, so they are fast, easy-to-use, and give the result directly. On the other hand, the spectrum analyzer requires a fair amount of manual manipulation of the controls. First the spectrum analyzer has to be set up to make the proper measurement: the bandwidth of the filter, the sweep width and the sweep speed must all be determined. Then, the amplitudes of each component must be measured by examining the plotted curve and the mathematics must be performed to derive the rms sum of the components. Finally, this value must be compared to the amplitude of the fundamental—a tedious project. However, in spite of these additional difficulties, in some cases the spectrum analyzer or wave analyzer technique is the only one that will yield an accurate result.

The notch filter technique does not give a reading of distortion, but rather a reading of distortion-plus-noise, plus any other signal that is not the fundamental. In a high-quality system or a controlled measurement, the noise and other unrelated components may be significantly lower than the distortion products. In this case the distortion reading is valid. However, this is often not the case. For example, in acoustic measurements or measurements of tape recorder performance, noise and other unrelated signals may be of the same magnitude, or even greater than the harmonically-related components. In these cases, the simple distortion analyzer will give a significantly erroneous result. The analyzer on the other hand will give an extremely accurate result. For this reason the Institute of High Fidelity (IHF) has recommended in their new "Standard Methods of Measurement for Audio Amplifiers" that a spectrum analyzer be used for measurement of harmonic distortion.

There is another drawback however, to the use of the spectrum analyzer or wave analyzer. The typical dynamic range of contemporary distortion analyzers is in the order of 80 dB. Some newer analyzers can achieve as much as 90 dB. This means that these devices are capable of measuring distortion to residuals of around 0.01 per cent In fact, allowing for errors in the analyzer, the ability to resolve details at the low end of the dynamic range and the mathematics used to combine the components, a spectrum analyzer with an 80 dB range may only be able to measure distortion down to between 0.02 and 0.05 per cent—quite a bit higher than contemporary distortion analyzers. For extremely critical measurements there is a way out: use both techniques simultaneously. If the spectrum analyzer is preceeded by a simple notch filter, the dynamic range of the spectrum analyzer is improved by an amount equal to the notch depth of the filter. Thus a notch filter with a depth of 40 dB and a spectrum analyzer with a dynamic range of 80 dB or more will allow distortion measurements to be made to below 0.001 per cent. This dramatically eases the burden on the measurement instruments. A simple way to do this is to use a spectrum analyzer to analyze the residual components that come from the distortion monitor output of a normal distortion analyzer. Once such a measurement set-up is calibrated, it provides a highly-sensitive and extremely-accurate measurement facility. For highly critical applications, this is one of the best techniques to be used.

Not everyone has the budget or the desire to use such sophisticated techniques in routine distortion measurement. Careful use of the filters on a conventional distortion analyzer can greatly enhance its accuracy. These filters allow a reduction of the bandwidth of the measuring circuit. They can thus allow the instrument to exclude noise or hum and concentrate on the measurement of the harmonics of the signal under test. The residual distortion signal should also be presented on a dual-trace oscilloscope, along with the fundamental signal (as a reference) on the second trace. This allows instant observation of the character of the residual signal, which not only shows up measurement errors but can give a wealth of information as to the cause of the distortion. An unfiltered or unweighted distortion residual may show the presence of a significant hum component, high-frequency oscillation, rf pickup, or some other abnormality that is not distortion. All of these signals will immediately show up on the oscilloscope display. The high-pass and low-pass filters provided on most analyzers can, in some cases, reduce the effects of these signals sufficiently to yield a reasonably-accurate distortion result.

The character of the residual should also be noted; even-order components, such as a dominant second harmonic, would indicate an asymmetry in the device under test. Dominant odd-order harmonics are characteristic of

Hewlett-Packard 339A Distortion Measuring Set. Combines an ultra-low distortion oscillator and high sensitivity analyzer. Can Measure total harmonic distortion down to 0.0018 per cent (—95 dB). Also measures level down to —80 dBv. Includes automatic set level and automatic nulling. Manual adjustments such as tuning and range selection are facilitated by high/low indicator lights. The meter is a true-rms type for accurate indication of harmonic content. Switchable meter response characteristics of normal or VU and an a.m. detector facilitate measurement to broadcast standards. One high-pass and two low-pass filters provide noise and distortion residual qualification capability. A "relative level" control permits an arbitrary 0 dB reference to be set.

tape recording systems; sharp spikes will reveal crossover distortion, typical of power amplifiers with incorrect bias and, finally, unstable residual waveforms will indicate the presence of asynchronous signals, such as cross-talk or line-frequency power supply hum.

