SERVING THE CREATIVE AUDIO AND MUSIC ELECTRONICS INDUSTRY

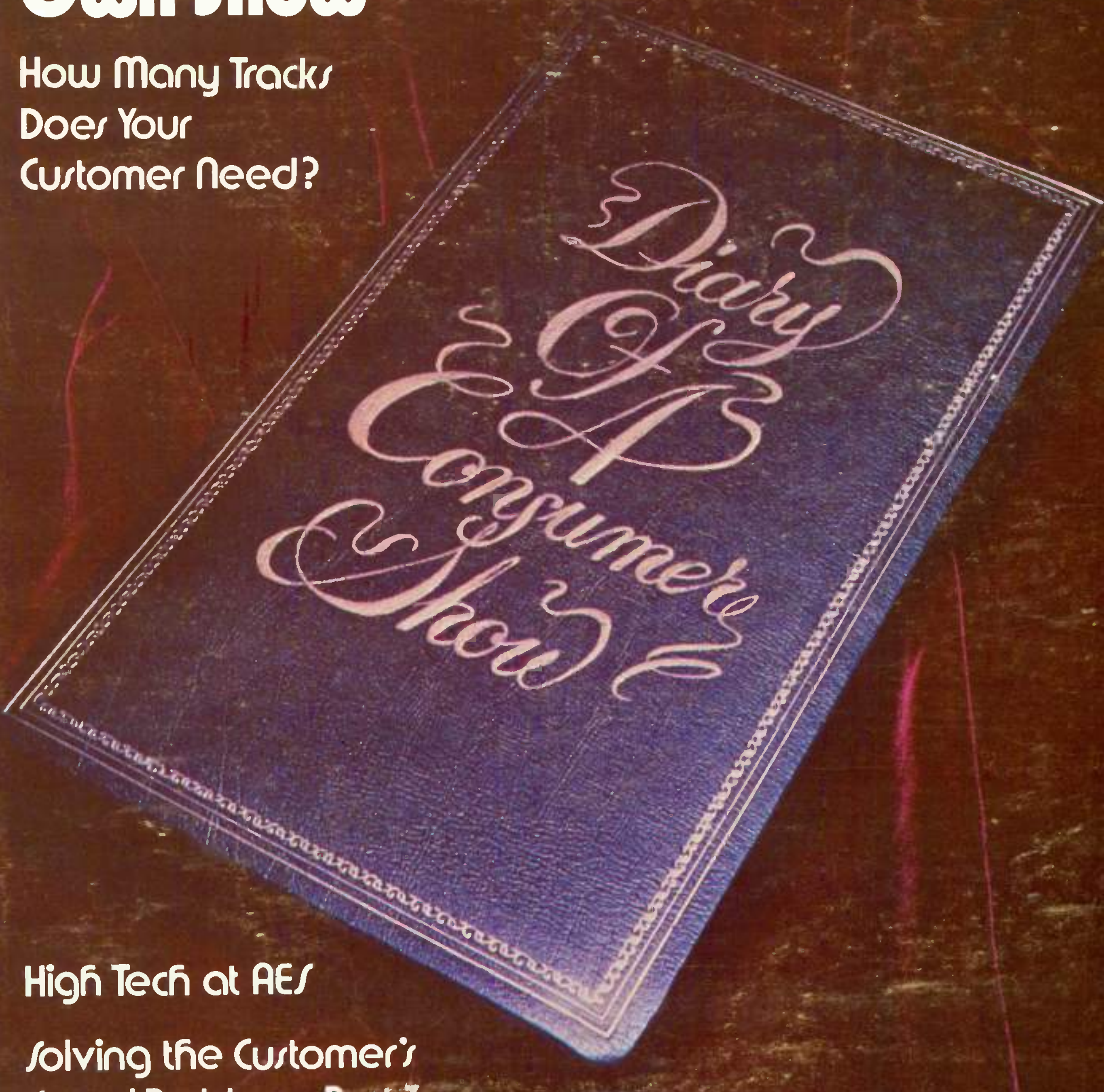
SOUND ARTS

MERCHANDISING JOURNAL

VOL. 1 NO. 12
JANUARY 1979

Setting Up Your Own Show

How Many Tracks Does Your Customer Need?



High Tech at AES

Solving the Customer's Sound Problems, Part 3

THE LONG AND THE SHORT OF SOUND REINFORCEMENT.

You know about the long part. Separate components can keep your hands full, what with the extra help and time needed to get your sound reinforcement act together.

Now for the short part. The Yamaha EM-200 and EM-300 stereo output integrated mixers. They leave you free to concentrate on the creativity of your job, not the mechanics of it.

You get the mixer, power amplifier, 9-band graphic equalizer, echo and reverb control **all in one unit**—great flexibility with options to expand and enlarge.

The EM-200 and EM-300 are ideal for small to medium size reinforcement applications, wherever you need a precisely placed, superbly clean and well-defined sound from a compact source that is easy to set up and operate.

The EM-200 has eight input channels and 120-watt speaker output. The EM-300 has 12 input channels and 200-watt speaker output. For increased flexibility, both the EM-200 and EM-300 have hi and lo impedance monitor output levels (+4dB into 10K ohms, and 0dB into 600 ohms). Additionally, eight patch points allow you to connect accessories directly to the mixer's power amp for dramatically lower noise levels.

The EM-200 and EM-300 give you the short-cut to reinforcement that won't short-change the quality of your sound. They're convenient to set up, operate and locate... at your Yamaha dealer soon.

YAMAHA

P.O. Box 6600, Buena Park, CA 90622

CIRCLE 99 ON READER SERVICE CARD

Listen to the Electro-Voice story. Your customers are.

As a dealer, you should be interested in the Electro-Voice story, because whether you are selling to the music market, the high fidelity market, the commercial market or the professional market, Electro-Voice is the leader.

The music that your customers listen to at home was probably recorded using Electro-Voice professional microphones and mixed using E-V Sentry® studio monitors. Is it any wonder that E-V Interface® high fidelity speaker systems are rated among the finest for home systems?

If music is your business, it's good to know that the famous EVM loudspeakers are not only standard in many manufacturers' "premier" lines of enclosures, but are the replacement speakers of choice by many concert sound men. These same speakers are standard in *every* Electro-Voice music speaker product. And E-V microphones are seen being used by more vocalists and instrumentalists on stage than ever before.

Commercial Sound? Think of installations like the Pontiac Silverdome, Yankee Stadium and the Las Vegas Convention Center. They're all Electro-Voice. No wonder so many contractors turn to Electro-Voice sound systems for their church, gymnasium and office building contract-sound installations.

If your business is selling sound, Electro-Voice has a story to tell! A story your customers will want to hear. A story that will make a lot of profitable sales for you. To hear the Electro-Voice sales story in person, contact Dave Rothfeld, General Sales Manager, Electro-Voice, Inc., 600 Cecil Street, Buchanan, MI 49107. Phone 616/695-6831.

EV Electro-Voice®
a gulton company
600 Cecil Street, Buchanan, Michigan 49107

CIRCLE 92 ON READER SERVICE CARD

ONE OF TEAC'S BEST IDEAS WASN'T A TAPE RECORDER.

It was a market.

1970	First 4-Channel Tape Recorder with Sync for less than $1,000 (TCA-40)
1972	First Mass Produced 4-Channel Tape Recorders with Sync. (A-3340 & A-2340)
	First Studio Quality 8x4 Audio Mixer for under $2,000. (Model 10)
1973	First Studio Quality Recorder/Reproducer to provide 8-Tracks on Half-inch tape. (Series 70)
1974	First Mass Produced 6x4 Audio Mixer for less than $300. (Model 2)
1975	First Studio Quality Mass Produced 8x4 Audio Mixer. (Model 5)
1976	First Studio Quality Mass Produced 8-Track Recorder/Reproducer for less than $3,000. (Model 80-8)
1977	First Studio Quality Mass Produced 16-Track Recorder/Reproducer to use One-inch Tape and cost less than $16,000. (Model 90-16)
1979	

TEAC®
First. Because they last.

When we introduced our first multitrack tape recorder in 1970, we were so far ahead of everyone else that many people thought it was a quad machine.

But the customers we built it for knew exactly what it was: a four-channel tape recorder with sync for overdubbing that cost less than $1,000.

Since then, TEAC has continued to develop new products with price/performance breakthroughs as big as the market we discovered.

In fact, multitrack products—including our TASCAM Series—make up one of the most innovative and successful lines in the history of this business judging by the number of dealers who became wealthy selling it, and the number of competitors it spawned.

Getting to the market meant breaking a lot of rules and killing a lot of sacred cows. We put eight tracks on half-inch tape, for one example. And 16 tracks on one-inch tape, for another.

Doing things like that required a profound understanding of our customers' needs and their goals. But still it wasn't easy. We invested huge amounts of time, money and manpower to develop the market.

We created software to demonstrate how the products work. We developed consumer awareness through innovative sales programs using the latest videotape techniques. We even made consumer sales easier through our unique Finance America credit program.

Today, the power of the market is being recognized at last.

As profitable as it is, though, the business may not be for every dealer. It requires commitment, skill and imagination. It could be *your* best idea.

TASCAM SERIES BY TEAC®
A new generation of recording instruments for a new generation of recording artists.

TEAC Corporation of America • 7733 Telegraph Road • Montebello, California 90640 • In Canada TEAC is distributed by White Electronic Development Corporation (1966) Ltd.

SERVING THE CREATIVE AUDIO AND MUSIC ELECTRONICS INDUSTRY

VOL. 1 NO. 12

SOUND ARTS

MERCHANDISING JOURNAL

JANUARY 1979

CONTENTS

THE FEATURES

THE STAPLES

COMING NEXT ISSUE!

First Anniversary Issue—The State of the Market One Year Later
Digital Synthesis: How It Works
Detailing Retail Advertising

Cover Photo by Doug Hanewinckel

Sound Arts Merchandising Journal is published monthly by Cowan Publishing Corp., 14 Vanderventer Ave., Port Washington, N.Y. 11050. Design and contents are copyright by Sound Arts Merchandising, Inc. and must not be reproduced in any manner except by permission of the publisher. Controlled circulation postage paid at Hanover, New Hampshire 03755. Subscription rates for other than qualified individuals or companies: $12.00 for 12 issues; $22.00 for 24 issues. Add $3.00 per year for subscriptions outside of U.S. Subscriptions must be paid in American currency. Postmaster: Send Form 3579 to Sound Arts Merchandising Journal, 14 Vanderventer Ave., Port Washington, N.Y. 11050.

SERVING THE CREATIVE AUDIO AND MUSIC ELECTRONICS INDUSTRY
SOUND ARTS
MERCHANDISING JOURNAL

JUDITH MORRISON LIPTON
Editor

AUDREY KURLAND
Assistant Editor

CRAIG ANDERTON
MIKE BEIGEL
LARRY BLAKELY
GARRETT DONALDSON
CLAIRE M. FORD
CHARLIE LAWING
GLEN E. MEYER
RALPH MORRIS
Contributors

MELANIE DEUTSCH
Assistant to the Publisher

LORI RESSA
Production Manager

KIM SMITH
Production Assistant

BILL TRAVIS
Art Director

SHERYL STERN
Art & Design

LIZ BEENER
HAROLD PERRY
BARRY SIMON
Art Staff

JANET KURTZ
Circulation Manager

BILL SLAPIN & CO.
West Coast
Advertising Representative

MYLES GROSSMAN
Advertising Director

VINCENT P. TESTA
Publisher

Editorial and Executive Offices
Sound Arts Merchandising Journal
14 Vanderventer Ave.
Port Washington, N.Y. 11050
516-883-5705

COWAN PUBLISHING CORP.
RICHARD A. COWAN Chairman of the Board & President
CARY L. COWAN Vice President
JACK N. SCHNEIDER Vice President, Marketing
RICHARD A. ROSS Vice President, General Manager
MARC L. GILMAN Credit Manager
AMY C. GILMAN Secretary/Treasurer
SANFORD R. COWAN Founder & President Emeritus

Editorial contributions should be addressed to The Editor, Sound Arts, 14 Vanderventer Ave., Port Washington, N.Y. 11050. Unsolicited manuscripts will be treated with care and must be accompanied by return postage.

A LETTER FROM THE EDITOR

Feet, don't fail me now. At press time, our receipt of pre-show press releases indicates zooming technology and a plethora of equipment to see at both CES and NAMM. My feet have been prepped (my mind already is) and SOUND ARTS will supplement them by having roving reporters at both shows.

Of equal importance to me personally during this month is the opportunity to meet and re-meet many of our readers personally. It gives me a chance to get in on the thoughts out there on the retail selling floor—and allows us to keep SOUND ARTS on target. The calls and letters during non-show months are interesting and helpful, but face-to-face always beats voice-to-voice.

Several calls that I've received recently have been of special interest to me—and perhaps to you too. Quite a few retailers have called to tell us that they have been using SOUND ARTS in their internal sales training sessions.

Admittedly, when this magazine was inaugurated, we didn't have this kind of structured purpose in mind. We merely wanted to help the retailer get the information he could use to carry on—a large calling in any case. The fact that some retailers have seen fit to do point-by-point sessions using our material has been an added ego high for us and reinforces our direction.

Which brings me to the feature I consider of major importance to this magazine — Common Consumer Cuestions.

Common Consumer Cuestions is a multi-faceted feature. It can help you to help your customer with information as basic as how to read a spec, or as complex as how to design a sound system. Our questions come from several sources—from retailers themselves, from manufacturers' customer service departments, and even from an occasional consumer who has somewhere seen SOUND ARTS.

The answers too come from retailers, manufacturers and other experts in the field. The more input we have in Common Consumer Cuestions, the more we fulfill our purpose as an intra-industry communications medium. Send us the questions you're finding yourself fielding—either those that you find most frequently, or those that are of some difficulty to answer. Send your own answers if you have the time. Or just send us the questions; we'll get the answers. CCC is a linchpin of the magazine as a whole and your participation will keep its quality high.

Regards,

Judy Lipton

Judith Morrison Lipton

Meet the AKG "C-450 Microphone System"

At our place, there is an interesting credo ..."When perfection is attained, improve upon it."

Eight years ago, AKG introduced the *first* modular concept in professional microphone design. It was *then*, and is *now* the only true condenser modular system to provide for a myriad of applications through interchangeable components...much the same as cameras with quality interchangeable optics. Since the inception of the system, each component has been continuously upgraded and refined to the ultimate in technical perfection...while new ones have been added to form the broadest in-depth product range suitable for any known application.

Matching the widest possible scope of end-use, the C-450 Condenser System presently offers: *four* cardioid capsules, *four* different omni capsules, *one* figure-eight capsule, *two* shotgun capsules, *five* preamplifiers and *seven* powering options. Added to this is an assortment of swivels, pads, extension tubes, shock mounts and windscreens. These quick-change "screw-on" components provide an almost limitless variety of combinations to extend the flexibility and capabilities of the microphone as changing uses in audio applications require.

Returning to our credo...we are constantly searching, and as applications requirements, materials and techniques are discovered to further improve upon the system components, they will first be found bearing the trademark AKG.

AKG acoustics

The Mark of Professional Quality... in microphones, headphones, phonocartridges, reverb units.

PHILIPS AUDIO VIDEO SYSTEMS CORP.
A NORTH AMERICAN PHILIPS COMPANY
91 McKee Drive, Mahwah, N.J. 07430 • (201) 529-3800

CIRCLE 84 ON READER SERVICE CARD

FORUM

Just finished reading "Impedance Matching" by Irwin Zucker in the November issue. Pretty good article, not the usual "How to Make Big Bucks on Merchandise with Little or No Margin" (which seems to be the dream of this industry). It's good to see a down-to-earth, specific, and technical article in your magazine.

But I wanted to call your attention to an error in figure 5 [page 33]. The connection for unbalanced line will short out the whole audio industry and cause sound systems across the country to go dead! The jumper wire should be connected between pin 1 and pin 2 (not pin 3, which is hot). I have found the standard to be pins 1 and 2, not 1 and 3. Check with Tapco, Peavey, Sunn, Biamp, etc.

We enjoy your magazine, and please feel free to call me any time for comments or information. I'll be glad to share what I know.

Sincerely,
Gary Gand
Gary Gand Music
Highland Park, Illinois

Irwin Zucker replies: *It seems that both Gary and I are partly right. (Or, if you pessimistically prefer, partly wrong). I rechecked the EIA, DIN, and JIS standards concerning microphone wiring with 3 pin XL connector lines. They all agree, pin 2 is high (hot) in a balanced line.*

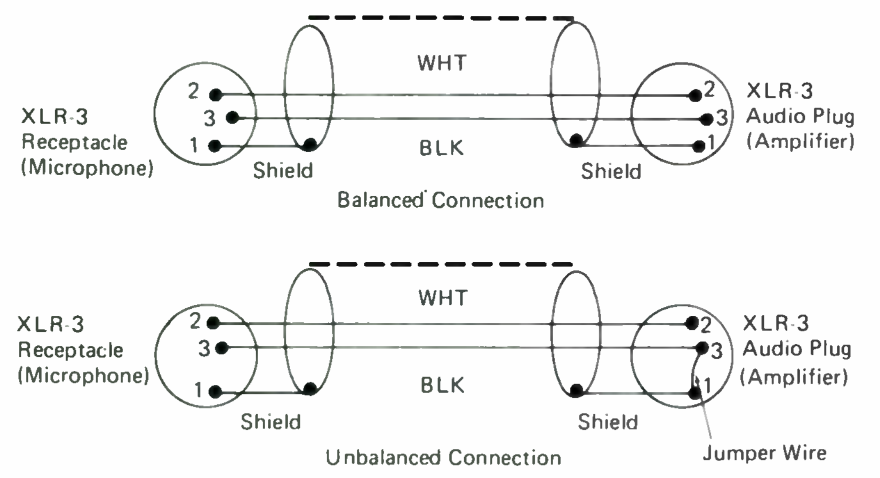
Altec Lansing used pin 3 as high for years, but standardized on pin 2 high a while back to conform to the international standards. In checking with other manufacturers we found that Tapco uses pin 2 high for balanced, but says to tie pin 2 to ground for unbalanced. Uni-Sync shows pin 3 high, and Yamaha shows pin 2 as high, with a "phase reverse" switch on their PM-1000 models to make pin 3 high. Yamaha is evidently aware of the confusion that exists on this matter. Virtually every microphone manufacturer wires for pin 2 high.

It seems the old U.S.A. is out of step Everywhere else, pin 2 is high. We are apparently resisting the metric system. But Gary is right in this respect: Many current manufacturers specify pin 3 as high.

Gary is incorrect on two points: 1. Pin 2 high is the standard, not pin 3 high. 2. Connection as shown in figure 5 of "Impedance Matching" will not ". . . short out the whole audio industry . . ." The only way the signal will be shorted out is if both pins 2 and 3 are grounded. While it may be assumed that any mixer would be consistently wired within itself from input to input, and since virtually all mics are wired the same, the place to be careful is in wiring mic cables. If all cables in a system are wired the same way, no harm done. But when pin 2 high cables and pin 3 high cables are mixed, a phase reversal (not a short) occurs in the mixing buss, causing frequency cancellation. The system will still work, just not as well.