INTERMODULATION DISTORTION

Another type of distortion is intermodulation distortion. Originally developed by the motion picture industry to measure distortion on optical film sound tracks, measurements of this type of distortion are seeing increased interest. Unlike harmonic distortion, intermodulation distortion is measured by using two tones, mixed in a particular format. The original film technique, defined by the SMPTE as a standard, dictates the use of two frequencies at 60 Hz and 7 kHz, mixed in a four-to-one power ratio; that is, the 7 kHz signal must be 12 dB below the 60 Hz signal. The low frequency signal was chosen for convenience, as it can easily be derived from the power line. The high frequency signal was chosen because of the bandwidth limitations of film, which did not extend much beyond 7 kHz.

A device with intermodulation distortion will show the occurrence of modulation of the high frequency signal by the low frequency signal. Additional components will be generated at the sum- and difference-frequencies, and lesser-amplitude signals will be generated at the sum- and difference-frequencies of the harmonics of the original signal. Thus, in the case of the SMPTE standard, harmonics will be generated at 6.94 kHz, 7.06 kHz, 6.88 kHz, 7.12 kHz, and so on. Additional components may also be generated around 14 kHz, 21 kHz and above. The primary areas of interest are the side bands clustered around 7 kHz, and this is the area that is measured on most conventional imd analyzers.

The most popular technique for measuring imd is as follows: the two signals are generated by fixed, tuned oscillators, mixed in a summing amplifier in the proper ratio and fed to the device under test. Looked at on a scope, this composite signal will look like a "fat" sine wave. In fact, the sine wave is 60 Hz and the fuzziness of the sine wave is the small 7 kHz component. The analyzer consists of a high-pass filter, which first removes the low frequency component, leaving only the 7 kHz and imd products. In effect, these products are: modulations of the 7 kHz signal, and are riding on the 7 kHz carrier. Demodulating the signal will yield the results. An absolute-value circuit, followed by a low-pass filter, provides the residual. The amplitude of this residual—as a proportion of the amplitude of the 7 kHz signal—is the percentage of intermodulation distortion.

As in harmonic distortion, a spectrum analyzer may also be used to measure intermodulation distortion. A spectrum width centered around the high frequency component will show the side bands clustered about it. Their responsive amplitudes can be determined, the rms sum calculated and compared to the amplitude of the high frequency signal.

Imd is by no means limited to the SMPTE standard. Indeed, several other standards exist, as well as many non-standard approaches. The low-frequency signal can be anywhere within the pass-band of the device under test and below the cutoff point of the high- and low-pass filters in the analyzer. In practice, this usually means 20 Hz to 500 Hz. The actual choice of frequency has little effect on the final reading. However, the choice of the high frequency signal is more critical. The lower constraint is the cutoff frequency of the high-pass filter in the analyzer, and the upper limit is the bandwidth of the device under test. Just as harmonic distortion measure-

Sound Technology 1701A Distortion Measurement System. The original model 1700, introduced a few years ago, marked a new generation of analyzers achieving significantly higher performance. The 1701A measures down to 0.001 per cent mid-band with convenient automatic nulling and push button operation. A new feature is selectable meter response characteristics of true rms, average or peak to facilitate noise measurements and distortion measurements to the new IHF standards. Selectable low- and high-pass filters, balanced input and tracking low distortion generator enhance the system's capabilities. Options include automatic set level and intermodulation distortion measuring capability to SMPTE standards.

ments are made at several frequencies, particularly in the high frequency region, intermodulation distortion should also be measured at several frequencies. Some instruments even allow the high-frequency signal to be swept, and a plot of imd-versus-frequency generated.

Proponents of imd measurements claim several advantages over harmonic distortion measurements. For one thing, they state that the imd test signal is a closer approximation to music or speech than the single-sine wave of thd measurement. For another, they say the device with high imd will sound more annoying than one with a high thd. This is because the imd components are not harmonically related to the original signal, as in harmonic distortion. Also intermodulation distortion results in more components than harmonic distortion, and is usually higher, since more energy is used for the two signals. Some devices may show quite good harmonic distortion performance while the intermodulation distortion is excessive. Certain abnormalities lend themselves to discovery far easier with imd measurements. Also, imd measurements can be made right up to the upper bandwidth limit of the device under test, while meaningful harmonic distortion measurements can only be made up to about one-third of the upper bandwidth limit of the device. For example, harmonic distortion measurements in a tape recorder at 15 kHz may show a deceivingly-good result, simply because the second- and third-order products are outside the pass-band of the medium. On the other hand, the im products clustered about the 15 kHz signal, are well within the measurement capability of the instrument.

DIFFERENCE-FREQUENCY DISTORTION

In the general case, intermodulation distortion can be measured with any two signals. It is not really necessary to use one low frequency and one high frequency signal. A popular practice in Europe uses two high frequency signals with a small frequency difference; for example, 7 kHz and 7.1 kHz. Difference-frequency distortion is a special case of intermodulation distortion and considers only the difference products. The second-order difference product in the above example will show up at 100 Hz; that is, the difference between 7.1 and 7 kHz. It has recently become popular to make im measurement on amplifiers and speakers using this twin-tone method. The most sophisticated instrumentation allows the two signals to be swept with a constant frequency difference. The lower frequency limit imposed by the analyzer is usually around a few hundred hertz. The high-frequency limit again is the upper bandwidth cutoff point of the device under test. Two sine waves with a constant frequency difference of say, 100 or 1000 Hz are swept from 1 or 2 kHz up to the upper limit of the device, perhaps 30 to 100 kHz. The new IHF-im standard specifies a frequency separation of 1 kHz, with measurements to be made from 2.5 kHz to the upper band edge of the device under test. However, it also specifies that all distortion components be measured; that is, the second-through the fifth-order, not only the difference-frequency component.

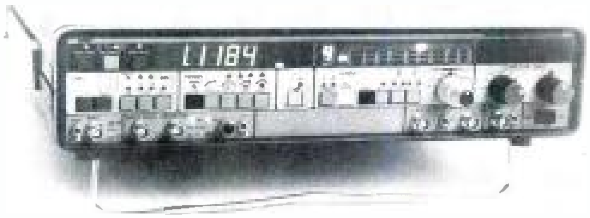
TRANSIENT INTERMODULATION DISTORTION

Another special case of intermodulation distortion is transient intermodulation distortion. This form of dynamic intermodulation distortion is caused by slew-rate limiting of a feedback amplifier during a rapidly-rising input signal. Under such conditions, an amplifier will exhibit a high amount of non-linearity. The same amplifier may exhibit quite good total-harmonic and intermodulation distortion measurement prior to the slew-rate limiting bandwidth, as these tests will not exhibit this problem.

A number of tests have been devised to measure tim. One such test involves the combination of a square wave and a sine wave fed to the device under test and a spectrum analysis of the signal available at the output. The rapid rise time of the square wave will cause the tim problem to manifest itself. The measurement is not easy however; the spectrum analysis plot is extremely detailed, and requires considerable interpretive analysis. The author is unaware of any method of measuring tim that has been adopted by any standards group. Similarly there is no simple tim analyzer on the market which reads out in per cent tim. Indeed, using the popular measurement techniques, the fabrication of such an instrument would be considerably complex and unwieldy.

Fortunately, the mechanisms that cause tim can be measured with more-conventional measurement techniques. We have stated that tim is caused by slew-rate limiting of the device under test. It has therefore been called slew-induced distortion or sid. Sid will show up as excessive harmonic distortion at high frequencies. Thd measurements in the area from 20 kHz to 100 kHz will usually show strong evidence of the presence or absence of tim. Just about every amplifying device shows increasing thd with increasing frequency. If a discernable break-point is noticed in the thd plot from 20 to 100 kHz, and high orders of thd above this break-point are found, it is almost certainly an indication of high tim. Such measurements are relatively easy to perform, standard instrumentation is available, and standards have been written for their measurement. They are not subjective measurements requiring interpretive analysis, but yield a single numerical quantity.

Careful use of standard methods of thd and imd measurement with sufficient variation in parameters will usually document all relevant non-linear distortion characteristics of the device under test. Parameter variation means



Amber 5500 Distortion Analysis System. A powerful audio measurement and analysis instrument incorporating a high degree of automation. Microprocessor based, it will measure level, total harmonic distortion, intermodulation distortion, difference-frequency distortion, and narrow-band amplitude. It incorporates a highly accurate and fast digital readout and, with the computational power of the microprocessor, can display results in volts, watts, dBm, dBv, dB, per cent, Hz or kHz. Internal intelligence allows the instrument to configure itself rapidly for a variety of tests, auto-ranging over a 160 dB range, auto-tuning over a 4 decade frequency range, auto-nulling and automatically adjusting any other required electrical parameter to generate a test result. Eight selectable filters and weighting networks and six detector characteristics—including true rms—provide flexibility in noise measurements. Unique capabilities include the ability to sweep and generate thd, imd or dfd plots versus-frequency and the ability to store up to a hundred test set-ups for production line testing. Can be interfaced to the IEEE-488 bus.

the measurements must be made at several frequencies and several amplitudes. The ideal situation is a family of curves; that is, a plot of thd or imd-versus-frequency-versus-amplitude. Recently introduced instrumentation has the power to generate these curves in a matter of minutes, something that would normally have taken days of effort using more conventional instrumentation. Such a set of curves is an extremely powerful tool in development and maintenance of high-performance audio equipment. (For more information, see Almon Clegg's article; "Three-dimensional Analysis: It's About Space" in the April, 1979 issue of **db**—Ed.)