In conclusion, Gary's letter has brought to light three facts: 1. There is an urgent need for compliance with established standards by all affected manufacturers (and the dissemination of knowledge of the standard to all concerned). 2. SOUND ARTS *is an excellent forum for bringing out such inconsistencies. Keep up the good work. 3. There are some live ones out there who take a vital interest in what they do and what they read. Keep those cards and letters comin' in, Gary.*

Good listening,
Irwin Zucker
Director Product Development
Altec Sound Products



CONNECTIONS FOR BALANCED vs UNBALANCED LINES

Figure 5

First of all, let me compliment you on your fine journal. It fills a gap that is so often neglected in trade publications—that is, up-to-date knowledge. So often I read articles which are supposedly "in-depth" which turn out to be nothing more than long advertisements for a particular product. It's great to be getting information from a publication such as yours. Thank you.

In your September, 1978 issue, you stated that back issues are available to persons in the trade. Would you please send two sets of issues (Vol. 1, Nos. 1 through 7) for reference use in our store?

Thanks again for a reliable source of information.

Sincerely,
Bill Adams
Simon Music Center
Moline, Illinois

TERMS:

A CONTINUING INDUSTRY GLOSSARY

RECORDING

By Larry Blakely

Pre-Fader: Refers to the electrical audio signal just prior to the fader or level control. In other words, the signal just before it gets to the fader or *pre*-fader.

Post-Fader: Refers to the electrical audio signal just after it has left the fader or level control. In other words, the signal just after it leaves the fader is *post*-fader.

Pre- and post-fader modes often appear on recording consoles. The most common feature utilizing pre- and post-fader is in the echo send. The echo send control adjusts the amount of signal level that is sent to an echo chamber or echo device. There will often be a switch above or below the echo send control that will be labeled pre/post. This pre/post switch allows the signal being fed to the echo send pot to come from either before or after the fader. An echo send signal that is taken post fader will increase and decrease proportionately with the level of the fader.

For example: You have added echo to a vocal solo and you have just the right amount of echo with the vocal. If you have the echo send signal derived from post-fader, when you increase or decrease the level of the vocal with the fader, the echo send level will change in direct proportion. This means that this vocal will always have that same vocal versus echo proportion regardless of the fader level setting. With the echo send signal being derived from pre-fader, the echo signal level will not change in relation to the fader level. In other words, if you have adjusted for the proper amount of echo and vocal, and you later reduce the level of the fader, the vocal level will decrease, and the echo level will not. Now you have more echo on the vocal than you had previously. However, this can prove quite handy if you want a simple means to drop a vocal back into a lot of echo for special effects in your music recording.

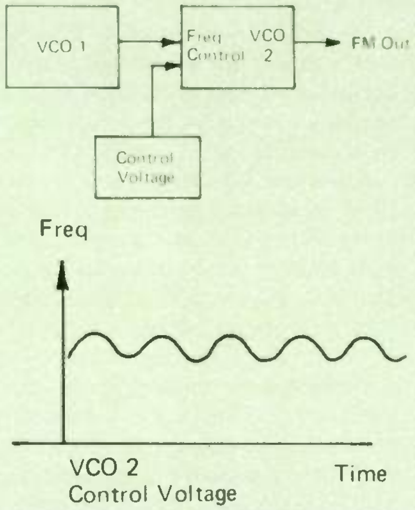
ELECTRONIC MUSICAL INSTRUMENTS & ACCESSORIES

By Mike Beigel

Synthesizer Terms (Continued):

"Sync'ed" Oscillators: Normally there are slight tuning or sweep variations between voltage-controlled oscillators. This results in "beat frequencies" and perceivable pitch variations, even when the oscillators are tuned in "unison." (This is often desirable for natural-sounding tones.) Some synthesizers have a feature which synchronizes an oscillator to the exact pitch of another oscillator, thereby eliminating beat frequencies. The second oscillator can also be tuned to a harmonic multiple of the first and "sync'ed" to that harmonic. A "sync'ed" oscillator is dependent on the oscillator it is "sync'ed" to, and is independently tuneable only to harmonic multiples of the master oscillator frequency.

Frequency Modulation (FM): If a voltage-controlled oscillator is connected as a control input to another voltage-controlled oscillator, the second oscillator's frequency will vary, depending on the amplitude, frequency and waveform of the first.



For sub-audio frequencies (of VCO 1) the result is simply a vibrato. As the frequency of Osc 1 increases, the output of VCO 2 becomes increasingly complex, producing many extra frequency components.

SOUND REINFORCEMENT

By Glen E. Meyer

Level Variations With Distance (Reverberant Environments): Where sound is reflected from walls and other surfaces and the environment is reverberant, there is a point at which the direct sound and the reverberant sound are both at the same level. This point is known as the "critical distance." Beyond this point the reverberant field dominates and the sound pressure level remains nearly constant. The methods of calculating this distance will be discussed at another time. The distance, however, is typically 10 to 30 feet from the speaker, and is longest for the least reverberant rooms and the most directional speakers. Because of the reverberant field, the sound pressure level obtainable in a room over any reasonably wide band of frequencies is much higher and more constant than that predicted by the inverse square law alone. However, for proper sound system design, the frequency response and dispersion information is still necessary in order to obtain satisfactory distribution of the sound direct from the loudspeaker (direct field), which still follows the inverse square law.

In installations where applicable, it is good practice to make sure that the direct field is no more than 12 dB below the reverberant field to ensure satisfactory intelligibility.

Level Variations Due to Multiple Speakers: The net effect on the total sound pressure level in an environment where two or more speakers are operating, with the same program at the same time, is fairly complex. However, some useful guidelines are possible. When there are two widely spaced, broadband sources of equal intensity, the sound pressure level of the combination will be 3 dB higher than that of either source alone. When the two signals differ by any substantial amount, like 9 or 10 dB, the combined sound pressure level is only slightly higher than that of the loudest source alone.

TEAR HERE

TERMS: (CONTINUED)

A CONTINUING INDUSTRY GLOSSARY

RECORDING

Sometimes recording consoles have pre/post switches with the cue send (phones). The cue system allows a musician to have a mix (musical balance) on his headphones in the studio, and yet allows another (completely different) mix to be utilized in the control room. When the cue mix is derived post fader a change in the level of a fader in the control room *will* change the level in the cue send as well. However, with the cue signal derived from pre fader, level changes in the control room will not affect the cue send signal whatsoever. In short, if you want the same musical balance in the control room and on the cue system (phones), utilize post-fader cue send. If you want different mixes in the control room and cue system, the cue send signal should be derived pre fader, thereby allowing two separate mixes of the same signal where changes in one do not affect the other.

The terms *pre* and *post* do not always refer to before or after a fader. Pre and post are used in other places as well. For example: Pre-equalizer simply means before the equalizer, and post-equalizer means after the equalizer. And as you can imagine there is a before and after with most anything. So pre and post it instead.

Three-Knob Equalizer: Means that there are three frequencies at which the equalizer can work (*i.e.*, boost or cut) at the same time. In other words, a typical three-knob equalizer would work at low frequencies, mid frequencies, and high frequencies. Usually, three-knob equalizers are of the selectable frequency type (*i.e.*, having selectable frequencies available for each of the low, mid, and high frequency sections). Still other types of three-knob equalizers may have fixed frequencies on the low and high with a selectable midrange section. If such a three-knob equalizer had an available selection of 12 frequencies (*e.g.*, four selectable frequencies for each knob), it might be referred to as a 12-frequency three-knob equalizer.

ELECTRONIC MUSICAL INSTRUMENTS & ACCESSORIES

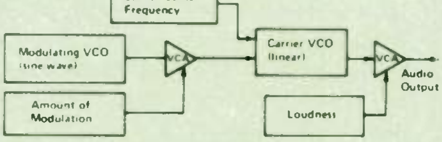
FM Synthesis: A technique for synthesizing musically interesting tones, discovered by John M. Chowning. Instead of generating sounds by means of a voltage-controlled oscillator, envelope generator and filter system; Chowning's method uses a unique property of frequency modulation theory. A pair of (digital) sinewave oscillators interact to create tones with dynamically changing audio spectra. The FM soudns seem to sound more "alive" and natural than conventional synthesized sounds. (For further information on FM Synthesis and the terms following — through *Dynamic Spectra* — see John M. Chowning's paper "The Synthesis of Complex Audio Spectra by Means of Frequency Modulation," *Journal of the Audio Engineering Society*, Volume 21, Number 7, September 1973.)

Linear FM Oscillator: A voltage controlled oscillator in which the output frequency is a *linear* function of the control voltage, instead of an exponential (volts-per-octave) function. This type of oscillator is preferable for FM synthesis.

Digital FM Oscillator: A "computerized" model of a voltage-controlled oscillator. It has the unique ability to produce a negative frequency when it receives a negative "control voltage."

Reflected Side Frequencies: "Negative" frequencies produced by the digital oscillator actually "reflect" around 0 Hz and appear in the oscillator's output 180 degrees out of phase in the positive frequency domain.

Modulating Oscillator: The oscillator which FM modulates the carrier oscillator. This oscillator's output controls the characteristics of the audio spectrum associated with the carrier oscillator's output. (See diagram.)



SOUND REINFORCEMENT

Please note table below.

When Two Signals Differ By:	Add to the Larger Reading:
0 dB	3.00 dB
1	2.50
2	2.10
3	1.70
4	1.40
5	1.10
6	.97
7	.79
8	63
9	51
10	43
11	35
12	26

Combined Level Two Sound Sources

Impedance: The complex sum of resistance and reactance of a circuit that opposes the flow of alternating current.

Impedance Matching: In connecting a speaker to an amplifier, the idea that impedances must match is somewhat fallacious. Actually, the typical speaker load "seen" by the amplifier (like 8 ohms) is very much larger than the impedance "looking into" the amplifier's output terminals (less than 0.5 ohms). Matching is required, however, in the sense that the speaker load impedance must be suited to the amplifier: Amplifiers, with or without output transformers, are designed to deliver rated power at rated distortion into the rated load impedance. Thus, in multiple speaker designs, the combined speaker impedance should be calculated and "matched" within reasonable limits to the amplifier's rated load impedance.

Impedances higher than rated load will reduce the power delivered to the speakers, although distortion and other performance characteristics will be essentially unchanged. Impedances substantially lower than rated loads should be avoided. Power at rated distortion will be reduced and damage to amplifier output stages or activation of protection circuitry may result.

TROUBLESHOOTERS' BULLETIN

1

RECORDING MALFUNCTION CHECKLIST

During a recording session, your customer may have a situation where he doesn't get any sound. Don't immediately assume the equipment is at fault. You can give him a quick checklist of possible causes:

1. Is the power on?
2. Is the mic switched on?
3. Are the mic leads okay?
4. Is the channel trim turned up?
5. Are the channel faders, submaster faders and main output faders turned up?

2

6. Are the monitor speakers on and turned up?
7. Are you switched to the correct mic/line position?
8. Have you checked the position of the monitoring select switch?
9. Are all the interconnecting leads firmly in place?
10. Were you in record during the recording? (It happens!)
11. Were you switched to the ready position?

TEAC CORPORATION

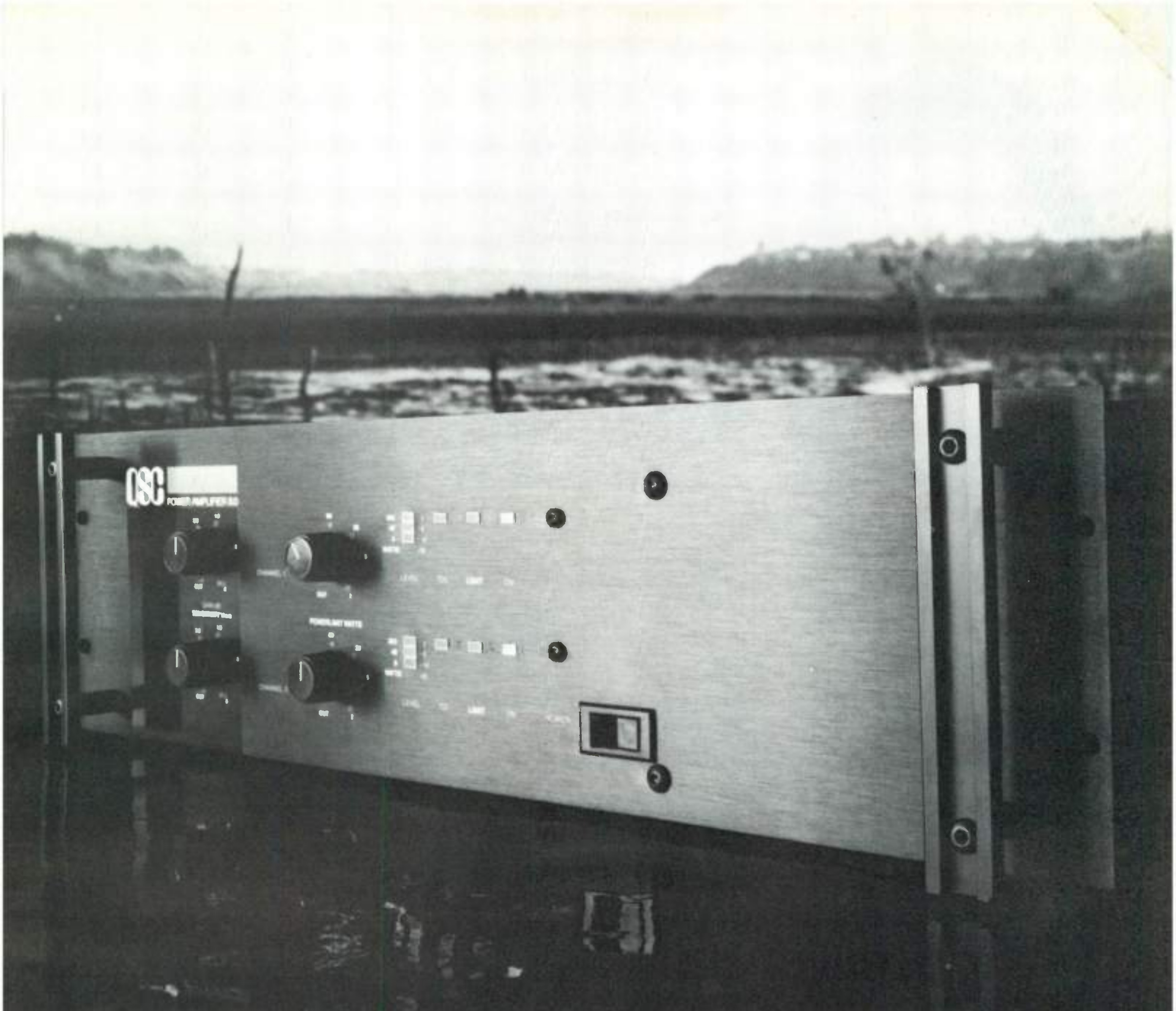
3

MORE REMINDERS FOR YOUR LOUDSPEAKER CUSTOMERS

1. Never hook up a speaker when the amplifier is on and being driven by a signal.

2. Never turn on low level electronics (mixer, graph, etc.) after the power amplifiers are on.

3. Always use a DC blocking capacitor on high frequency compression drivers when bi-



IF YOU'RE A PRO-AUDIO DEALER YOU CAN'T AFFORD TO MAKE EXCUSES ABOUT THE EQUIPMENT YOU SELL. YOUR CUSTOMER'S LIVELIHOOD DEPENDS ON IT.

Dealers kept asking us "How about a high-power amp with low distortion that's loaded with options and has an exciting list price?" We listened and set out to build "The Complete Amp" with reliability, power, specs, features, price and profitable margins. We've succeeded. Our reputation has been built on the design and construction of cost-effective gear combining maximum performance with simplicity and reliability. Now QSC offers a package you can't find in any other amp, REGARDLESS OF PRICE OR OPTIONS. The A 8.0 delivers 300 watts of clean power to each channel (20-16 kHz with less than .09% THD rising gradually to 0 2% THD at 20 kHz into 4 ohms) and 600 watts into 8 ohms with the same specs in the bridged-mono operation.

Features include: Power Limit Controls; Fan Cooling; 3-way Load Protection; LED displays for level, distortion and limiting indicators; Balanced Imputs with XLR type 3-pin connectors; and Outputs with 5-way binding posts, phone jacks and speaker protection fuses.

As a professional audio dealer you can't afford to make excuses. At QSC we make no excuses, period.

QSC
AUDIO PRODUCTS
1926 Placentia Avenue
Costa Mesa, CA 92627
714-645-2540

CIRCLE 91 ON READER SERVICE CARD

So... u want to know about Effects Part III

By Craig Anderton

Last month, we looked at echo devices based on tape recorder technology. This month, we're turning our attention to the new breed of solid state echo and delay units.

Until quite recently, the idea of taking a musical signal and delaying it with respect to time was impossible to achieve inexpensively; luckily, two major technological events have turned this situation around. The *bucket brigade device* (BBD for short) is an electronic part that was specifically invented to create time delays, and has only come out of the lab and into commercial applications within the past few years. BBDs are *analog* delay lines. Simultaneously, the development of microcomputers has created a whole family of inexpensive memory and storage components in *chip* form. By storing a signal in these memory chips, and then retrieving the signal at a later time, you've created a time delay. This *digital* delay line approach has been around for some time, but the transition from expensive military/industrial applications to inexpensive consumer applications was made possible only by substantial price reductions in the cost of this type of technology. This decrease in cost has been comparable to the decreases in cost that have occurred with calculators and digital watches.

In this column we're going to get a little technical, because I'd like to give you a feel for the mechanics of how time delays are generated electronically. Next month, we'll get applications-oriented and point out features and characteristics of different types of solid-state echo units.

THE BBD

The BBD is an example of analog technology; in analog delay circuits, the signal to be delayed is processed directly by the delay line. Referring to figure 1, let's say you have a simple, sine-wave-like signal (although other signals are also acceptable). The delay line is shown as having eight stages, but this is a drastic oversimplification; most BBDs have between 1,000 and 4,000 stages. In addition to the BBD itself, there is a *clock* whose function as a timing reference will become clear as we go along.

The clock operates at a very high frequency (usually above the range of human hearing). Most people think of clocks as going "tick-tock," and really, this one is no exception; however, instead of making audible clicks, the

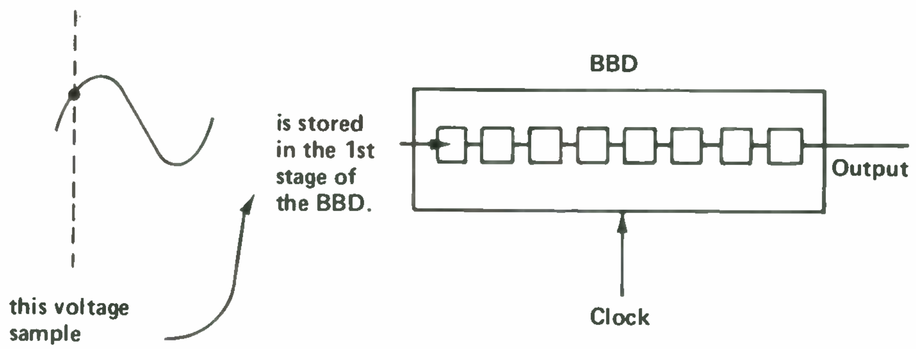


Figure 1

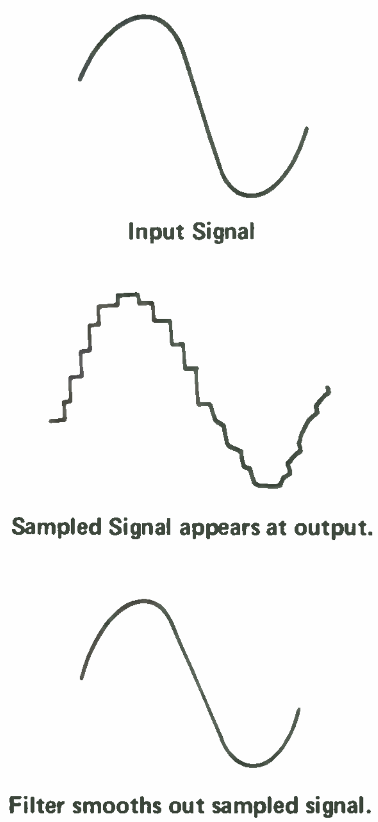


Figure 2

COMMUNITY
The best sound equipment there is. Because from conception to completion we make no compromises.

We start with a job, not a market.

Every Community horn, every cabinet was designed because there was a need for it, a particular professional application that no other product could fill. So that whether you are considering rock and roll for the masses or one voice speaking to one ear, Community has you covered.

We engineer the physical design that precisely fills that requirement and construct it exactly as laws of physics dictate it must be built.

If we can't do it correctly the first time around we work until we find a way. We won't bend it, we won't shorten it, because we found out a long time ago that you can't cheat physics. We don't use inferior materials because the result is always an inferior product. No gingerbread, no inadequacy. Just simple, straight-forward designs with the built-in toughness and strength that have become the hallmark of every Community product.

We give our products the best warranty in professional sound.

We provide the most accurate, usable technical data available on horn-loudspeaker performance in the industry.

No tricks, no inconsistencies, just factual information on what you can expect from us.

We distribute through the most knowledgeable, reliable professional retailers in the country.

Not every big city can boast a Community dealer, but a lot of small towns can, because we go where the talent is.

In other words, we give our best so that you can give your best. We take pride in what we do and we sincerely believe that what we do is build superior pro sound equipment.

Try us. Our business is professional sound.

Community

COMMUNITY LIGHT & SOUND, INC., 5701 Grays Avenue, Philadelphia, PA 19143 • (215) 727-0900

CIRCLE 85 ON READER SERVICE CARD

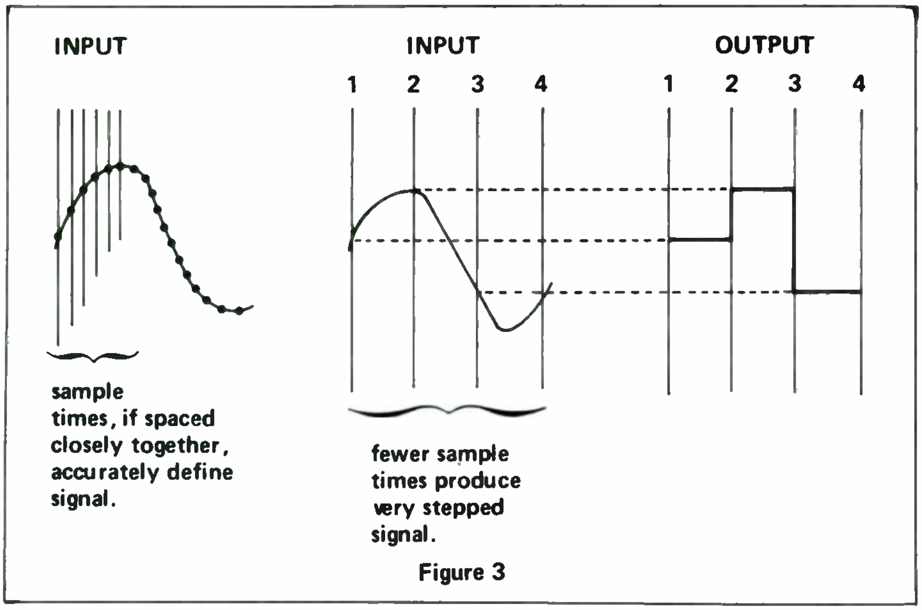


Figure 3

clock output swings between two possible voltage levels.

During the "tick" of the clock cycle, the bucket brigade device "looks" at the input signal for a tiny segment of time, indicated by a dot on the figure of the original signal. Since the sampling time is so small, the signal appears not as a complex waveform, but as a specific voltage. The value of this voltage is momentarily stored in the first stage of the BBD.

Next, during the "tock" of the clock cycle, the bucket brigade stops looking at the signal, and instead transfers the signal from the first stage into the second stage.

During the next "tick" of the clock, the BBD looks again at the signal, takes a second sample, and transfers it to stage 1. During the "tock," this sample moves into the second stage, and the sample that was in the second stage moves into the third stage.

By now you're probably getting the picture: The original signal is broken down into a series of samples, and these move down along the delay line. The more stages, the more delay, and by the time you've clocked the signal down a couple thousand stages you've built up quite a lot of delay.

The output of the delay line looks like a bunch of little sampled voltages; adding some filtering at the output smooths this stepped signal into a signal that more closely resembles the original. (See figure 2.)

As you might imagine, there are some problems with a scheme that has such a Rube Goldberg element to it. First of all, the delay is dependent on the speed of the clock. A faster clock will rapidly move samples down the bucket brigade, resulting in short time delays. Slowing down the clock slows down the rate of transfer, thus creating longer delays. However, if you clock too slowly, there are two problems: You may hear the clock as an audible tone; and you may not be sampling the signal enough to maintain reasonable fidelity. For example, in figure 3 we see a signal that's sampled enough to give a really close approximation of the original waveform; but look at the undersampled signal . . . it hardly resembles the original signal at all, which means far lower fidelity. Additionally, the clock signal itself becomes part of the waveform, and must be removed at the output. Finally, there are certain kinds of losses as the signal moves down the delay line; in other words, the more buckets, the more likely the chance that what comes out is going to be different from what went in (noisier, more distorted, and the like). Despite these disadvantages, by keeping the clock speed fairly fast to make an adequate sample of the input signal, and by using good design techniques, an analog delay line will do an acceptable job. As a bonus, it is relatively inexpensive.

DIGITAL DELAY

Digital delay is similar, but differs in one very important respect. Referring to figure 4, an input signal is again sampled, but this time the voltage sample is converted into a digital code that *represents* that voltage (the exact type of code varies). This digital code is then either passed along the digital equivalent of a bucket brigade (called digital shift registers), or is stored in a computer-like memory and recalled at a later time (also giving delay). At the output, this coded signal goes through a decoder to restore the code back to an audio signal.

This whole "coding" process may seem more complex and expensive than simply working on the signal with analog techniques . . . and it is. But there is a very important advantage to digital technology. Digital codes are formed of *bits* of information (as letters are *bits* of information that make up words), and these bits are represented by either a *presence* of voltage or an *absence* of voltage. When the digital code hits the decoder at the end of a delay chain, that decoder only cares whether a voltage is present or not; it doesn't care whether there is hum or noise added along with that voltage. So, any noise picked up during the process of delay is ignored by the decoder . . . it simply takes the code and decodes it back into a noiseless (almost) audio signal again.

As a result, the only times noises and distortion are introduced is during the coding and decoding process, and by using fairly sophisticated circuitry these problems can be held to a minimum.

If this still doesn't make sense, consider the photos that were sent to the earth from Mars. Had the scientists simply tried to send the photos as a regular TV signal, the picture would have been pretty poor due to the vast distances. Instead, each photo was broken down into a number of areas,

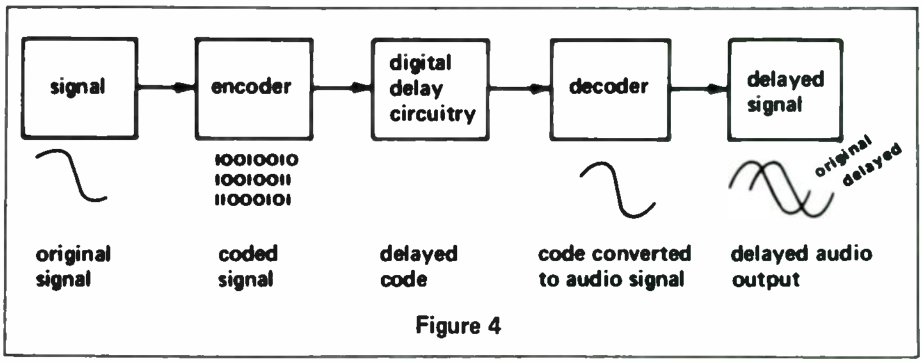


Figure 4

Hot sound vs. hot air.

Lathes–Disk Cutting	Microphone Mixers	Microphones	Mixer/Consoles–Portable	Noise Reducers	Open Reel Recorders 1
Scully 31.1%	Shure 45.6%	Neumann 18.3%	Tascam 16.4%	dbx 48.8%	Ampex 27.8%
Neumann 29.1%	Ampex 11.4%	Shure 17.5%	Teac 9.6%	Dolby 47.8%	Scully 17.5%
Presto 18.6%	Tapco 4.5%	AKG 15.4%	Shure 8.9%	Burwen 6%	3M 7.4%
Westrex 4.7%	Custom 4.2%	Electro Voice 14.6%	Custom 7.7%	Kepex 6%	MCI 6.8%
Fairchild 4.3%	Sony 3.6%	Sennheiser 8.7%	Ampex 5.0%	Other 2.2%	Teac 6.7%
Rek O Kut 4.0%	Teac 3.6%	Sony 7.9%	Sony 2.8%		Tascam 6.0%
Other 8.2%	Altec 2.4%	Beyer 5.1%	Tapco 2.8%		Other 27.8%
	Voice Mix 2.4%	RCA 4.0%	Interface 1.4%		
	Tascam 2.2%	Altec 1.1%	Yamaha 1.2%		1) Fewer than 16 tracks.
	Other 20.1%	Other 7.4%	Other 44.2%		

U.S. EQUIPMENT BRAND USAGE SURVEY

It's the sound!

Open Reel Recorders 2	Phono Cartridges	Speakers–Monitor	Synthesizers	Turntables	Video Tape Recorders
MCI 36.4%	Shure 49%	JBL 34.4%	ARP 43.4%	Technics 27.2%	Sony 56.9%
Ampex 26.5%	Stanton 26.8%	Altec 20.5%	Moog 25.3%	Thorens 12.6%	JVC 11.2%
3M 15.7%	Ortofon 3.1%	Auratone 10.7%	Oberheim 3.1%	Dual 6.8%	Ampex 9.9%
Scully 9.1%	Audio Technica 2.9%	Electro Voice 7.8%	EML 2.7%	Philips 5.2%	Panasonic 9.9%
Studer 6%	Pickering 2.7%	KLH 2.4%	Korg 2.5%	QRK 4.9%	IVC 4.0%
Stephens 3.8%	AKG 1.7%	Westlake Audio 1.8%	Syn Aire 2.3%	Garrard 4.7%	RCA 2.9%
Other 2.5%	Empire 1.6%	Advent 1.3%	Yamaha 2%	Rek O Kut 4.3%	Other 5.2%
	Micro Acoustics 1.4%	Big Red/Mastering Labs 1.3%	Cat 1.5%	Sony 4.1%	
2) 16 or more tracks.	Other 10.8%	Klipsch 1%	Roland 1.5%	Pioneer 3.1%	
		Other 18.8%	Other 15.7%	Other 27.1%	

Billboard 1978-1979 International Recording Equipment & Studio Directory

Recording studios choose a synthesizer because it has the best sound, not the lowest price. And in *Billboard's International Recording Equipment and Studio Directory 1978-79*, their choice is clear. If you want great sound, the choice is ARP.

ARP

45 Hartwell Avenue
Lexington, Massachusetts 02173

CIRCLE 93 ON READER SERVICE CARD

SUBSCRIBER SERVICE
CHANGE OF ADDRESS

Planning to move? Please let us know six weeks in advance so you won't miss a single issue of SOUND ARTS.
Attach old label and print new address in space provided. Also include your mailing label whenever you write concerning your subscription to insure prompt service on your inquiry.
SOUND ARTS
14 Vanderventer Avenue
Port Washington, N.Y. 11050

Attach Label Here

New Address Here — *Please print*

Name ______________________

Address ______________________

City ______________ State ________

Date ___/___/___ Zip ________

and each area was assigned a code. The code, which was much easier to send to earth, was then decoded and the picture reconstructed here on earth. The important point is that a code is *not* a sound, and is therefore not degraded in the same way sound is degraded by being delayed.

But as you've probably guessed, there are some limitations with this system, too. First of all, the process of encoding and decoding a signal contributes a certain amount of distortion, even though the delay process does not modify the code. Therefore, much of the final quality depends on the efficiency and accuracy of the encoder, or analog-to-digital converter, and the decoder, or digital-to-analog converter. Also, these circuits are relatively expensive and consume more power than analog delay lines, requiring a heftier power supply.

As a result, digital delay lines tend to be more common in studios where quality is all-important, and in the repertoire of upper-echelon touring bands. The analog techniques tend to be used in less critical applications, such as budget solid-state echo units and flangers (especially those designed for stage use).

CONSIDERING THE DIFFERENCE

That's pretty much all the theory we need for now. There is some confusion regarding the merits of digital versus analog design techniques; you may have heard that "digital is the only way to go," and in a way that's true . . . *if* cost is no object. But analog techniques can produce excellent results with devices requiring short time delays (like flangers), at far less cost, if designed correctly. Neither technique invalidates the other, and they both have their uses.

In any event, time delay units are powerful audio processing units that are here to stay. Complex and beautiful effects, once only achievable at great expense in the most advanced studios, are now available to every musician who has an available $100 or $200. But using these new devices properly means knowing what to look for in terms of features and options, and that's what we'll investigate in the next installment.

These are the "big guns" in "professional" power amplifiers. Each of these amplifiers has individual features and abounds with specifications to impress potential buyers and to satisfy the professional user but they are **not** created equal... especially in reliability under professional (rack mounted) conditions.

Some of these "big guns" have been talking about everybody else being "behind", others are talking about comparator LED's, while others depend mostly on their good looks. The Peavey CS-800 comes out on top when you consider the features, the specifications (which are as good or better than anybody's), total power output, and price per watt of professional power.

Some companies have recently "discovered" LED's and comparator circuitry that Peavey pioneered and has been using for years. These recent "converts" were most vocal in the past against LED's...that is, until they updated their "plain Jane" units. Some of the other companies spend a lot on cosmetics but not much on built-in forced air cooling and large numbers of output devices to enable reliable rack mounted operation under continuous professional use.

Each channel of the Peavey CS-800 features 10 output devices and 2 TO-3 drivers bolted to massive modular heatsinks that are forced cooled by a 2-speed fan, has special distortion detection circuitry and LED indicator (not simple overload), as well as a functional patch panel on the rear to facilitate the use of plug-in balanced transformer modules, electronic crossover modules and speaker equalization modules custom tailored to Peavey's SP-1 and SP-2 speaker systems.

In comparing pro amplifiers, one should apply the old commercial sound "dollar-per-watt" rule. The CS-800 is again "on top" at 81¢ per professional watt. The fact is...Peavey is not behind **anyone** in power, durability, features or performance.

Below are the respective published specifications of the "heavies" in pro amps. Check for yourself to see how we all stack up. You might be surprised.

Peavey

Peavey Electronics
711 A Street
Meridian, Miss. 39301

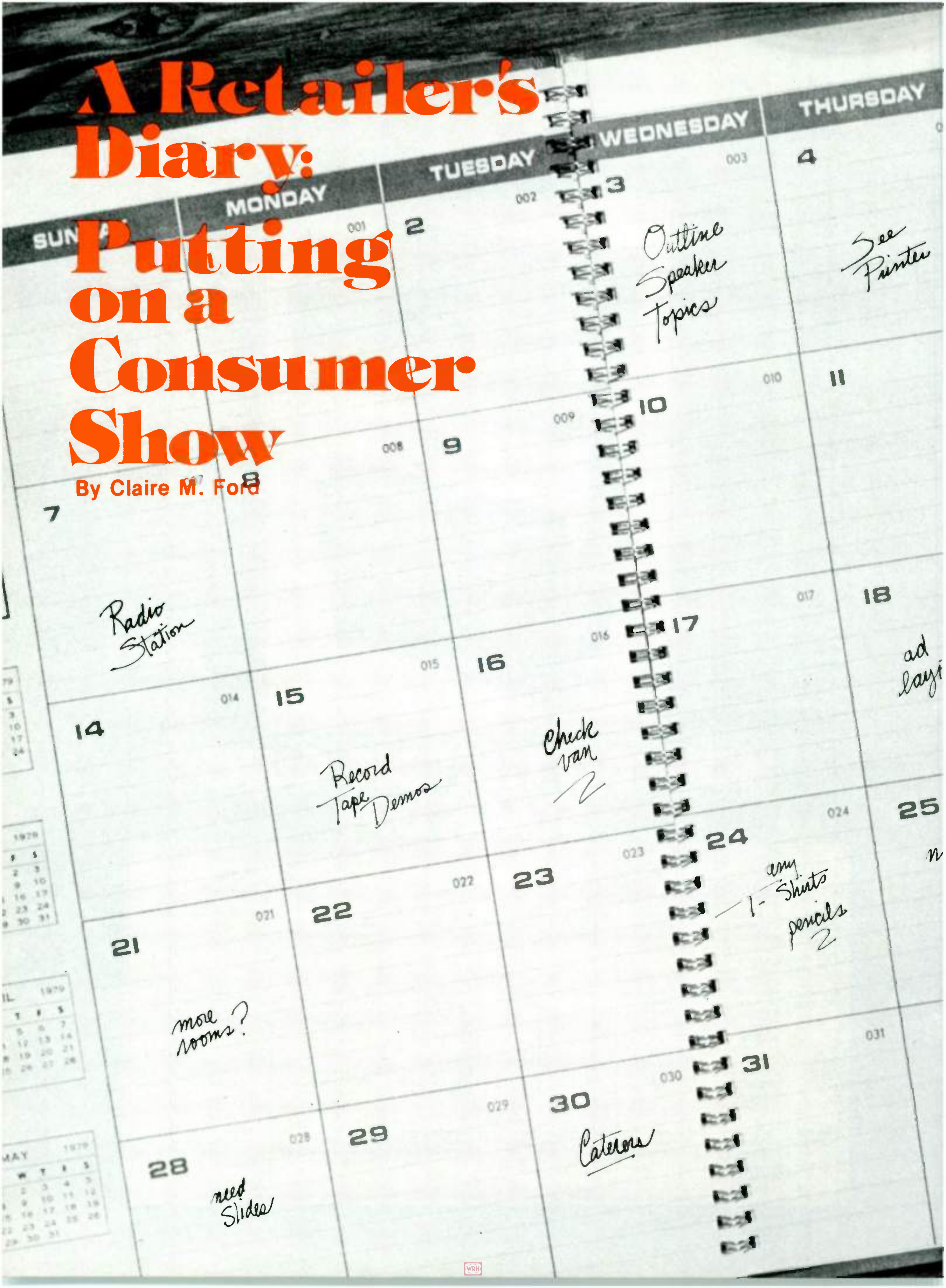
HOW DO THE "BIG GUNS" STACK UP?