AUTOMATIC DISTORTION ANALYZERS

Virtually all total harmonic distortion analyzers measure distortion by using a notch-rejection filter. To read distortion, four basic adjustments must be made to this type of analyzer. First—because the circuit is basically a filter—it must be tuned to the frequency of the fundamental signal. Second, the amplitude of the fundamental signal must be measured, and this usually means the input gain of the measuring instrument must be adjusted to bring the signal within the measuring range of the instrument. Third, the filter parameters must be adjusted to achieve a maximum null or rejection of the fundamental. Without this adjustment, attenuation is only about 30 dB, but with adjustment the attenuation can increase to over 100 dB. Fourth, the amplitude of the remaining residual (after the fundamental rejection) must be measured, and compared to the amplitude of the full signal, to provide the percentage distortion.

Most contemporary analyzers have achieved some degree of automation in these adjustments. Tuning is simplified by ganging the frequency controls of the oscillator and the analyzer. Newer analyzers have "automatic set-level" which simplifies the measurement of the full signal amplitude. The input gain of the instrument is manually set within the correct 10 dB range and the automatic circuitry takes over from there. Almost all new analyzers have automatic nulling circuitry: as long as the filter is tuned within a few per cent of the correct frequency, internal circuits will automatically null out the fundamental to the maximum rejection capabilities of the instrument. These degrees of automation mean that distortion measurement can be made by easily setting a few step-type controls to the correct value. The accuracy of the measurement is not dependent on the skill of the operator adjusting variable controls to achieve proper circuit performance.

Some instruments have achieved an even-higher degree of automation. Input and meter circuits contain automatic ranging, or, the filter can automatically tune. All this is in addition to the normal automation features of nulling and set-level. The result is an instrument that, with no manual intervention whatsoever, will lock on to a signal, make all the electronic adjustments necessary, and provide a readout of percentage distortion. This automation usually also achieves quite rapid operation—so rapidly in fact that the instrument will track a sweep. The signal to be measured may be sweeping in frequency or amplitude, with the automatic distortion analyzer providing a continuous readout of percentage distortion. Connect an XY recorder and you have a plot of distortion-versus-frequency, or distortion-versus-amplitude. Repeat the plot several times at different amplitudes and you can generate a family of curves of

Amber model 3500 Distortion Measurement Set. A compact battery/mains operated unit small enough to fit in a briefcase. Measures wide-band level, narrow-band level, total harmonic distortion, noise and cross-talk over the range 10 Hz to 100 kHz. Auto-nulling, auto-set level, true rms detection, selectable weighting networks, tracking analyzer/oscillator and spectrum analysis capability are among the features. State-of-the-art performance includes distortion measurement to 0.002 per cent and level to —120 dBv. Led "turn signal" indicators speed operation and a residual monitor output is provided for an external oscilloscope.

distortion-versus-frequency-versus-amplitude within just a few minutes.

These are examples of the complexity of the instrument increasing at the same time as its operation is simplified. A completely-automatic distortion analyzer is incredibly more complex internally than a simple manual analyzer. Auto-ranging, auto-tuning and auto-nulling circuits are difficult enough to achieve in conventional measuring instruments such as voltmeters, but in a distortion analyzer the circuits must also not introduce any added residual distortion. Conventional approaches to auto-ranging must be abandoned.

The benefits achieved from such degrees of automation include increased speed and far-simpler front-panel controls. As a matter of fact, the only front-panel controls necessary are those required to receive decisions made by the operator, such as: what measurements are to be made, how is the readout to be presented or, what type of filtering is to be used. These inputs are solely to do with requests by the operator. Front-panel controls that adjust circuit parameters to achieve proper instrument operation are no longer required. With the sophisticated circuitry available today, and particularly with the advent of the microprocessor, the instrument itself is usually in a far-better position to make decisions on what adjustments should be made than is the operator. Additionally, these adjustments can be made more accurately, more rapidly and more consistently. The results are: higher levels of confidence in the readout, more rapid generation of data, and, relief from the tedious adjustments required when a large quantity of measurements must be made, such as in production testing, or detailed analysis in an R&D application.

AUTOMATIC TEST EQUIPMENT

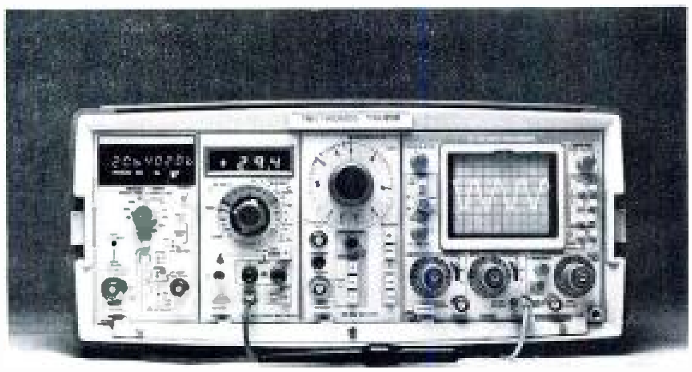
Automatic test equipment (ATE) is becoming increasingly popular. Just about every user of test equipment has requirements for some degree of automatic test. Production-line applications are the most obvious, where high volumes of equipment must be tested rapidly and accurately. Here, the cost of the automatic test system is usually secondary to the savings realized by time and manpower reductions.

The requirements for automatic testing have been met with a number of developments. For one thing, individual test equipment itself is becoming more automatic, as described earlier. But an ATE system can seldom be satisfied with a single piece of equipment, no matter how sophisticated or intelligent that equipment is. The requirement to configure ATE systems consisting of more than one piece of equipment and have some means of programming or controlling a sequence of tests lead to the development of an interface standard. Hewlett-Packard originally developed this standard, and with minor modifications it was adopted as an IEEE standard, available to the entire industry. (General Purpose Interface Bus [GPIB], IEEE Standard 488-1975—Ed.) It is now possible to buy almost any type of test equipment with this interface incorporated.

Such an ATE system usually consists of one controller and a quantity of up to fourteen additional instruments referred to as "talkers" or "listeners." A talker would be an instrument that gathers data and must talk to the interface bus, such as a voltmeter. A listener would be a device that simply receives instructions from the bus, such as an oscillator. (Note that the voltmeter "talks" to the interface bus, telling it about the voltage it is measuring. The oscillator "listens" for instructions to change amplitude and/or frequency. The actual voltage being measured, and the output of the oscillator, are not part of the interface bus structure—Ed.) The controller's function is to orchestrate the entire system. A program defining a series of tests is written and stored in the controller. The controller then talks to all of the individual pieces of test equipment, instructing them to configure themselves in a particular fashion, generate a particular signal, or, make a particular measurement. The measuring devices follow these instructions and send their measured data back to the controller over the interface bus. The controller may then manipulate this data in a number of ways. It may simply store the results, or print them out. It may generate a graph, or it may perform a series of arithmetic functions on this and other data, or, it may use these results to modify the program for future measurements.

Almost every equipment user's requirements are unique. The IEEE-488 interface bus permits the user a high degree of flexibility in configuring his particular ATE system. Theoretically, he can select equipment from a wide variety of manufacturers, connect the equipment together, write a program and have an ATE system. In practice its not quite that simple, but it is certainly a lot easier than earlier techniques. Previously, ATE systems required the construction of custom equipment, the building of special interfaces and the fabrication of elaborate controllers. The custom requirements of such systems certainly dominated the cost and in spite of this expense the system was rigidly defined. The industry-wide interface standard changes all this. It is normally possible to configure an ATE system using off-the-shelf standard components.

Progress in interface-compatible audio testing equipment has not been as fast as other segments of the electronic industry. There are numerous programmable oscillators and synthesizers on the market, but none of them have very low levels of distortion.



Tektronix TM-500 Series. A family of over 40 modular instruments and six mainframes. Included in the series are 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 kHz 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.

Similarly, the only way to measure distortion in an ATE system has been using an interface-compatible spectrum analyzer. This is considerably more expensive than a conventional distortion analyzer, and the method of measurement is more tedious and time consuming. Also, in general, the spectrum analyzer cannot measure to the same low residual levels as a conventional distortion analyzer. However, the system is programmable, and can be controlled in an ATE system.

Recently, a new instrument has been introduced that measures distortion completely automatically. With a microprocessor controlling all internal functions, the instrument can also be controlled by an ATE system over the IEEE-488 bus.

As audio consoles and other types of audio equipment become programmable, increasing use will be made of ATE systems in more-routine audio testing requirements. Imagine a system testing a programmable multi-track recording console and using a series of programmable audio test instruments, such as a distortion analyzer, plotter and so on. The hundreds or thousands of individual circuits and signal paths could be completely tested for all audio parameters thoroughly and rapidly. The final output data could be in whatever format was desired, the data could be presented as hard-copy graphs of response, distortion, noise, cross-talk and phase response. The system would print out upper and lower limits: that is, the highest and lowest recorded distortion or response deviation, as well as the average. The system could even check these results against stored data and indicate those circuits that were out of spec.

Such a system could achieve in hours what would previously have taken days—or more likely, weeks—to achieve. Additionally, the accuracy of the information would not be a function of operator fatigue or interpretation. The thoroughness of the measurement would no longer be subject to time factors or human error. Once an automatic test program has been written, it will be followed rigorously by the equipment. The equipment has neither the ability nor the desire to editorialize on the written program, it doesn't get tired, it doesn't forget and it doesn't take coffee breaks.

FUTURE DIRECTIONS

Test equipment is becoming more and more intelligent. Computers are finding a welcome home in more and more diversified test equipment. Microprocessor-based test equipment is becoming the standard for almost all new types of test equipment being developed. The architecture of this equipment is such that the system configuration is defined by software, and not by hard-wired hardware, as in previous instruments. This imparts a whole new dimension to the flexibility of the equipment. New features or measurement techniques can often be achieved by changes in software.