	Mfrs. Min. Power @ Recc. Load	Output Transistors	Cooling System	Speaker Protection	Construction General	Turn On Delay	Output Circuitry	T.I.M.	List Price	Dollars Per Watt
Peavey CS-800	800 W Total 400 Watts/Ch. @ 4 Ohms; 260 Watts/Ch. @ 8 Ohms (Both Ch. driven)	20	2 Speed forced air cooling	Yes	Totally Plug-in Modular	None Required	Quasi Complimentary. All rugged NPN Silicon Outputs	Not given. No accepted Measurement standards Presently exist.	$649.50	**$0.81 per Watt** Based on 4 Ohms/Ch. min. load
Crown DC-300A	360 W Total 180 Watts @ 8 Ohms; 4 Ohms Not Given	16	Conventional Passive Airflow Only	No	Hard Wired	None Required	Quasi Complimentary. All rugged NPN Silicon Outputs	Not given. No accepted Measurement standards Presently exist.	$919.00	**$2.55 per Watt** Based on 8 Ohms/Ch. min. load
BGW 750 B	720 W Total 360 Watts/Ch. @ 4 Ohms; 225 Watts/Ch. @ 8 Ohms	20	2 Speed forced air cooling	Yes	Modular	Relay Circuit	Collector drive Complimentary using PNP & NPN Silicon	.02% No measurement details given.	$1099.00	**$1.53 per Watt** Based on 4 Ohms/Ch. min. load
Yamaha P 2200	700 W Total 350 Watts/Ch. @ 4 Ohms; 200 Watts/Ch. @ 8 Ohms	12	Conventional Passive Airflow Only	No	Hard Wired	None Required	Emitter follower drive complimentary using PNP & NPN Silicon	Not given. No accepted Measurement standards Presently exist.	$1095.00	**$1.56 per Watt** Based on 4 Ohms/Ch. min. load

All above figures based on manufacturers' published specifications and minimum recommended load impedances as of 11/1/78

A Retailer's Diary: Putting on a Consumer Show

By Claire M. Ford



Consumer seminars—or consumer shows—have elicited much interest from retailers of recording and musical instrument equipment. But the success of a retailer-sponsored show depends, first of all, on the avoidance of chaos. The inherent difficulty of putting on this sort of event can be alleviated by careful attention to a timetable and equally careful organization of the detail work that is necessary. Other retailers may be interested in the seminar experience we've had at Ford Audio and Acoustics.

About four years ago, we began to think of putting on a first-rate consumer seminar program. From our dealings with our customers and from our involvement in teaching courses for the Recording Institute of America, we had seen a craving for knowledge among consumers, and we thought we could present some of the product-oriented knowledge they wanted, which would eventually work to the benefit of our business.

In July of 1977 we finally made the financial commitment to have a consumer show. And this year—on July 22 and 23 of 1978—we put on a full two-day seminar program in the 18,000-square-foot ballroom of the Skirvin Plaza Hotel.

The idea for the 1978 seminars originated in July of 1977 as we concluded our first seminar attempt. Jim Ford's final statements to those attending obligated us to having a follow-up seminar a year later. And all of us agree he was probably right in doing so. The success of the first seminar was apparent immediately.

Our objective for the 1978 seminar centered around education. Basically, we felt our community should be made more aware of the professional audio business and what products and services are available for the creative audio enthusiast. For our customers who had already purchased equipment, we felt that better education would provide maximum performance and greater satisfaction with the equipment. We were not so much interested in making sales at the seminar, but were more interested in developing long term customer confidence in our ability to solve their audio problems. As a professional company, we felt the more we could inform the user, the more we would be stimulating sales.

With these objectives in mind, we wanted to conduct a seminar in a professional manner—as first class as possible. And we wanted to plan out an entertaining event which included activities and demonstrations that were enjoyable as well as informative.

The nature of our business dictated in our own minds the format of the seminar. Since we do a large church business, the first day was arranged specifically for a Church Sound System Seminar with exhibits and seminar topics geared toward that audience. The second day was designed as a Sound Seminar open to the general public. As a result, and as expected, the first day was a shirt-and-tie affair, while the second was come-as-you-are.

Planning for our seminar began a year in advance. July was chosen as the best time, since we wanted to be sure of good weather for people to travel in. It is also a post-show (CES, NAMM) time, so we could count on manufacturers having more time for participation. And college students would have more time to attend the Sound Seminar.

Since the Skirvin Plaza was, we felt, the only site in our city suitable for our seminar, and since it is a popular place, we reserved the room a year in advance. Then we began to plan the program.

In determining the program, we tried to determine the topics that people want to learn about, and the topics that people will come out on a Saturday or Sunday (some from long distances) to see and hear. We did not want to bore the audience with heavy scientific data that would be meaningless to most. Although in some ways we were attempting to educate technically, all sessions were kept to the basics. The demonstrations were carefully selected and were intended to have a dramatic and meaningful effect.

Early in the planning stage we had to sort out individual assignments for our own staff. As purchasing agent for the company, I had year-long contact with the manufacturers and their reps, so that I could discuss our plans with them from the start and gauge their enthusiasm and willingness to commit themselves. Our bookkeeper was in charge of all financial arrangements for the seminar. Our receptionist handled room accommodations. Our installers and carpenters were assigned to the physical work of set-up and tear-down. And Jim Ford was in charge of the technical arrangements. Since we didn't use an advertising agency, the services of our engineering draftsman were indispensible for our promotion materials.

We then had to get down to the nitty-gritty of the program to be presented. The church seminar and sound seminar were kept separate as far as program. The topics were chosen carefully to be consistent with our original objectives. The programs for the two seminars were as follows:

CHURCH SOUND SYSTEM SEMINAR

Main Sessions

Sound System Design and Operation
Speakers and Horns: How They Work
Power Amplifiers and Equalizers Demonstration (Tuning a sound system with a real time equalizer. Acoustical gain test for a sanctuary.)
Microphones: Types and Uses
Microphone Mixers: What's Inside?
Platform Design; Choir Monitors; Radio, TV and Recording; Cassette Ministry
Demonstration (Digitally delayed sound systems. Directivity of sound systems.)

Advanced and Special Sessions

Advanced Sanctuary Acoustics and Sound System Design
Advanced Recording Techniques
Use of Music Sound Tracks
Wireless Microphones
Technical Training: Wires, Connectors, Soldering, Hi and Low Impedance, etc.

SOUND SEMINAR

Multi-Track Recording (4- and 8-track)
Hi-Fi Systems
Rock and Roll P.A. Systems
Microphones

We then had to determine who would be the speakers on the various topics. Last year, we did the whole show ourselves. And our staff retained a portion of the speaking assignments this year. But it seemed to us that there was so much talent available at the manufacturing level, we should—for our attendees' benefit and for our own—make use of it. The problem of which manufacturers to ask was a real one; we were certainly in no position to slight any one. But we requested that all speakers refrain from using brand names. And we determined that we would rotate the manufacturer talent, using other manufacturers next time, since we did and do intend to have more seminars.

No manufacturer that we contacted refused the opportunity to provide a guest speaker. We did furnish suggested outlines for their talks. With no exception, the speakers turned out to be excellent. The speakers included: Ewald Consen, JBL; Jim King, Crown International; Bob Miller, AKG; Larry Benson, Benson Sound Studios; Frank I. Donnelly, Jr., Vega Corp; Dick Rosmini, Tascam; Jim Westmoreland, Technics; and Mike Petterson, Shure Brothers.

We then had to determine appropriate demonstrations to complement the lectures. This admittedly led us into a rental cost, since we rented projection equipment, including large screens; and a closed circuit television system consisting of one camera focused on a real time analyzer and four large TV monitors spread for audience viewing. We even rented a pointer. Although a house P.A. system was available, we chose to do our own with a sound system of high quality for maximum effectiveness, including a podium microphone, five guest panelist mics on the stage platform, and a wireless microphone system to pass among the audience for questions to the panel. In addition, we broadcast the lectures—at a low level—to the exhibit room.

We determined what special equipment was needed for demonstration. This included pre-recorded tapes; a real time analyzer; a graphic equalizer; and digital delay. Jim went out and made his own demonstration tapes a couple of weeks before the seminar. For instance, for the church seminar, he recorded, from a balcony in a church, *before* and *after* segments, using a digital delay for the *after.* The difference was, of course, noticeable.

As we got closer and closer to the event, more details had to be dealt with. We rented 45 tables, with appropriate cloths, laid out the exhibit, had signs and decorations designed and made up.

We also planned our special section devoted to audio history. Here we were lucky, since we own some beautiful antiques including an early gramophone and an old wire recorder. This exhibit provoked some very enthusiastic response.

We ordered seminar badges. Last year we had used preprinted badges, but found the extra cost unnecessary, and this year we opted for standard office-supply badges. Room accommodations had to be kept track of. We had an option on a number of rooms; an informal agreement was in effect for the hotel to notify us if they were pressed for the rooms and we had not definitely contracted for them at the time. Happily, there were no other large events in town that weekend.

To back up a bit, our promotion plans had to be put into effect long before the actual event, in order to serve their purpose. As I said before, we did not use an advertising agency. For our church seminar, we sent out over 1,000 direct mail pieces to large churches. Since we maintain our own mailing list for churches and we send out a monthly newsletter, we began to promote this event four months in advance. In addition we bought radio spots on Christian radio stations in Oklahoma City, Tulsa, Dallas and Amarillo. We bought newspaper space in the church section of the Oklahoma City and Tulsa papers.

For our general public seminar, we handled the promotion very differently. We do not maintain a mailing list for this target population. We bought over 400 radio spots on Oklahoma's most listened to stations. We bought newspaper space in Oklahoma City. In addition, we placed advertising flyers on cars at rock concerts. We lucked out here, since our contract with one of the radio stations included the services of placing the flyers and we thus had no extra labor costs. There was some Co-op money available for advertising, but most of the companies we deal with have no co-op allowances.

We planned a number of special events in keeping with our original objective of making this a fun event. A customer for whom we had outfitted a 16-track mobile recording trailer permitted the trailer to be parked inside the convention hall for display. We also used it to record some of the lectures. The same customer provided a professionally equipped video van for display in the lecture area; this was used to videorecord some of the lectures that were given.

Special notebooks were prepared for distribution. These bore specially printed covers and had to be ordered three months in advance from a bindery company. Although we had to use some judgment in determining the number of notebooks to order (before having a firm commitment of attendees), we intentionally over-ordered on these, since they can be used for other purposes. The notebooks contained notepaper, and dividers with headings for most of the major equipment brochures made available. (Manufacturers had been requested to provide their literature pre-punched for the binders.)

A banquet luncheon, using the facilities of the hotel, was provided on Saturday for all seminar participants.

On the second day, entertainment was presented by an excellent local band, whose services we acquired for a trade in equipment. Besides entertaining, the band was simultaneously

demonstrating some of the live P.A. and recording techniques discussed.

Two manufacturers—JBL and 3M—independently contracted with the hotel for their own demonstration rooms which worked into the JBL "listening room" and the 3M-Scotch "tape comparison room." We set up a second lecture room for smaller group lectures, which sometimes began spontaneously with some of our guest lecturers inviting those who wanted to discuss further a particular aspect of their material into the smaller room. This aspect is an important indication of the tenor of our seminars. There was an informal quality to them, with the manufacturer's people mixing with the audience, rather than just fulfilling a basic obligation.

On Saturday night we held a get-acquainted party for all the manufacturers, reps, Ford Audio employees and their respective families. Most of the reps and factory personnel arrived late the day of the setup and many of them (especially guest speakers) were new faces to us. We had reserved the hotel swimming pool for a setting to get acquainted over sandwiches and hors d'oeuvres and hoped for some swimming. But after a late afternoon chilling rain, there was no swimming. Bob Sellmansberger of the Bose Corporation helped out with the new Bose 802 speaker (which was not yet on the market). With little effort, the swimming party turned into a disco party. Everyone enjoyed the music, and the opportunity to get to try out the new speakers.

(Incidentally, we hired two security guards to protect the equipment at night.)

Most of our manufacturers were extremely cooperative. Almost all of them helped in the set-up of their area and interfaced with the customers directly. Many made available small complimentary gifts like cassettes or T-shirts that heightened the excitement of the event. Tapco arranged a drawing for some equipment. Overall we were very impressed with the support and willingness to help from our suppliers.

We charged $20 for attendance at the Church Sound Seminar and $4 (with a possible $1 discount) for attendance at the Sound Seminar. The $4 charge was instituted mainly to encourage committed attendees, rather than off-the-street bypassers. I estimate that these charges offset our costs by approximately 25 percent. No business was written at the seminars. No attempt was made to write business. We had 320 attendees at the Church Seminar and 500 at the Sound Seminar.

We would have to say the seminar was a success. We accomplished our goals. Additionally, we experienced

some side benefits that we had not anticipated. One was the fact that our walk-in business picked up noticeably after advertising for the seminar began. We have received many complimentary letters, some from people who had traveled a long way—some from as far away as South Carolina. It seems to us that an awful lot of people are really hungry for information. There is so much technology available; but the average person finds it difficult to find a source for this information. We belive the seminar has been a major factor in making our company an important force in the sound business throughout Oklahoma and the surrounding area.

Looking back, the seminar was a lot of work, but we will do it all again.

HOW MANY TRACKS DOES YOUR CUSTOMER NEED?

By Larry Blakely

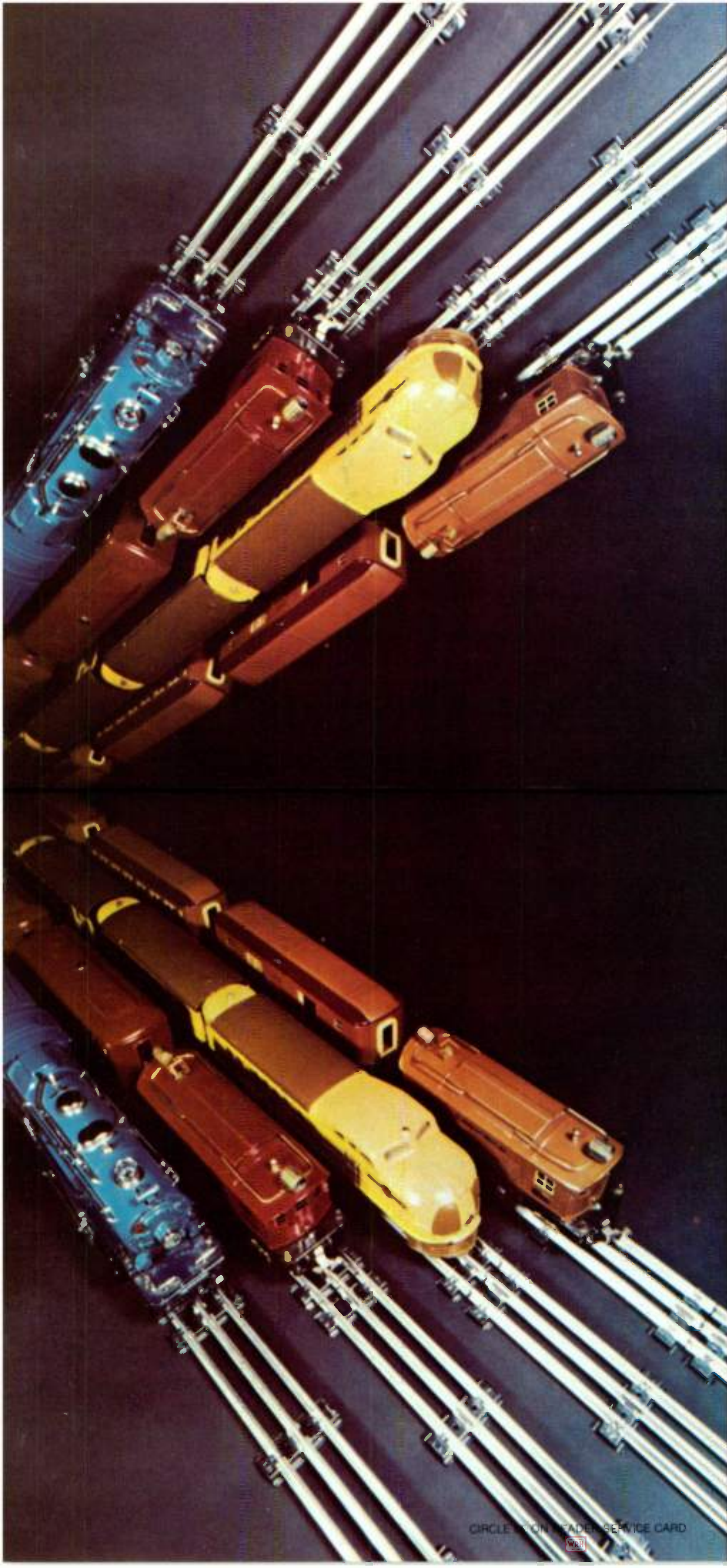
"Do I need an eight-track, four-track or two-track tape recorder?" This is a common question asked of salesmen of professional audio recording equipment. Some salesmen may suggest the most expensive machine with the greatest number of tracks. Most salespeople sincerely want to suggest the machine that will most fit their customers' needs. This brings up the question again! What are your customer's needs and how many tracks does he really need on his multitrack tape recorder?

I feel that one of the best ways to understand multitrack recording is to know its history. Multitrack recorders have been around the professional recording industry for some twenty years.

Some back-to-basics explanations may be in order to steer your customer along the path to the right equipment. The professional tape recorders utilize three heads: an erase head, a record head (to record) and a play head (which follows the record head). A tape can be played while recording via the playback head. However, there is approximately a two-second delay. This delay is the time it takes the tape to move from the record head to the playback head. It can be easily seen that if a musician was to listen to tape tracks previously recorded from the playback head and record new tracks on the record head, there would be an approximately two-second time delay. This means that the previously recorded tracks and the newly recorded tracks would be "out of sync" with each other.

HISTORICALLY SPEAKING

The first multitrack tape recorder was conceived in the mind of a musician, whose name of course is Les Paul. Ampex built the first multitrack tape machine for Les Paul in the early 1950's. This tape recorder had a special feature that would allow the record head to act as a play head on those tracks that were previously recorded. This feature was named "Sel-Sync" by Ampex. The Sel-Sync feature allowed the musician to listen to previously recorded tracks and record new ones on additional tracks. By utilizing the record head to play previously recorded tracks and to also



record the new tracks, all recordings would be done at the same time on the tape. All the music played and recorded would be in "sync."

This conception by Les Paul of a tape machine on which tracks could be recorded and on which one could at a later time re-record additional tracks "in sync" was destined to revolutionize the recording industry.

Some time in the early 1950's Ampex delivered Les Paul's multitrack tape machine (a very special order). Les Paul and Mary Ford used this machine to make many hit recordings that were technologically ahead of their time.

During this time the professional recording industry was at a whole different place. It took the recording industry a number of years to react to the multitrack recorder.

In about 1955 the recording industry was first experimenting with stereo (two-channel) tape recordings. These recordings were typically done with close microphone techniques. This meant that there were two or three microphones on drums, a mike on bass, a mike on guitar, a mike on piano, with separate microphones also placed on brass sections, string sections, background vocalists, and of course a microphone on the vocal solo. All of these many microphones were fed to a large microphone mixer (recording console) where the musical balance (mix) was obtained as well as echo and equalization added. In these early days of stereo (two-channel) recording, it was necessary for all of the musicians, vocalists, and the recording engineer to record an ideal performance. If anyone made a mistake, that section would have to be re-recorded and edited into the master stereo tape. All microphones were mixed at the console and fed directly to a stereo tape recorder, and all the music was recorded in one pass. If the vocalist had a cold, the session had to be cancelled, since everyone had to perform at the same time.