Audio equipment is becoming more complex. It follows that test and measurement equipment must also become more complex, but at the same time, users are demanding simplicity of operation. The solution to this dilemma is the ubiquitous microprocessor. It almost lets the manufacturer build an instrument that is all things to all people. The microprocessor takes incredible levels of complexity in its stride and provides a wealth of features.

The incorporation of a microprocessor lends immense power to the instrument, going beyond even mere simplification of the front panel controls. A requirement of almost all measurement situations is determination of the performance and the accuracy of the test equipment itself. Intelligent test equipment can perform this function as an internal routine function. This can be completely transparent to the user. Future instruments will calibrate themselves. According to a prearranged schedule, the instrument will go through an elaborate sequence of internal measurements, log a whole series of error data and use this data in the computation and display of its final measured results. Such an instrument will require no manual calibration. It will also be possible for such an instrument to completely test its own total internal performance. If any function is inoperative, or out of specification, a display or print-out will advise the operator. This lends a high level of confidence to the final results.

Many measurements that would have been too time-consuming or cumbersome to achieve with conventional instruments will be possible with future microprocessor-based instruments. Data reduction or "number crunching" becomes trivial for such equipment. You do not have to see an endless stream of numbers. You only want to know whether the average is within spec or, what are the values of the highest and the lowest reading. In an R&D application, a graph can show a trend of how one performance parameter varies with another. Generation of such a graph may require hundreds, or thousands, of readings to be taken with a large number of parameters varied in several dimensions. A design engineer would not normally undertake to make all of these measurements. The microprocessor-based instrument may make them with a signal push of a button.

Test equipment has entered a new generation—the generation of the intelligent instrument. We can look forward to higher accuracy, simpler operation, faster results, more-comprehensive presentation of data, less operator-intervention, and more measurement capability, as the intelligence of instruments increases. ■

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TWO GATELY PROKIT SM-6A stereo mixers. Six mike/line inputs with EQ, reverb. **Ken Upham, RD2 Box 95, Stone Ridge, NY 12484 (914) 687-0445.**

ELECTRODYNE CONSOLE. 20 in 16 out. Excellent condition! Stereo echo, patch bay, many extras! $12,500. Must sell immediately. **California Recording, 5203 Sunset Blvd., Hollywood, CA (213) 666-1244.**

TASCAM 80-8 with DX-8 dbx. Four months old. Asking $4,000. **Mike Fuller (215) 687-3731.**

MCI-JH 16-RECORDER, with autolocator, 8-track heads, vari-speed, mint condition, best offer over $12,500.00. 16-channels dbx-157, $2,500.00. Call: **(201) 359-5520.**

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

UNISTAGE invites you to visit the Northeast's newest professional audio display complex. 4000 sq. ft. stock with working P.A. systems featuring selections from over 40 brands of electronics; multi-track decks, mixers, and monitors; custom keyboard and guitar systems; and complete disco sound arrays. Sales, rentals, and service to the professional entertainment industry. **Unistage, Inc., 330 Genesee St., Buffalo, NY 14204 (716) 853-6500.**

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NEUMANN SM69 fet with NS69i, matrix system, and 100 feet of cable; $1500. Two Ampex 440C electronics $500.00 each. One Ampex PR10 ¼-track stereo $350.00. **L. Cooper, Box 453, Shamokin, PA 17872.**

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AMPEX PR10 TRANSPORTS and Ampex AG500 transports and electronics—mono and stereo. All used. For sale. **Call: Tony—(212) 581-5025.**

FOR SALE: ALL EQUIPMENT IN excellent condition. 1-Eventide 1745A DDL, 1-Eventide 1745M DDL, 1-Ampex 300-2 in console, 1-Ampex 300-4SS in console, 1-set 8 track heads for 3M-M79, 1-Ampex 351-2 with Inovonics electronics, 5-OP-AMP labs model SM 100 50 W/Channel. Contact **Frank Tarsia (215) 561-3660.**

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

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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: Immediate sale & occupancy. Partially equipped midtown NYC recording studio. Excellent acoustics and location. Wired for video. **(212) 679-5670, (203) 226-4200.**

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 Rodler Dr., Glendale, CA 91201; 75 Commerce Way, Hackensack, NJ 07601.**

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.**

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.**

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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.**

WANTED

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

EMPLOYMENT

24 TRACK RECORDING STUDIO in the Baltimore-D.C. area seeking qualified engineer with years of experience. $20k to qualified party. **Dept. 61, db Magazine, 1120 Old Country Rd., Plainview, NY 11803.**

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Self-starter to develop both technical and creative copy for pro-audio, Hi-Fi, musical & computer equipment. Relaxed atmosphere in hills, 45 minutes from downtown LA. Writing experience helpful, but we may train the right person. Full time preferred. Send resume.