Around 1955, the "three track" recorder was introduced. This machine had three tracks and Sel-Sync. This allowed the recording engineers and musicians some new-found flexibility. The orchestra could be placed on two of the three tracks and the vocalist was placed on the third track. Now, if vocalist had a cold, he could do the best he could when recording with the orchestra. However, the recording engineer would *only* concentrate on getting a good performance and recording of the orchestra. The vocalist could then come back to the studio at a later date, listen to the previously recorded orchestra tracks (using the Sel-Sync feature) and re-record the vocal track.

This gave the vocalist the opportunity to re-record his vocal as many times as he felt necessary to capture his best performance. When the vocal was recorded the three tracks were again fed into the mixing console. The orchestra tracks were fed to the left and right channels and the vocal track was fed to both channels (equally), which placed the vocalist in the center of the stereo mix. This process of re-recording the three track (multitrack) tape on to a stereo (two-track) master tape was called a "mixdown." The people in the recording industry found a new flexibility of recording with the new "three-track" or multitrack tape recorder.

In approximatley 1962, a "four-track" recorder was introduced. This provided recording engineers and musicians an additional track to use. Now the orchestra could be split into more parts if desired. The orchestra could still be placed on two tracks, the third track could be used for the background vocals, and the fourth track for the vocal solo. This allowed the orchestra tracks to be recorded at one time (two tracks) and the background vocals and vocal solo to be added at a later date using the Sel-Sync on the tape recorder. This process of listening to previously recorded tracks and recording additional tracks is called "over-dubbing."

GAMES PEOPLE PLAY

Rock and roll was a big thing in the recording industry at this time. The rock musicians, record producers, and recording engineers found some additional tricks they could do with the four-track tape recorder. They found that they could place the drums and bass guitar on one track, organ and guitars on the second track, background vocals on the third track and the vocal solo on the fourth track. This meant that the instrumental background music could again be recorded on two tracks. This is called the bed. The vocal tracks could be recorded at a later date. When all the tracks were full, the four tracks were mixed down to a stereo master tape. By having the portions of the band and vocals split up on separate tracks, they could give a great deal of attention to the musical balance between the four tracks during the mix-down process.

Another thing commonly done during this time was to record bass and drums on track one, organ on track two, and guitars on track three. Now these three tracks would be mixed together with the recording console (achieving the proper musical balance) and transferred to track four. When this operation was successfully done, tracks one through three could be erased and yet additional recording could be done on the now empty tracks. This process is called ping-pong. Now strings could be recorded on track one, brass on track two and vocals on track three. When this was completed, a normal mixdown was done to a stereo master tape. Musicians and recording engineers would often repeat the ping-pong process a number of times. This process would allow them to effectively get eight to ten tracks from a four-track machine. However, when tracks are ping-ponged in this fashion, there is an increase in noise and a loss of high frequency response. Ping-ponging cannot be done too many times before there is a great loss in quality. I would like to point out that the Beatles' Sgt. Pepper album was recorded on a four-track machine using these recording techniques.

TRACK RECORD

Now multitrack recording was getting into full swing. Everyone started yelling, "I need more tracks so I don't have to ping-pong." Around 1967 the eight-track recorder appeared. Everyone loved this! Now the drums, bass, guitar, organ could each have their own separate tracks, and there were stull four more tracks left for the brass, strings, background vocals and vocal solo. People saw now that almost each instrument or sections of instruments could have their own individual track. When the multitrack (eight-track) tape was made, all attention and detail was on the quality of the performance and not the musical balance between instruments.

The musical balance was not of concern until the multitrack tape was mixed down. During the mixdown process all the concentration was on the musical balance, stereo placement (left, center, or right), echo, and equalization. Ping-pong was still done with the eight-track machines and the musicians, record producers, and recording engineers were still screaming for more tracks. "We want a track for every instrument," was the cry. At this point it became a real horsepower race. The studios with the most tracks got most all the business.

Around 1969, the sixteen-track recorder appeared on the market. Now more tracks were available and less ping-pong was required. By this time most all recording was done by a cut-and-paste method. If someone didn't like his performance he could most always re-record his track at a later time. No concern had to be given to the musical balance until the mixdown process. The way in which things were recorded had changed a lot between 1959 and 1969.

A common term began to be used when something didn't sound right: "We can fix it in the mix." Multitrack recording was a good tool for those who could utilize its flexibility. However, it became almost a cop-out way of recording for those who didn't know what they were doing. It was common for an album to take 200 hours of studio time. The multitrack recording would have tracks recorded and re-recorded countless number of times. It was also not uncommon to have a mixdown of a tune done ten or twenty times. If you didn't like the mixdown you could always do it again.

In about 1971 the first of the 24-track machines appeared. These machines allowed even more additional tracks. Today 24-track machines are often locked together by an electronic device that will allow 48-track recording. Where does it all stop? I don't know. There was a cartoon in a professional recording magazine in the early 1970's. There was a picture of a recording engineer at the console and an enormous tape recorder in the rear of the control room. This tape machine was labeled the "Ajax 72 track recorder." The caption read: Track 70 cowbells, track 71 dripping water left, track 72 dripping water right.

The use of 24 to 60 tracks does reach a point of diminishing returns and there is a degradation of quality due to the noise buildup of a great number of

SIX WAYS TO CLEAN UP YOUR ACT.

If you play electric piano, organ, bass, rhythm or lead guitar, or sing—JBL K Series loudspeakers can make you sound better.

Each of these high-efficiency speakers is specifically built, as only JBL can, to provide the cleanest, clearest, and best possible reproduction of your music.

And that's exactly why more professional musicians and performers choose JBL.

Maybe you should, too.

JBL K Series loudspeakers are available in six different sizes, from 10" to 18".

JBL

GET IT ALL.

tracks. However, some musicians do feel the need for this enormous balance of tape tracks.

WHAT DOES IT MEAN?

Anyway, you have just read the history of multitrack recording. This background information could give your customers a good idea of what multitrack recording is all about, its practical applications, as well as its advantages and disadvantages. How does all of this relate to you? How many tracks does your customer need in his recording studio?

To equip a studio, your customer will of course need an assortment of microphones, stands, direct box, mixing console, multitrack recorder, two-track master recorder, monitor amplifier, monitor loudspeakers, headphones, and a headphone amplifier. These are the basic ingredients that are required to do multitrack recording.

As usual, the prine consideration is the kind of budget your customer has for the purchase of a multitrack recording setup. If your customer, after budgeting enough funds to purchase the equipment listed above, can only afford a four-track tape recorder, the answer is simple: Many good recordings have been done on four-track recorders. It does mean that more attention must be given to the musical balance of instruments occupying the same track (this cannot be changed). It also means that your customer may have to use ping-pong in his recording if he needs additional tracks. You can make good recordings with a four-track recorder. It requires more attention to detail, and more skill to achieve good results.

It is also necessary to take into account the type of music and number of instruments your customer is recording. If he is recording three or four instruments and a vocal, a four-track is fine. If he is recording six instruments, background vocals, and a vocal solo, it can still be done on a four-track. He has the option of grouping instruments together and using just the four tracks. He can also record in building blocks. Bass on one track, drums on another track, and guitars on the third. Get a performance of these instruments that he hs happy with, then mix these three tracks together and ping-pong them to the fourth open track. Now the original three tracks are open to record other instruments or vocals.

If your customer is just starting to learn about recording and this is his first recording system, I feel that a four-track is a good place to start. He can learn recording skills far better by having less flexibility (four-track instead of eight-track) because he will have to work a little harder for the results he gets. If he were to start out with an eight-track, the additional flexibility often makes him a little lazy. He won't have to work as hard to get instruments mixed on the individual tracks utilizing eight-track as when combining them together because of fewer available tracks on four-track. As I mentioned earlier, a four-track makes an excellent machine for the beginner. This allows him to jump in and get his feet wet for the least amount of money, and enables him to learn the multitrack recording process. If he is successful, believe me he will be banging on your door for an eight- or sixteen-track at a later date.

If your customer is already familiar with the multitrack recording process and desires a recorder with eight or more tracks, by all means give him what he wants. If you likewise have a customer who wants to purchase his first multitrack setup and wants eight or more tracks, and has the budget for it, accommodate him.

However, as I mentioned earlier, such a customer may not need a great number of tracks. Talk to him about what he is going to record and what his requirements are. Make sure you fill his needs and desires.

Up until this point I have been talking about a customer who is purchasing multitrack recording equipment for his own home recording studio. This customer is only going to use the studio for recording his own particular type of music.

However, if your customer is purchasing multitrack recording equipment for a studio that will be open to the public or a number of different recording groups with various types of instrumentation, this is an entirely different story. Such recording studios that are in the business of selling studio time to clients will normally require more sophisticated equipment and recorders with a larger number of tracks.

One of the first questions out of a recording studio customer's mouth will be, "How many tracks do you have?" Now you need to be in the horse power, or should I say track power, race. Typically, such recording studio clientele will use the studio with the greatest number of tracks. They will normally prefer an eight-track studio over a four-track studio.

If your customer is going to open a recording studio to sell studio time to musical groups, you should try to sell him an eight- or sixteen-track recorder.

TAPE CONSIDERATIONS

Another area that needs investigation is the cost of the tape used on different types of multitrack tape recorders. Tape recorders utilizing the "quarter-track" format will need ¼″ tape for four-track, ½″ tape for eight-track and 1″ inch tape for sixteen-track. Tape recorders utilizing the "half-track" format will need ½″ tape for four-track, 1″ tape for eight-track, and 2″ tape for sixteen-track.

I bring up this question of tape size because this again relates to dollars. A reel of ¼″ tape is less expensive than a reel of ½″ tape. A reel of ½″ tape is less expensive than a reel of 1″ tape. Many reels of tape will be used when doing multitrack recording. Make sure that your customer doesn't purchase a large multitrack recorder and not have enough money to purchase tape to feed the monster. He still must purchase tape for the two-track master recorder to do his mix-downs.

TIPPING THE HAT

The original question, "How many tracks does your customer need?" is a loaded question. There are a number of features that need to be investigated and discussed with your customer to give him an honest answer. But first he has to know what multitrack recording is all about. How does the multitrack recording process work, how do you utilize it, and what do you need it for? It was for this purpose that I outlined the history of multitrack recording. Knowing this history should give you and your customer a much better feel for how multitrack will apply to his particular needs.

The anwswer to the question is not always an easy one. But a good insight into the multitrack recording process should guide you and your customer toward an intelligent and honest decision.

Last but not least we should all tip our hats and say "Thank you Les Paul" for the modern day recording process known as multitrack recording.

HIGH TECH AT THE A.E.S. CONVENTION

By Mike Beigel

By this time nobody is surprised at being surprised. We are already accustomed to new developments, new kinds of products, new buzz words and the like; and we don't even become alarmed that the acceleration of the technological revolution itself is accelerating. One even stops wondering where, when or if it will end. Some products are obsolete before they are produced; news is sometimes old before it's digested.

I'll bet the editor is wondering when I will stop raving and start reporting on the Audio Engineering Society convention in New York City in November. Just one more point. Even the people who are responsible for all the "breakthroughs" are suffering from severe future shock. It's no small problem. Technological change is taking place at a rate that makes the inflation rate look lethargic.

You are probably a store owner or employee, or a musical product manufacturer, or maybe a musician or recording studio person. You're more interested in your work than most people or you wouldn't be in this crazy business. In one way, all the new products, fads and "things" in general are exciting and interesting. In another way they make "business as usual" an impossibility.

At the AES show, two major types of "wares" are displayed: the newest of new products in the whole audio field; and the theoretical and experimental work which precedes new products. The professional audio industry is somewhat different from the musical products industry or the consumer electronics industry, and in the sense that it is different, so is the AES convention different from NAMM or CES. The AES is at the crossroads between science and commercial application. As such, professional audio is traditionally a few years ahead of both the consumer audio and the musical products industries, and it's a likely place to look for trends that may eventually become rooted in these industries. Now more than ever, because the time lapse between "breakthrough" and "off-the-shelf" is shortening.

It is a showcase for new products, a proving-ground for prototypes, and a forum for exchange of knowledge. The products on display are not for sale at the show. Technical sessions , at which papers are presented covering all aspects of audio research, run the full four days of the show.

The professional audio industry is a small but sophisticated marketplace, consisting mainly of recording studios and similar high-budget enterprises. Professional equipment is made in small quantities, and to the highest standards, so it is obviously quite expensive. The cost represents research and development expense to a greater extent than for larger markets.

DIGITAL TAPE RECORDING AND EDITING

The parable of the ten-dollar pocket calculator: In three or four years, a product that cost thousands of dollars and fit on a desk-top has shrunk to credit-card size and a ten-dollar price tag. Please note: The digital audio products presented at AES and described here cost "thousands of dollars" and fit "on a desk top" more or less.

Digital audio recording is the most dramatic advance in sound storage since the invention of magnetic tape. Signal-to-noise ratios (without any form of pre- or- post-processing) of 90 dB. Flat frequency response. No amplitude-dependent distortion. No "dropouts." Absolute level encoding. Error-correcting codes for archival tapes. No print-through. No wow and flutter. No signal degradation in duplicating tapes. Invisible splicing. Where can I get one?

Digital tape recording is so clearly superior to "analog" tape recording that it will probably create the equivalent of a product landslide in the next few years. Right now, in its introductory forms, we observe the usual kinds of difficulties common to emerging technologies. First, the price is high because the technology is new. Second and most important, there is insufficient standardization of product forms. Each company in the field has worked out its own "best" solution, and they don't necessarily agree. There are questions of number of tracks, tape width, "sampling rates," conversion methods, word-lengths, error correcting methods, etc. etc. All legitimate questions, but until (or if) there is general agreement about the answers, you will be committing yourself to company A when you buy company A's first product.

The products themselves: Ampex, 3M and Soundstream have (or will soon have) studio-type machines with four to thirty-two tracks. All boast similar specifications and performance features. They resemble standard professional tape decks. Sony has a two-track system which interfaces to their standard videotape recorder, and has specs similar to the others.

The electronic editing method used in these devices deserves special attention. Two details are noteworthy: You don't splice the tape; and the editing is microscopically accurate. The nature of digitally recorded signals allows precise identification and manipulation of each signal-sample, which allows display of the audio-signal on a CRT screen. This also enables the user to find the perfect "splice points" in two pieces of musical material, and to assemble a virtually noise-free splice. The cumulative result of the digital recording and editing systems is really a quantum jump in audio technology.

DIGITAL DISCS

The "record" spins at about 450 RPM. The "needle" is a laser beam. The "grooves" are tiny potholes, each containing a digital code word. The record has the same incredible specifications as the digital tapes mentioned previously, except that it's prerecorded and holds over two hours of program material. The "record player" is similar in size to the average home turntable. It has no tone-arm, since the laser-beam needle tracks the record from underneath, with no physical contact. So the record won't wear out, either.

You can't get one yet. Until the prospective manufacturers agree on a standard format for digital discs, there isn't much sense in going into production. Soon, though, these problems will be straightened out. Can you imagine what this will do to the hi-fi marketplace? Sony and Panasonic exhibited digital discs.

DIGITAL DELAY & REVERBERATION PROCESSORS

The number, variety and quality of digital processors have increased dramatically. Definite product classes are beginning to emerge, and I will risk an attempt at defining them:

- Delay-only devices of fixed or switch-selectable length, for sound-reinforcement or simple sound-effects applications. They consist of a single digital delay with reasonable signal-to-noise figures and bandwidth specs, and a budget price-tag.
- Delay and special-effects units, consisting of one or more variable delay lines, feedback controls, and sweep-oscillators. In addition to pure delay, they provide effects like flanging, "hard" reverberation, multiple echos, and possibly pitch shifting or chorus effects. Often the bandwidth (frequency response) is variable to accommodate extra-long delays.
- Advanced delay and reverb processors. A requirement for natural-sounding reverberation is a multiplicity of delay taps, with provision for special filtering and summing programs, in addition to the more common effects programs. A wide variety of features exists here, as well as differing design philosophies and price compromises.
- Modular memory systems. A new breed of devices, relying on an expandable "audio memory," and a number of optional delay processors to create very versatile, open-ended systems.

At first, the wide variety of devices is a bit confusing, but on closer examination one sees that the trade-offs among audio quality, versatility and price leave a place in the market for each of these products. Many companies are in the field, including Lexicon, Delta Lab, EMT, Ursa Major, MXR, Sound Workshop and others. To properly cover all these devices, a separate article would be necessary.

ANALOG TIME PROCESSORS

Digital devices haven't completely taken over the so-called "time processing" field. High quality analog delays, flangers and special effects devices are available as a cost-effective alternative to the all-digital units. Companies like Loft Modular Devices, MXR, MICMIX, and Advanced Music Systems exhibited flanger-delay units with excellent performance and versatility.

A noteworthy analog processing device is the Marshall Electronics Time Modulator. By itself, or in connection with Marshall's 5050 Stereo Effects Expander/Synthesizer Interface Unit, a veritable warehouse of delay effects is available. The device exhibits 95 dB signal-to-noise ratio.

DIGITAL DISPLAYS

The turntable that reads out proudly "33.33" in a bright LED display may not be the most practical example of the new display technology, but it is (so to speak) an indicator. Of more profound significance are the multi-channel plasma bar-graphs, LED arrays, CRT view-screens adorning the newest mixing boards and audio control panels. The visual impact of twenty-four channel line indicators is a perceptual help to an audio engineer trying to control such a mass of information. The changing economics of these visual displays will soon give them a competitive edge over traditional metering systems, and the ability to tailor their response characteristics to psychoacoustic parameters is a benefit to the recording art.

AND DIGITAL TEST EQUIPMENT

Now it is possible, economically and technically, to produce audio analyzers which give a graphic readout over the whole audio spectrum: instant spectrum analyzers. They can be used for acoustic and electronic measurements, in such a wide variety that they promote creativity in all audio measuring situations. The newer analyzers can store audio data, provide different scaling functions, and even provide overlapping displays for different signals. Even in color, with different color bar-graphs for different channels or events. Ivie, Barclay and Crown each exhibited products with different approaches to the problem. Within five years, instruments like these will be on every self-respecting test-bench, recording studio, and live-performance setup.

THE PHENOMENON OF "EXCITATION"

It's an effect you can't define—in fact, you don't always know it's there. An impressive array of performing artists seems to rely rather heavily upon it. There is a definite "mystique," and you can't buy one. So far, Aphex rents them to studios and well-heeled musicians. Their new-found competitor EXR will be selling a somewhat similar device. It has something to do with the manipulation of phase characteristics and/or second-harmonic enhancement. The term "presence" seems to be applied to the effect. I have a strange feeling that someday soon a large population of "exciter" boxes will be traveling in musicians' gigbags. A patent has been issued on the effect.

AN INTERESTING MICROPHONE

What if you had a microphone that could alter its spatial characteristics to provide a completely selectable three-dimensional response? First, close in to the performer, then out to the full soundfield of the concert hall. Then looking up, or down, or sideways. With cardioid, omnidirectional or multichannel spatial characteristics.

The microphone system developed by Michael Gerzon for Caltrac Audio, Ltd. not only provides all these functions, but records all the acoustic information on four channels of tape so that the response can be edited and manipulated after the performance by the microphone control electronics. The implications for live music and theatrical recording are wide-ranging.

OTHER PRODUCTS: ADVANCING NORMALLY

A major segment of the audio industry was on display including amplifiers and speaker systems, cross-over networks, parametric and graphic equalizers, noise reduction equipment, mixing consoles, analog reverb systems, analog tape-recorders and associated noise-reducers, magnetic tape, vocoders, and a synthesizer or two. Product introductions and improvements continue at a more relaxed pace. Since these product lines are more familiar and more accessible through their general exposure in the music business, I will leave the documentation to some other writer.

TECHNICAL PAPERS AND PRESENTATIONS

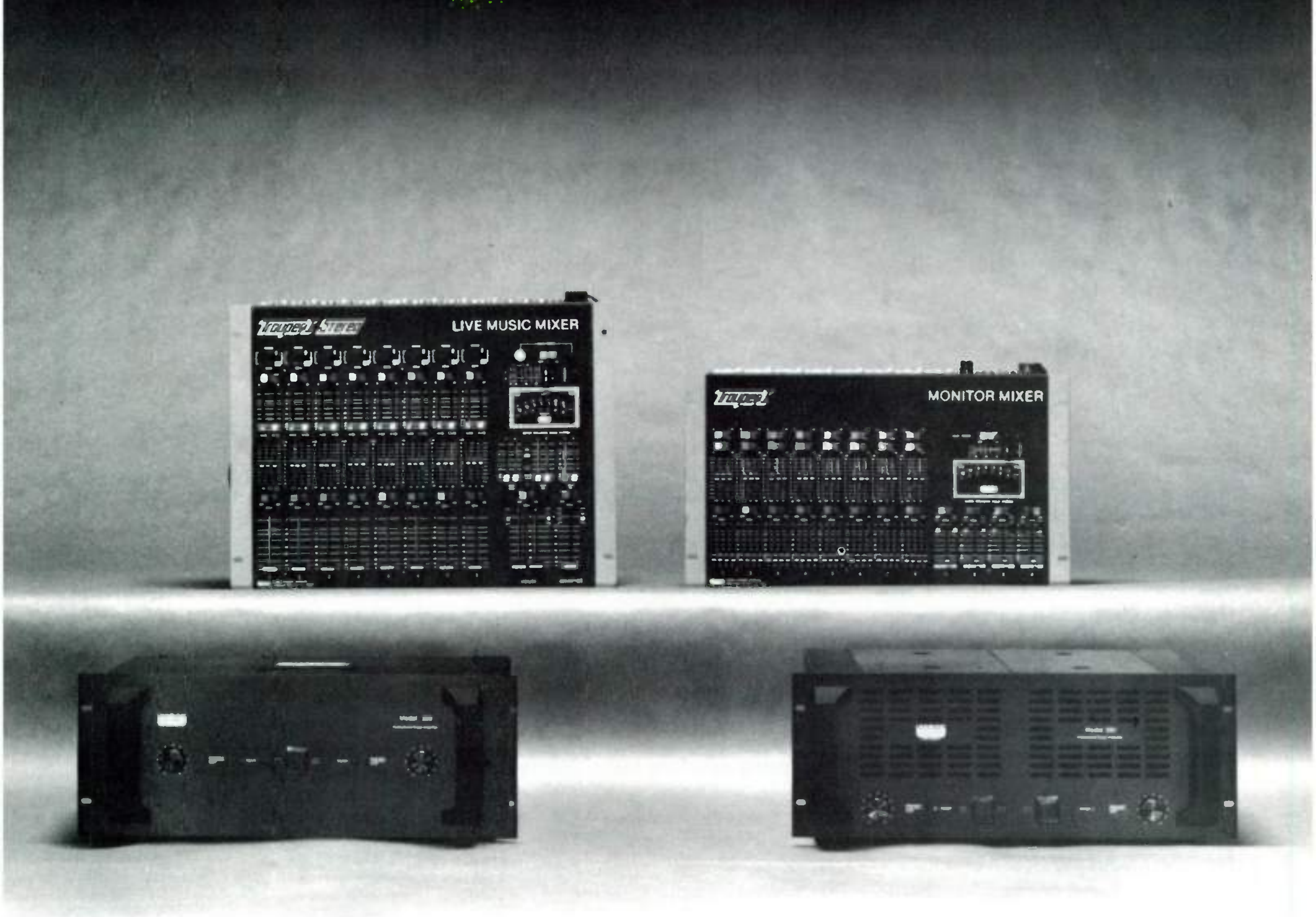
Actually, the AES is not mainly in the convention business: its prestigious *Journal of the Audio Engineering Society* is a chronicle of the scientific and engineering breakthroughs in the audio field, and a basic reference publication. The materials for the AES Journal are gathered from among a large number of presentations and "pre-prints" contributed by audio engineers and scientists. These pre-prints are sold by the AES, and presented by their authors (often with an accompanying demonstration of results) at the technical sessions.

The format and content of the technical sessions changes over the years, as the basic issues in science and technology change. This year, the following major divisions of current technology were reflected: audio in medicine; audio measurement and instrumentation; management/engineering, semi-pro studios; disc recording and reproduction; magnetic recording; sound reinforcement, architectural acoustics; digital techniques; audio in broadcasting; electronic music; transducers; subjective judgments of audio; signal processing; and applications of digital technology to audio recording.

Over eighty presentations were delivered at this convention, covering a wide variety of topics at differing levels of research and application. To name a few: Walter Gish analyzed and re-synthesized instrumental sounds on a computer with astoundingly "real" results. Brian H. Berkeley designed a digital audio mixing system with impressive performance, a number of papers dealt with different aspects of digital recording (often with conflicting points of view). It's no use; they covered everything from cardiovascular medicine to disco sound systems. The list of papers is available from the AES, at 60 East 42 St., NY, NY, and they can be purchased.

For those with inquisitive or scientific leanings and some technical proficiency in audio, the AES is a source of information relevant to the present as well as the future.

If your equipment isn't working, neither are you.



You're only as good as your equipment allows you to be. It is the showcase for your talents. Quite simply, Uni-Sync created the Trouper Series mixers and Professional Series Amplifers to work, when you do. Write or call for our complete product line and the name of your nearest Uni-Sync dealer.

UNI-SYNC
A BSR COMPANY

742 Hampshire Road/Westlake Village, California 91361/(805) 497-0766
Uni-Sync Div. of BSR (Canada) Ltd., 26 Clairville Rd., Rexdale, Ontario M9W5T9/(416) 675-2402

CIRCLE 87 ON READER SERVICE CARD

SOUND REPORT

Solving The Customer's Sound Problems - Case History Part 3

By Ralph Morris

This series was initiated, you will remember, as our response to a consumer who quite clearly needed help. His problems were comprehensive enough to constitute the consumer question par excellence, necessitating an equally comprehensive answer from the retailer. You may have forgotten the details, so we are reprinting the original letter from the musician, followed by the conclusion of Ralph Morris's individualized answer. For continuity, check the November and December issues of SOUND ARTS to see whence this all began.

—Editor

CONSUMER'S LAUNDRY LIST (printed verbatim—again)

"Concerning our sound system: We're a local band and don't have a concert system, and can't afford the best. But I would like to know how to get the best results with what we have.

"We are working with two bass reflex cabinets (no name) that hold two JBL bass speakers each (1 cab. per side), 4 Altec high frequency horns multi-cell (bigger than the ones used in Voice of the Theaters), medium priced drivers for power—a BGW 500 for the bass cabinet and two (100 watt) Altec amps for the horns. We use two 8 channel stereo Yamaha mixers (unpowered), a mono JBL crossover, a stereo Soundcraftsman EQ and Roland space echo. We use AKG (D-130?) and Shure SM-58 mikes.

"What I need to know is: Does everything match up ok? Is the EQ ok to use with the Yamaha mixers? Are we losing anything there? How can we get the best drum results? You know that big, deep in-concert type sound? We can't get it quite right. Would a small compresser be of any help? How would you suggest wiring the whole thing up to achieve its full potential? Would you suggest midrange horns and a new crossover? We don't get as much bottom as we should. Our bass cabs. are twice the size of Altecs Voice of the Theaters. The whole thing seems to be lacking something. I know it's hard for you because you've never actually heard it and can't be precise in saying what's what, but you people know what's happening. How can we put it together to do it right? It's bi-amped now of course. Would running in stereo help?"

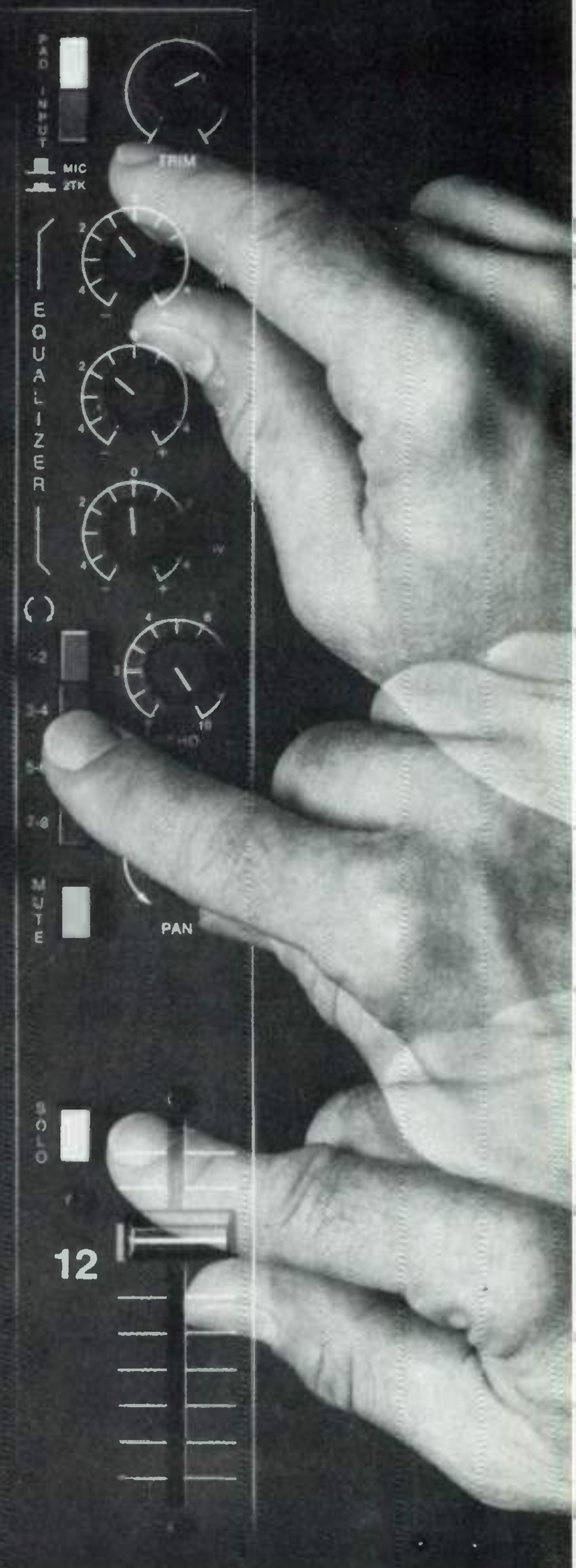
I received several comments about parts one and two of this series, so I will try to take them into account, and address the pertinent ones. The impertinent ones, I would like to ignore, but I will comment on them, too.

I should, first of all, allow that there are a few correctly designed bass horns on the market which do what they do very efficiently, and if you like that sort of sound they might be just right, for you.

For those who inquired about self-made enclosures, I would recommend that the instructions supplied by the manufacturer of the loudspeakers be followed. The back of the cabinet should be put on permanently, with lots of screws and glue, so that it is really sealed air-tight, and the loudspeaker(s) should be mounted from the front with cork gaskets and clamps or machine screws and T-nuts. If a ported (bass reflex) enclosure is preferred, the port should be made of a correct size in relation to the size of the box, according to the manufacturer's recommendations, or a small port should be started with and worked up to something less than a maximum of one-half the effective cone area of the loudspeaker. Since our "ear's" memory is not quite precise, two enclosures should be used, with the port enlarged

THE ONE-KNOB SQUEEZER.

A compressor/limiter that gives you a free hand.



There are times in the life of every studio operator when an extra hand would make things a lot easier. It's for times like those that dbx designed its new Model 163 compressor/limiter. We call it the "one-knob squeezer" because it has only one control—to adjust the amount of compression desired. As you increase the compression ratio, the 163 automatically increases the output gain to maintain a constant output level. It's quite clearly the easiest-to-use compressor/limiter on the market.

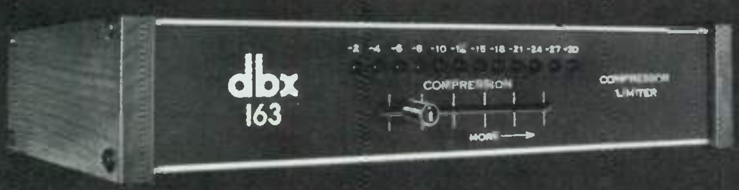
But that's not all. Because the 163 is an "Over Easy" compressor/limiter, too. Which means that as the signal level crosses the threshold, the 163 gradually adds the desired amount of gain change over the range of several dB. The result is the most natural-sounding compression you've ever heard.

The 163 is as easy to install as it is to operate. It's light and compact—two may be rack mounted in a 1¾″ space—and it interfaces easily with phono connectors.

But the easiest part of this "Over Easy" limiter is its cost. The nationally advertised value of the 163 is $189.* With the money you save on a pair of 163s, you can get two extra hands in the studio. You can hire yourself an assistant.

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

dbx
UNLOCK YOUR EARS



*Nationally advertised value. Actual prices are set by dbx dealers.

in first one and then the other, so comparison can be made. When you reach the optimum size port, according to your ear, or the rule-of-thumb maximum stated above, replace the front baffle board with a new piece of plywood, so all your enclosures are the same. Obviously, don't glue the front baffle board in place until you have made up your mind, just use lots of wood screws, so they can be removed. If you remove them several times, use the next larger screw size when you get ready to put it together for good. Remember that the cut-out for mounting the loudspeaker from outside the box is smaller than the outside dimension of the loudspeaker. For the purpose of the maximum port size rule, above, a twelve-inch loudspeaker has an effective cone area of approximately eighty-five square inches, and a fifteen-inch loudspeaker has an effective area of about one hundred thirty-five square inches.

A recipe for bass reflex enclosures for two fifteen-inch loudspeakers includes a sturdy box of one-half or five-eighths inch plywood, with two by two (actually one-and-a-half inches square) corner braces, glued and screwed. Make the box two by two by four feet, outside dimensions, which will provide an internal volume of ten or eleven cubic feet, depending on whether or not the mid-range horn and driver are inside the same box.

The loudspeaker manufacturer will usually furnish the correct size of the cut-out required to mount the speaker. Make these cut-outs for the two speakers next to each other (not too close, so as to weaken the baffle board) and make the port at one end. Don't put the port between the loudspeakers, or make two small ports, or anything like that, or you'll spoil the system. If you make the port six by twenty-two-and-a-half inches, it will equal one hundred thirty-five square inches, just about the effective area of one of the loudspeakers, or one-half the combined area of the two loudspeakers, for our rule of thumb. This will tune our enclosure at about fifty-five Hertz, which means that from about thirty-five to seventy-five Hertz, we will get some free bass boost, just where we need it. By the way, none of these dimensions are critical, so if you make a mistake of a whole inch in the port size (7 × 22½ = 157½ square inches, for example), the tuning would occur at about fifty-seven Hertz, only two Hertz different. If you make the box smaller or larger

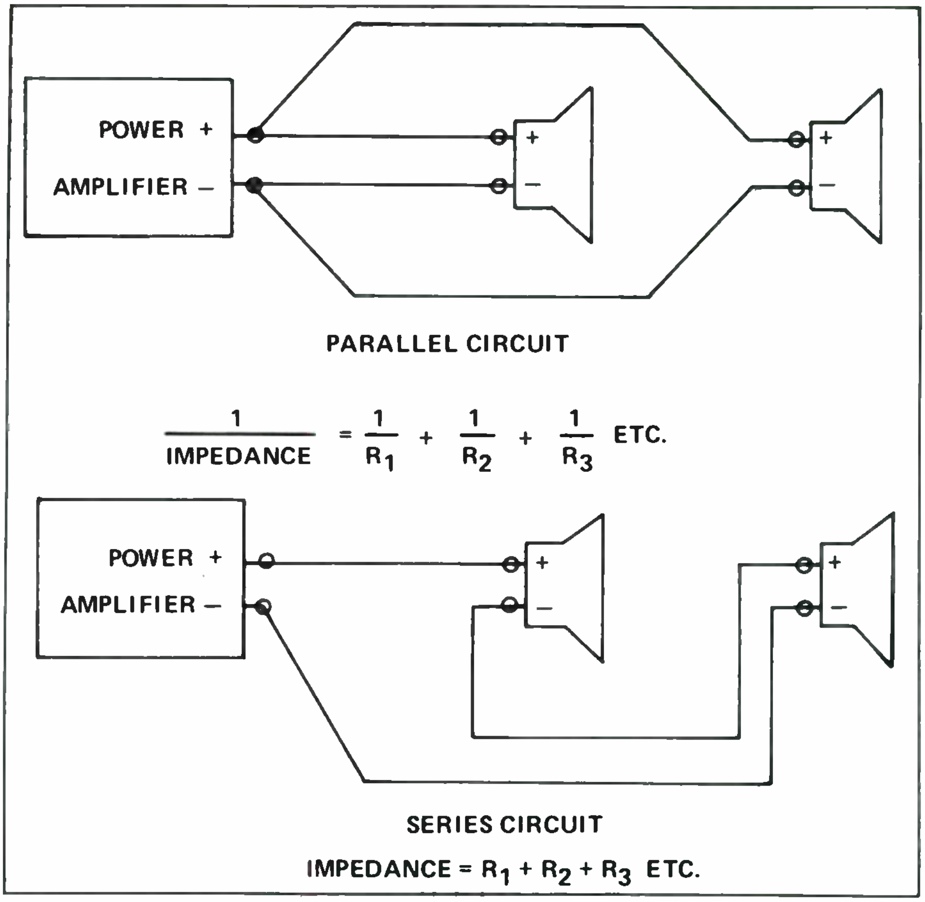


Figure 1. Parallel and Series Circuits

by a whole cubic foot, the smaller one would be tuned at about fifty-eight Hertz and the larger one at about fifty-two Hertz. There are really no secrets to bass reflex enclosures.

Bass horns are all right. I just happen to prefer the smoother bass response of those big woofers when they're looking out at you from the front of a bass reflex or sealed box.

Bass reflex enclosures provide a gain in efficiency, compared to a sealed box. However, since we can apply more power to the same loudspeaker in a sealed box, that could be made nearly as loud as the ported enclosure, although it would still not be so efficient in low-end response.