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People/Places/Happenings

• **Dawn Recording Studios, Inc.** of Farmingdale, NY, home of the **Audio Recording Technology Institute,** has recently relocated into a new studio complex. The new studios feature a custom built Xedit 16-track recorder. API and Gately 16-track consoles, and a Van Epps stereo cutting system for discs. Under construction is Dawn's second 16-track studio.

• Establishing West Coast operations, the **BTX Corporation** of Weston, Massachusetts will be located at 6255 Sunset Blvd., Hollywood. CA. Telephone number (213) 462-1506. **Jerry Hudspeth,** formerly of Pacific Video Corporation, has been named manager for BTX.

• **P & P Studios Inc.,** in Stamford, CT. announce the opening of their new John Storyk-designed 16-24 track studio. The facility also includes: two smaller studios, a soundstage, high-speed cassette and reel-to-reel duplication, film editing and transfer and multi-image programming and projection. P & P Studios has been in business since 1970.

• Named marketing director for **Furman Sound,** San Rafael, CA, **Lary Earl Collins** will be directing management of sales representatives, coordinating and planning for AES and NAMM shows, planning advertising and direct mail campaigns, setting up and managing Furman Sound's new distribution facility, and assisting in setting up the new co-op advertising program that Furman will be offering to its dealers. Mr. Collins was previously vice president of **Rothchild Musical Instruments** of Englewood. New Jersey.

• **Amber Electro Design,** a Canadian based firm. has agreed to allow **Track Audio Inc.,** a U.S. based firm, to continue manufacturing the Model 4550 Spectrum Display. All correspondence relating to this device is now to be directed to Track Audio, Inc. In addition, Track Audio, Inc. will be setting up a distribution network and is open to dealer inquiries. Inquiries are to be sent to: 33753 9th South, Federal Way, WA 98003. Telephone (206) 838-4460.

• **Darmstedter Associates, Inc.,** has been named rep. in the upstate New York area, for **Teac's** Multi-track and Tascam Series. Darmstedter Associates, Inc.. is headquartered in Baldwinsville, New York.

• Undertaking the responsibility for all sales promotion programs in the U.S. consumer market, **Robert B. Kiel** has been appointed sales promotion manager at **BOSE Corporation.** Mr. Kiel has served on the U.S. marketing staff at Bose. since January. 1978.

• Joining **Time and Frequency Technology, Inc., Frank J. Rich** has been appointed to the post of director of marketing for the company's full product line, including digital remote control systems, a.m., f.m., and tv requency and modulation monitors. emergency broadcast systems and studio to transmitter link. Mr. Rich previously held marketing positions with Video Logic and Syntex Laboratories.

• Terminating the operation of its Audio/Electronics Division, in California, **Dictaphone Corporation** is transferring production of its voice communications logging systems to the company's main manufacturing facility in Melbourne, Florida, in July. In addition, Dictaphone has agreed, in principle, to sell its line of **Scully** professional recorders to **Ram Management Corp.,** which also operates **Ampro Broadcasting** of Feasterville, Penn.

• **The Variable Speech Control Company** of San Francisco has delegated the responsibility for the design, manufacture and marketing of VSC-equipped products to the newly formed **VSC Corporation. George Leslie** has been named president of the VSC Corporation, with **Grady Hesters** appointed director of sales and **Linda Jones** responsible for advertising and public relations. VSC Corporation maintains its engineering laboratories, manufacturing center, national service facilities. and warehousing and distribution center at 185 Berry St.. San Francisco, CA 94107 (415) 495-6100.

• **ProTech Audio Corporation,** a newly incorporated audio products manufacturing company, has concluded an agreement with **Robins Industries** for the rights to manufacture all professional audio products formerly produced under the **Robins Broadcast & Sound Equipment Corp.** name. All production, servicing, maintenance, and warranty work will be performed by ProTech, at its Ronkonkoma, New York facility. ProTech is headed by president **William H. Murphey** (formerly production manager of Robins Broadcast & Sound); with **Rick Belmont,** vice president of marketing & sales, and **Joe Giambone,** vice president of engineering.

• Replacing **Larry Cutchens,** who was promoted to sales manager. **Mike Flood** has been appointed technical service coordinator at **International Tapetronics,** Bloomington, Illinois. Mr. Flood previously served as ITC's manufacturing project engineer.

• Appointed vice president, North American manufacturing for **Radio Shack, Robert M. McClure** will oversee operations of the company's 15 manufacturing facilities located in the U.S. and Canada: excluding those plants producing computer-related products. Mr. McClure joined the Radio Shack organization in 1972.