For those who prefer bass horns, I wouldn't recommend building your own. The curved wood required to make a real horn shape is too difficult for the average workshop, and even if straight panels are used to approximate the horn shape, it is a difficult thing to make, and the design and dimensions are critical as to whether it even works, or is more efficient, not to mention the "color" that the best horns add to the sound. Bass horns should be purchased from the loudspeaker manufacturers. If they are moved around too much, put in a few two-by-two inch corner cleats with screws and glue if they start to come apart.

BI-AMPING AND CROSSOVERS

We talked about bi-amping in Part I of this series, but there may still be some confusion, so we will try to clarify. Bi-amping is the practice of using two amplifiers for one (or more) two-way loudspeaker systems. This means that a separate amplifier (or at least a separate *channel* of an amplifier) is used to drive the bass speakers, and a separate channel is used to drive the horn, or treble elements in a two-way or three-way loudspeaker system. In the case of the three-way system (bass, mid-range, and highs), the high frequency element or tweeter may be crossed over or matched with the mid-range driver by a passive dividing network or crossover. This means that the tweeter doesn't have its own amplifier channel, as the bass and mid-range do. The tweeter and the mid-range driver are both driven by the amplifier channel we have designated for "high frequency," and the output of this amplifier is divided between them by a passive crossover similar to those found in

ADVERTISERS OF NOTE

"ROLLING STONE's coverage of all things musical consistently engages the musician-reader. It's an appealing environment for the creative artist, as comfortable as a well-lighted stage and satisfying as a responsive audience."

—Bob Hoffman
Vice President—Marketing
ARP Instruments, Inc.

"And now, a standing Ovation for ROLLING STONE. You've taken us to some pretty rare places and let us talk to all the right people—from roadies to rock stars, from students to stockbrokers, from hippies to homesteaders, from bureaucrats to backstage groupies. We couldn't have done it without you."

—Jim Hennessey
Advertising Manager
OVATION INSTRUMENTS

"ROLLING STONE helps us reach that important market between the ages of 18 and 34—the most potent buying group in America today. It's a "must" for a full-line musical instrument advertiser such as Hohner. Need I say more?"

—Jack C. Kavoukian
Director of Advertising
M. HOHNER INC.

NOBODY DOES IT BETTER

Rolling Stone

New York—(212) 758-3800 Los Angeles—(213) 461-3571 Chicago—(312) 782-2366

Come see us at NAMM—booth 716

CIRCLE 95 ON READER SERVICE CARD

two-way or three-way loudspeakers which are not bi-amped.

For bi-amping, it is necessary to have (in addition to extra amplifiers) a crossover especially designed to divide "low level," or line level signals, before they are amplified. These crossovers are often of the active, or electronic type, in that they require AC power. The electronic crossovers divide the signal into lows and highs so that they can be amplified separately. A simpler passive low-level crossover can be made at a fraction of the cost of an electronic one, for the frugal. On the other hand, some would not be content with a two-way, or biamped system, but insist on a three-, four-, or five-way electronic crossover, with multiple channel amplifiers and multiple loudspeaker array. I wouldn't recommend the latter, unless you want to spend a lot of time trouble-shooting.

IMPEDANCE

No discussion about loudspeakers would be complete without mention of impedance. Impedance is to alternating current (AC) circuits as resistance is to simpler direct current (DC) circuits. Loudspeakers are operated by an alternating current, the output of a power amplifier. Loudspeakers are rated by the manufacturers as having a certain impedance value, usually eight or sixteen ohms. This is important for us to consider, since the power output of solid-state, or transistor type amplifiers is usually specified by the manufacturer as if the amplifier was running into an eight ohm load. In other words, an amplifier rated at two hundred watts per channel would deliver its maximum rated power when connected to an eight ohm loudspeaker.

Now, if you connect two eight ohm loudspeakers to the same amplifier channel in a parallel circuit (plus to plus, minus to minus), the resulting load is only four ohms, and the poor amplifier will try to deliver twice its rated power, or four hundred watts (200 watts to each loudspeaker). Well, some amplifiers will live with a four ohm load and some will overheat or blow fuses. Cooling fans help avoid melted amplifiers, but you should check the manufacturer's recommendations. Also, don't exceed the loudspeaker's rated power capacity, unless you plan to provide expensive smoke effects.

If you connect two eight ohm loudspeakers to the same amplifier channel in a *series* circuit (see figure 1), the resulting load is sixteen ohms. In this case, our two hundred watt channel would deliver only one hundred watts, or fifty watts to each loudspeaker.

For the guy who thinks tube amps sound better, I suppose he likes the sound of your horse and buggy, too. Oh, I suppose the clop-clop-clop on the cobblestones is more melodious than the sound of a motorcar. On the other hand, a horse has an exhaust occasionally, which is certainly odious, if not less musical. The point of this digression is that if both the tube-type and the transistor amplifier are just about to exhaust from being over-driven, the tube type has a less offensive clipping sound than the transistor type. The moral in this story is to have enough head room (power to spare) to avoid any audible clipping. Easy to do with transistors, not so easy with tube amps. As in the case of bass horns, this may be a matter of taste.

To the several technicians who pointed out my error in recommending one-and-a-half volts for phasing test (Part II of this series), thank you. Nine volt "transistor" batteries will not only make the loudspeakers move enough to see whether it moves in the right direction, but the spacing of the terminals on these batteries matches up perfectly with the speaker terminals, so you only have to touch the battery to them, not having to hold two wires in four places.

Another word of caution is now in order. All loudspeaker manufacturers don't subscribe to the "positive voltage on plus terminal produces forward cone movement" rule. However, this doesn't matter, unless you're using some of these and some of those, and even so, it is only necessary to be sure that *all* the loudspeakers in your system move the same way. By the same token, you don't have to remember whether the positive voltage terminal on the battery is the small one or the larger, so long as you hold it in the same way to test each speaker.

Holy voltage! I just remembered that the musician who wrote that letter (printed at the start of this article) has Altec-Lansing mid-range drivers and JBL (James B. Lansing) bass speakers. Now, you'd think that two companies with the same last name would be in phase, so to speak, but the fact is that when Jim B. split from Altec, he decided that positive voltage on the plus terminal should cause cone movement to-the-rear, march! It's possible that the bass and treble elements in the bi-amped system described in that letter are operating out of phase. This would result in some cancellation of frequencies in the crossover range, which might account for the lack of definition in the drum sound. The test, in this case, doesn't require a battery at all. Simply reverse the leads to the horn drivers, and listen! (The old A-B test.)

Theoretically, there will be some phase difference between the various elements of a multiple loudspeaker array if the voice coils are not all located in the same plane, so it is always a good idea to reverse the leads to the high frequency elements, anyway, and evaluate the results by ear, when setting up a new system for the first time. Also, crossovers introduce a phase difference between the low and high frequency elements, and unless this is corrected electronically in the crossover, it will add (or subtract) to the overall phase relationship.

If you think the phasing problem is difficult, consider that only recently has there been any agreement on how to assign the wiring of a balanced mic line, and in this case there are *three* wires to mix up, which gives the possibilities of *six* different configurations. Take heart, though most of us have now agreed that pin 1 is connected to the shield, pin 2 is negative, and pin 3 is positive. The manufacturers of microphones, however, have not all agreed, and some microphones provide a positive voltage when the first "T" of the familiar "Testing, one two . . ." reaches the diaphragm, and some give a negative voltage. If you use different brands of microphones together, you should correct any phasing errors that might occur. You can make a microphone phase-reversing adaptor from an XLR-3 female to male adaptor by reversing the wires between pins two and three. You can also make this correction to the microphone itself.

In conclusion, it is fairly obvious that the question most often asked about sound equipment is *How do I put this stuff together for the best results?* If you have read all three parts of this series, I hope you will have met with some of the answers, but it will also be fairly obvious that all of the answers have not been discovered yet.

the SOUND SH

Seasons Greetings! and welcome to NAMM/CES once again. I wish I could be there with you, but ole St. Nick has got me working overtime here in Memphis. Our business has been good here at Strings and Things, and I hope yours has, too. Provided, of course, that you're from Alaska and can't beat my deal! Anyway. . . .

AKG has a new stereo headphone system, called a "free-field" stereo headphone because of the use of six passive radiators that encircle the driver at 30-degree intervals on each phone. The philosophy behind this design is technical indeed, and comes as a result of extensive research by AKG into the nature of human hearing and the nature of recording. Without trying to write a textbook on the subject of aural phenomena, I'll try to get the point across.

The basic problem with most headphones is that they distort true stereo reproduction because they create sound images that seem to originate *in* the listener's head, at his left or right side, rather than from in front of him (as in a natural concert environment). Most everyone agrees that in this "natural" environment, various instrument sounds reach the listener's ears at different times, thus providing him with sound location information. Also, the "comb-filter effect" is a vital property of the human ear that permits further localization information by the auditory system. The comb-filter effect is a series of dips or notches in the frequency response of the ear which occur when the ear is in an unrestricted listening environment. Resonances at the outer ear, and in the ear canal itself, cause these notches to occur. The sound waves bounce around your head and ears, right? But when you put on a conventional set of headphones, the comb-filter effect is eliminated, and the directionality of the sound sources will not bear an accurate relationship to the original source.

Furthermore, recordings are engineered to create natural separation when played back through loudspeakers, not headphones. Compensations are made in the master recording for the listening environment, where a sort of "acoustic mix" occurs (the channels run into each other in your living room). So stereo separation is increased during recording to compensate for the reduced audible separation that occurs during playback through loudspeakers. All well and good, until you put on the headphones. Then, the acoustic mix of the room is pre-empted, the comb-filter effect is lost as well, and the result is a very unnatural sound through the phones.

AKG circumvents this problem by the use of the six passive radiators on each side of the headphones. These radiators are transparent enough to let the ear listen in a virtually "free field," thus preserving the comb-filter effect necessary for natural reproduction.

This is only a thumbnail sketch of AKG's explanation of the K-240 headphones, but it should serve to indicate the kind of thought and research that goes into their product. I have been selling AKG microphones for a long time with no problems at all, and while most

OPPE

By Charlie Lawing

of my customers are performing musicians rather than audiophiles, it is nice to know that AKG has such a product available. Stereo buffs probably will be happy to know this, too, so pass the word along. If the customer wants a dynamite set of headphones, sell him a set of these. If my attempt at explaining the K-240's isn't enough, just contact AKG; they'll talk your ears off, free-field or otherwise.

CIRCLE 1 ON READER SERVICE CARD

Wherever you are from, surely you have heard of **Ibanez** by now. This distinguished company burst into the musical merchandise market only a few short years ago, and has already managed to produce a distinguished line of electric guitars, both solid and hollowbody, as well as a line of effects (or "stomp boxes" as they are called by some of the 'good ole boys' in Tennessee). And I am told, although I haven't actually laid hands on them yet, that the new Ibanez acoustic guitars are outstanding (this info coming *not* from Ibanez, but from another dealer nearby whose judgment I respect).

One particular Ibanez product that I *have* seen, though, is the AD-230 Analog Delay and Multi-Flanger. The list of professional endorsees of this unit extends far across the spectrum of contemporary music, all the way from Steve Miller, the Eagles, Jerry Garcia and Bob Weir to Billy Cobham and George Benson. The diverse styles of these artists bespeaks the versatility of the AD-230, I think, making it immediately obvious that the product can (and *is*) being used in a variety of situations, both live and in the studio.

The AD-230 is an analog unit, as opposed to a digital one, and I'm sure most of you know that this means several things in general: it is a little noisier than a digital unit; and it is considerably less expensive than a digital unit. However, this is a surprisingly good delay

unit, as analogs go, and Ibanez assures me that the signal-to-noise ratio is studio quality. The delay range is continuously variable up to 600 milliseconds, which is plenty of room for most of the desired delay effects that musicians want. Delay time, speed, width, and regeneration controls (all rotary) are also a part of the delay features of this unit. Both the delay level and the input sensitivity pots are connected to ladder-type LED indicators that are brighter and easier to see than most VU's, thus making the job of monitoring the levels on the AD-230 easier for the sound technician at the control board, or the musician on stage. LED indicators are also used to indicate the status of the output functions, whether normal, delay, or flanger.

The Multi-Flanger section of the AD-230 is pretty much standard fare as flangers go; delay time, width, speed, delay level, and regeneration are the variables over which the user has control, each by way of a rotary pot. I thought at the time I heard this unit that the Flanger was noticeably quieter than competitive units, and I find this a big factor in selling any and all effects. Contrary to what critics of rock and roll might like to to think, musicians don't want to make noise, they want a clean sound.

The AD-230 is a clean unit; it is 19″ rack mountable, and equipped with both ¼″ phone jacks and 3-pin cannon connectors. The unit weighs 19 lbs.

CIRCLE 2 ON READER SERVICE CARD

The word "crossover" can mean several different things, depending on which end of the music business you are in. To the audio specialist, a "crossover" is an electronic network that separates bass from treble frequencies in a sound system and directs those isolated portions of incoming signal to the appropriate speaker or horn. To the record company exec, "crossover" is what happens when a particular artist goes beyond his/her usual audience and begins to appeal to a wider range of people. Needless to say, the latter definition of the word as it applies to the music industry is the more glamorous and publicly acknowledged of the two, but just let me say in defense of the folks who design and build the electronic crossovers that without them, nobody would sound worth a flipping red cent, and there are no Grammy Awards waiting for those guys at the end of a hard year's work, either. It is sort of ironic that although Marshall is an amp manufacturer and not a rock group, they are trying to make a "crossover" in the best tradition of Warner Brothers, RCA, Elektra, or MCA.

Marshall amplifiers have been the undisputed leader in the rock music world for a number of years. They are made in England, but the American artists such as Jimi Hendrix, Billy Gibbons, Duane Allman, and a host of other guitarists in this country made them famous. (That is not to slight the British players like Beck and Blackmore and Pete Townshend who used the amps also, but the Beatles used Vox amps, so I defer to the Americans).

Anyway, the point is, Other fine amps notwithstanding, Marshall has been on top in rock circles for over a decade. But notice that I say "rock circles" specifically, because Marshalls were not popular among country or jazz musicians. The old Marshalls had to be cranked up, and there were no provisions for reverb or a smooth, "mellow" sound. There was no polarity switch. I have always supposed that the reasoning behind the absence of a polarity switch was that Marshalls were always played so loud that it didn't matter if it hummed or not, nobody could hear it, anyway. This did not hinder Marshalls on the concert stage, where sharp audio technicians could eliminate unwanted hum through electronic means, but it posed a problem to circuit musicians whose only recourse was to reverse the power cord going into the wall socket.

So, realizing that although their amps were a success, they were limited, and also probably realizing how well musicians were responding to competitive lines of smaller amps, Marshall designers came up with the 4140 "Crossover" Amp, which seeks to combine the best of both worlds, loud and nasty vs. soft and clean.

The new Marshall 4140 is a 100 watt (RMS) 2 × 12″ two-channel amp, and as you probably already have guessed, it has reverb! This is the equivalent of Black Sabbath releasing a gospel album! But it's true, and it is a versatile, compact, and attractive amp that does offer something to the people who don't play rock music, as well as those who do. The 4140 has plenty of power, and it has an overdrive circuit (a master volume that can be used in conjunction with a boost switch on Channel 1). Individual channel volumes are provided; bass, mid, and treble pots for both channels; high and low gain inputs on both channels, and a bright switch on channel one round out the front panel control plate.

The 4140 approximates the traditional Marshall sound well enough to satisfy my ears, though some die-hard users of the big stacks will disagree. The point is, the amp is not designed to replace the old Marshalls (nothing *can* replace them; they are an integral part of so much of what we call "rock" music); the 4140 is intended to complement the Marshall line, and to broaden the appeal of Marshalls in general. I think the amp achieves that, and I think it comes at a good time, a time when musicians are turning away from the big stacks (except for creating the illusion of tremendous power on stage) in favor of smaller amps and better sound reinforcement systems. At least that is the way I see it, and I think many dealers will agree.

CIRCLE 3 ON READER SERVICE CARD

Peavey Electronics Corporation has introduced a new line of monaural mixing consoles, the Mark 1 Series. Features include variable input gain, monitor send, 2-band EQ, effects send, and level sliders on each channel. The Master section features controls for high and

low EQ, effects level, reverb return, reverb contour, and effects return, along with main and monitor output sliders. Rear panel includes line out, high and low inputs for each channel; main, monitor and effects outputs; main, monitor and effects auxiliary inputs; and effects return.

CIRCLE 4 ON READER SERVICE CARD

A while back I mentioned the Crumar organ as an alternative to the bulky Hammond B-3. Portability is a big factor among musicians these days, especially keyboard players who are faced with the prospect of squeezing electric pianos, clavinets, organs, synthesizers and string machines into their budgets, and into already overcrowded vans, bob trucks and U-Haul trailers. Even if the keyboardist-at-large trades that Hammond in for a smaller Crumar or Yamaha organ, odds are he is not going to give up his Leslie tone cabinet, because nothing else sounds like a Leslie, not even the best phaser or flanger on the market. With one notable exception: the **Multivox** MX-2 Full Rotor.

The MX-2 is a small unit, weighing only 3½ pounds, that can be packed into an Anvil briefcase with ease, and it sits on top of most any keyboard, almost unnoticeable from offstage. Yet the MX-2 delivers the truest "Leslie" type sound of any special effect I've seen.

Users of the conventional Leslie know that it has two speeds, and that when you accelerate or decelerate, it happens *gradually*, because it is motor-driven, and it takes a little time for the motor to speed up or slow down. This is one of the nice things about a Leslie, and many keyboard players have learned to use that gradual change in speed for dramatic musical effects. Such is not the case with most phasers and flangers, which change from a "stop" position to a preset speed by means of a foot switch. Of course, you can gradually increase or decrease this speed by rotating the appropriate pot, but you sacrifice the use of one hand in the process. Combine this limitation with the fact that many keyboard players find phasers and flangers to be poor substitutes for the "Leslie" sound.

The MX-2, although it is electronic rather than mechanical, does speed up and slow down *gradually*, and many of my professional keyboard customers cannot perceive a noticeable difference in the sound of the MX-2 and a Leslie. Furthermore, the MX-2 has two inputs, thereby enabling the keyboard player to use it to good advantage on an electric piano, string synthesizer, or a clavinet as well as on the organ. With the optional foot switch, the keyboard player never has to take his hands off the keyboard to change speeds or stop the full rotor. The MX-2 also costs a whole lot less than a Leslie.

CIRCLE 5 ON READER SERVICE CARD

By Garrett Donaldson

DEALER DOSSIER

K&L Audio
Watertown, Massachusetts

During New England's first snowstorm of the year I made my way to K&L Audio on North Beacon Street in Watertown, one of the mini-suburbs of Boston. Following a sign announcing "showroom entrance," one finds a rather austere entry, a glass display counter directly ahead filled with tapes and accessories of the usual sort. To the right a corridor holds a utilitarian 4′ × 6′ display outlining the floor plan of K&L. The area to be concentrated on is on the second floor: Professional Sound. On the landing between floors one and two looms a superhero figure with a megabelt featuring "K&L" girding his waist. Rather pop in its appeal, this fellow adds a dimension of power to the entry of the Pro Audio Department.