• Appointed national sales manager for **Otari Corporation, Steve Krampf** assumes direct responsibility for the management of Otari's dealers and sales representatives. Most recently, Mr. Krampf has served as vice president at **Express Sound** in Costa Mesa. California. Prior to that he was a representative for **Tascam** and **dbx.**

• In the March issue of **db's** "People, Places, Happenings" we reported on the appointment of **Don Mereen** to the position of marketing director of the Broadcast & Professional Audio Products group at **Telex Communications, Inc.** In an oversite. however. we neglected to mention that Mr. Mereen will also be responsible for the customer service and technical training departments.

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.
- PRECISE CARDIOID polar pattern, uniform with frequency and symmetrical about axis, to provide maximum rejection and minimum coloration of off-axis sounds.
- EXCEPTIONALLY LOW (16 dBA) NOISE LEVEL.
- 120 dB DYNAMIC RANGE.
- ULTRA-LOW DISTORTION (right up to the clipping point!) over the entire audio spectrum for a wide range of load impedances. MAXIMUM SPL BEFORE CLIPPING: 135 dB; 145 dB with attenuator.
- 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)

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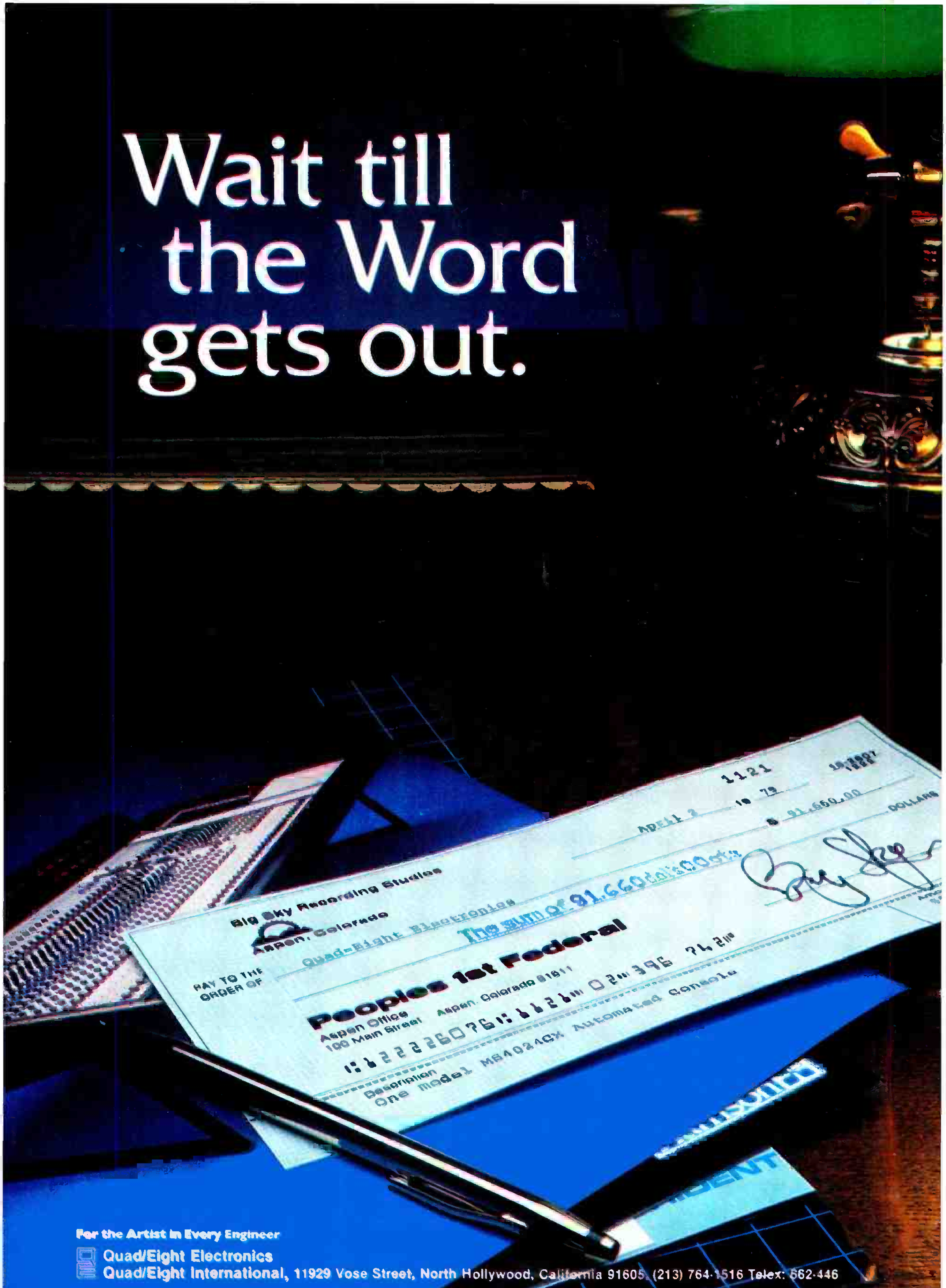

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