Entering this department we meet again an austere display case; nothing fancy, just featuring goodies and gadgets for the serious performer. Les Arnold, the manager and chief salesman for the pro audio department, and Art Williamson, the service engineer, showed me around. Displays include an API custom mixing board set up for mixing and matching any number and combination of power amps (Uni-Sync, BGW, ESS, Cerwin-Vega, Tapco), mixers (Tascam, Soundcraftsmen, Sound Workshop, Uni-Sync, Tapco, Malatchi), and speakers (Forsythe, Cerwin-Vega, Community, Gauss among the products handled by K&L). Lewis Freedman, the main honcho at K&L, arrived and we got down to business.

How did this operation start out?

Freedman: It started in May of 1971 in an apartment on North Beacon Street. After two years we moved to an old gas station. And in 1974 we moved to our current location, which is now an 11,000 square foot building. In fact, when we moved in here we had 7,200 square feet. We later added on to that, in 1976.

I had a friend at Cerwin-Vega named Gayle Martin who has been a real good guide to me. He kind of forced us into the professional side of the business. He said, "Here, take a few of these; you can sell them." And we were amazed. We could sell them. It wasn't easy, but the stuff was definitely sellable.

Our first professional sale was to Gary Burton, who bought his first PA from us. In fact, about a year later he came back. We still weren't established in pro audio. Burton bought another one. This was the spring of 1975. Then someone else came in and said he wanted the same thing that Gary Burton had bought. And I said to myself, "We're getting known!" I was kind of amazed. Then we did some business with Duke and the Drivers, which is another major Boston group. They bought some Vega bass cabinets. At that point we were the only Vega dealer in Boston and were concentrating on Cerwin-Vega and doing our thing.

In the fall of 1975 Ken Berger dropped in to buy some hi-fi equipment and asked me if I was considering doing pro audio. I'd been tossing with the idea for about three or four months. The potential seemed to be there. About two months later I called Ken Berger and said, "Hey, come on in; let's give it a shot." He did part-time pro audio and part-time hi-fi. I then called on Ken Forsythe, who had been my partner originally, and who had been out of the picture for about six years. He had been into pro audio more than into hi-fi—which is the

reason, I believe, that he left. I said "Hey, Ken, we're going to do a pro thing. Are you interested in managing it?" And he came back on board. At that point he had designed some cabinets, bass cabinets beyond just about anything in the field. And K&L Pro-Audio was born—around Cerwin-Vega and Forsythe sound cabinets.

Cerwin-Vega power amps started to make the thing jive. Then we picked up BGW and Community Light and Sound, Gauss, and a couple more mike lines and some other stuff and put the whole thing together. Since that time we've just been adding lines and adding more and more and doing more and more different types of jobs to the point where in 1976 pro audio had outgrown the 10×14 room they had downstairs on the first floor and were desperately in need of space.

In 1976, we were bulging at the seams in pro audio. We had done a lot of group work and a lot of small rock acts, but nothing really major because we weren't capable of it. Our showroom facilities were lacking so much. I then decided it was time to buy the building, which I had an option on since the day I leased it, and decided to add a second floor. We were operational for pro audio around the first of February, 1977, which increased pro audio's selling space from 140 square feet to almost 1,500. Which has made a difference of day and night.

We set up separate rooms for recording, disco, and reinforcement. Now we are contemplating one more renovation, because we have moved our esoteric hi-fi department out of the building into a store. We are going to take one of their old rooms and renovate it into a full-scale discotheque. It will be a disco showroom—with floor-lighting, stuff hanging from the ceiling. Just like the real thing.

When do you plan on doing that?

Freedman: I'm hoping to have that completed by April or May of 1979. Right now I would say that the disco end of our business is the fastest rolling, although we're still doing a lot of sound reinforcement and a lot of semi-professional recording.

Do you have any fear that the disco thing will fade out?

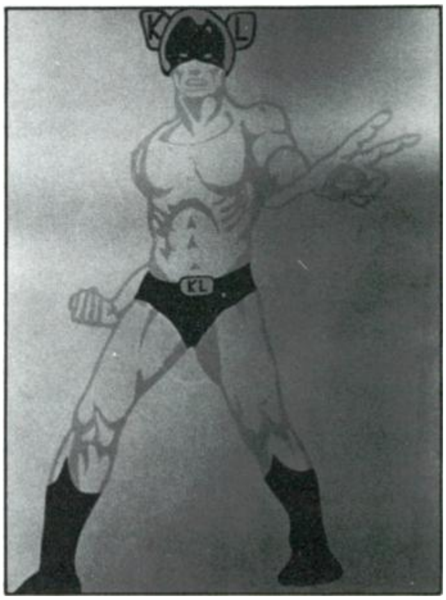
Freedman: The disco thing started to slide about a year and a half ago, and was in a downturn for about eight or nine months. Then they released *Saturday Night Fever, Grease,* and a few other things. And the thing just took off to the moon. We have more disco work right now than we know what to do with. We've been doing club after club in Boston.

How would you separate pro audio from consumer audio?

Freedman: Pro audio is the person who makes his livelihood from his equipment. Consumer audio is the type of person who buys the equipment to enjoy it. Ninety-nine percent of our customers depend for their livelihood on what they buy in our pro audio showroom. Therefore we offer a lot of extra services: free loans while the stuff is in the shop; extensive demonstration facilities; extensive warranty and service facilities. When the customer comes in here, he feels like he is going to be taken care of—as a person, as a good customer. It's very rare for us to ever lose a customer in pro audio. Most of the time, the customer comes in here, he sees the way we treat him, he likes it, enjoys it, and gets well taken care of.

Why go pro when there is still so much consumer hi-fi?

Freedman: It's a logical step. If you know anything about the history of K&L, you see diversification right from day one. K&L is like a mini-conglomerate of different areas of hi-fi. The consumers say we not only have a "normal" type store, which is regular hi-fi, car stereo, etc.—systems up to about $1,200 or $1,300—we also have a separate esoteric department, called the Audio Forum, which is in a separate location and specializes in the real audiophile-type gear—the really, really fine stuff that's made today. We also have a tremendous campus business. We sell reps on the college campuses across the entire country. That's a department that started with K&L almost from its infancy—something I created back in 1972. From there, we also got into a department where we actually have people out on the road calling on radio broadcasters.



Pro audio, of course, was started in the fall of 1975, and has grown tremendously. It started as one person part-time and now has three people full-time and three people part-time, along with a clerical staff and everything.

Speaking of help, how do you hire?

Freedman: Basically, the majority of the pro department has a lot of experience in sound. I met Les Arnold six or seven years ago, when he was with Delta Sound out in Worcester. In fact, most of our crew came out of Delta Sound. When Delta Sound folded up, I grabbed on to whatever people I could get from that operation.

Because of their extensive background in sound, we hired them. Most of the people who work in pro audio actually have a lot of hands-on experience and know a lot about the gear.

Art Williamson, I would consider a

ROLAND
MEANS BUSINESS.

LOTS OF BUSINESS.

All of the equipment pictured is Roland. It's part of the most impressive line of musical equipment in the world.

It's not just one line, but many lines. For example, Roland offers 12 models of guitar amps . . . signal processing equipment . . . combo organs and pianos . . . synthesizers . . . sound systems . . . Boss effect devices . . . rhythm units . . . the GR-500 Guitar Synthesizer . . . the MC-8 MicroComposer . . . you name it.

Roland is growing. And so are Roland Dealers. That's why RolandCorp US makes it easy to get started. Chances are there's a dealership plan just right for your business. After all, without your business, where would we be?

RolandCorp US
2401 Saybrook Avenue
Los Angeles, CA 90040
(213) 685-5141

For information about a Roland Dealership contact Tom Beckmen, President, RolandCorp US, 2401 Saybrook Avenue, Los Angeles, CA 90040 or phone (213) 685-5141.

super technician. He has straightened equipment out that we were told to junk. The guy has been just amazing in that respect. Whenever there's a problem this guy can find it.

Tell me about New England. Why New England?

Freedman: Because I'm from New England.

How would you view the New England market?

Freedman: The way things are done in this market doesn't make sense.

What would you say would have to be done in this market to make it as competitive with New York for instance?

Freedman: We are competitive with New York. Because of the services we offer. That's one thing you don't get in New York. You get pricing in New York; here you get both pricing and service. But distribution of lines in this territory is very, very difficult to read. In fact, things are so screwed up, they don't make any sense at all. If one guy gets in with a line it's like he's married to the line. If you get somebody new on the block, the guy gets frozen out. That's kind of strange. If the new guy on the block isn't capable of doing the job, fine; and if the old guy is doing the job, that's great, stay with him. But if the new guy is doing the business and going in different areas and not competing, give the guy a chance. Give the guy a line. You have to be careful. There have been times when conflicts arise. But I feel that all the political stuff that's going on right now with lines is absolutely ridiculous. We're probably the biggest disco installer in New England, but I don't have the biggest speaker line. It's the most ridiculous thing I've ever encountered in my life. We use a load of them, but the manufacturer won't sell to us direct. They insist we'll compete with their prime MI dealer—which we wouldn't. And to me, that hurts. It makes us try and find other products—which we have. We've gone so far as to import stuff from England.

How about the customer? What is the central theme when he comes in to K&L Audio's Pro Sound Department?

Freedman: What we try to do, basically, is to make sure that whoever goes in and out of here gets the proper equipment. They don't get what they don't need. They don't get too much or too little, they get what they need. Not anything more, not anything less.

Arnold: We have so many lines available, we just put together the right package.

Freedman: We've seen so many packages that other people have put together. We generally are under them by $1,000 or $2,000, not because we're giving anything away, but because we're using less of it to do more work. When we do a job we *engineer* a job.

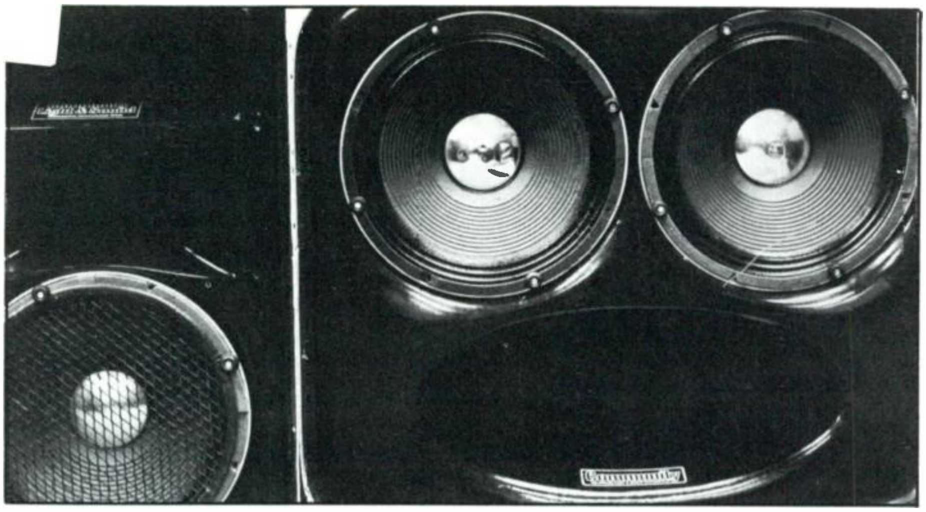
How much more does a customer know today—as opposed to five years ago?

Arnold: Five years ago even the guys in the sound business didn't have that much knowledge. First of all, the equipment wasn't there that is there today. You couldn't pick or choose. It was a stack of Shure mixers and some Voice of the Theatre cabinets. And that was it. Now there is so much equipment that you can narrow it down to exactly what you need. And customers are becoming more educated. We try and do that a lot. We

talk to people about what they are buying, why they should but what they are buying. We work with them when they get the equipment. In general, the customers' ears have developed. They know what they are listening to now.

The manufacturers and the merchandising departments are a hell of a lot more effective than years ago. Consumer awareness is like a day and night situation. You've got real professional magazines out there. That has made the semi-professional user more aware of what is also available on the professional side.

The gap between pro and semi-pro is shrinking; people are getting more and



more oriented towards the better equipment.

People are learning to make music as opposed to consuming music? Do you think that market is getting smaller.?

Freedman: It's hard to say, although I will say this: I think the high fidelity industry on the components side—as far as the independent retailer is concerned—has come close to peaking out in the last couple of years. This is the first year I've ever experienced a very, very small growth in our retail sales. However, I'm not sweating it too heavily, as our pro side has grown so tremendously. We have done some really heavy jobs in the last year.

What percent are disco?

Arnold: It's getting larger all the time.

Freedman: We'll break it down into disco, recording, and sound reinforcement. Disco at this point is about 60 percent of our sales. That primarily is the club business and the portable business. Sound reinforcement covers roughly another 25 percent and recording covers 15 percent. Of course, there are overlaps in all of those areas. Disco itself is becoming such a big deal. And not trailing off like a lot of people thought it would.

What would you say is the heavy time of the year?

Freedman: The heavy time for disco is through the summer. Sound reinforcement is heaviest going from March to the end of July.

What about recording?

Freedman: Recording is year-round.

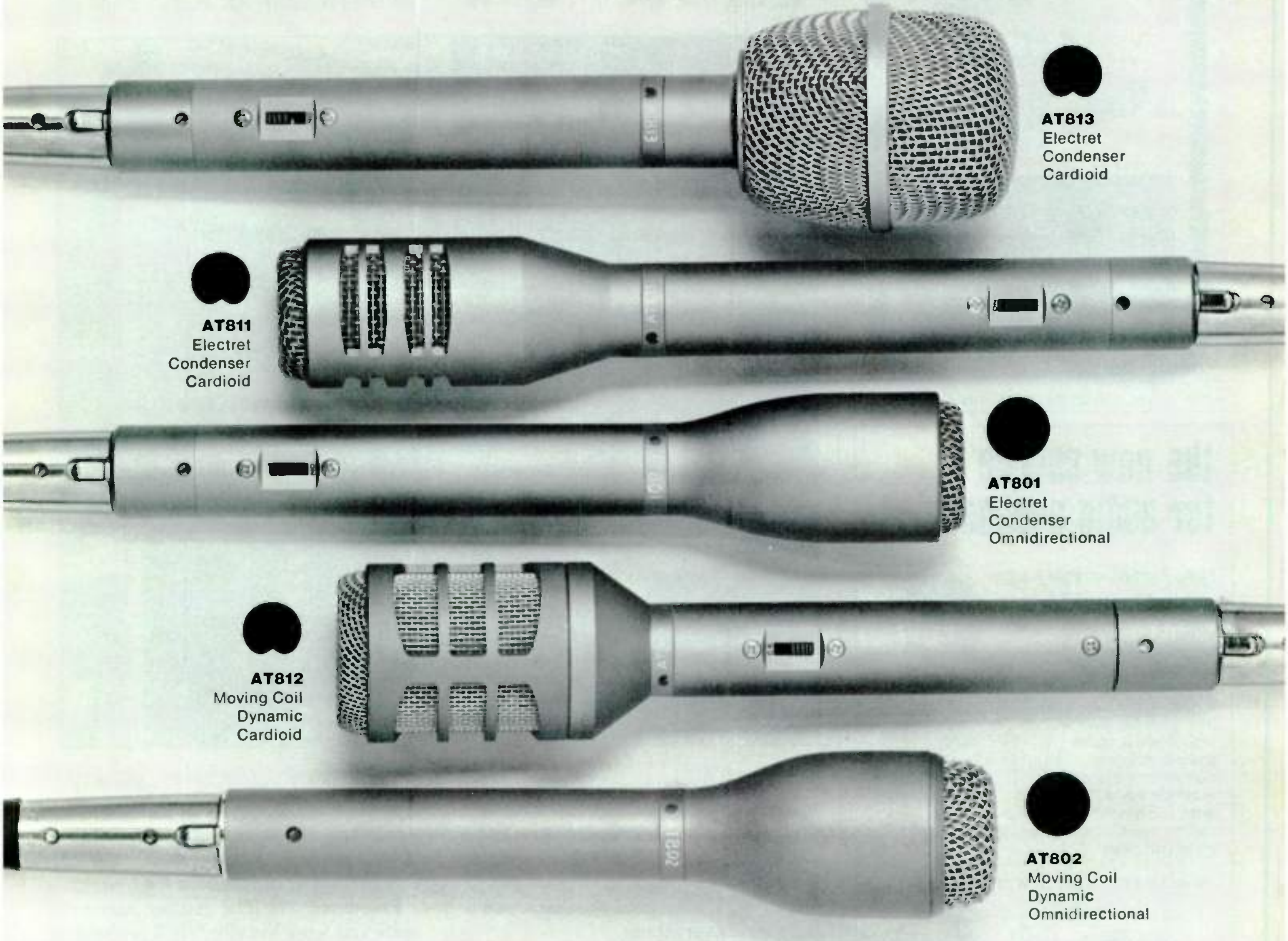
How much touring sound do you do?

Freedman: We're not really into touring sound. We've done some jobs. We have the equipment, we just haven't really pursued it. We don't have the vehicles. We have the people, we have the knowledge and we have the equipment. But we have never put together the transportation or the packaging. That's one of the next things we will be tackling.

When do you start putting together your inventory?

Freedman: We don't order for the season, that's the amazing thing. The only thing that's really seasonal is the mixer. Not the recording mixer, but

Audio-Technica introduces five new microphones... and a pleasant surprise.



Take a close look at these new Audio-Technica microphones. Three electret condensers and two dynamics. Plus two clip-on miniature electrets (not shown). All are superbly finished. Carefully thought out in every detail. With the right "heft" and feel. Professional A3M Switchcraft output connectors, of course.

Then listen in your studio. Full-range, peak-free, clean and crisp. With no distortion even when used close-up to high-level performers. And the balanced, phased Lo-Z (600 Ohm) output matches pro and semi-pro mixers alike.

Now for the surprise. The price. Both omnis are nationally advertised at just $60, for either dynamic or electret condenser element. The two basic cardioids are just $80, while the AT813 electret condenser with integral windscreen is pegged at $95. All complete with full one-year warranty.

Once you've seen and tried these new Audio-Technica microphones we think you'll welcome them. Not just because they cost so little...but because they do so much. Available now from your Audio-Technica Professional Products dealer.

audio-technica. ***Great sound, right from the start!***

AUDIO-TECHNICA U.S., INC., Dept. 29SA, 33 Shiawassee Avenue, Fairlawn, Ohio 44313 • In Canada: Superior Electronics, Inc.

the sound reinforcement mixer market, the disco mixer market. Those are seasonal.

When a band comes in here, and they really need a good system because they are going to do a showcase, what can you do for them for a night in terms of $500? Or for rental? Is $500 a lot for renting for one night?

Freedman: We have a formula that we use. If we're sending a crew out with it, it's a flat $75 per man, per night. There are also pickup and delivery charges. If they're picking it up and dropping it off, that's a different ballgame. For $500 in a night you could get $10,000 worth of equipment; you can do an awful lot with that. You may not have the greatest system in the world, but you'll have a more than adequate one. You'll have good power amps, you'll have good speakers, you'll have good everything.

Do you give out a private number?

Freedman: Some of the guys in the department actually have given out their numbers to customers and said to them if something happens, give us a call, we'll be more than happy to take care of it. As far as the disco side, we have two guys on call 24 hours a day.

What about shrinkage? Have you had much problem with that?

Freedman: In the pro side? With the crew I've got right now, I don't worry about it. With the crew in the past I don't think the shrinkage was very high; there might have been some problems but I think they have been pretty well solved. We have pretty good morale in the department. People are really into doing what they're doing. They enjoy what they're doing and I'm pretty much there for management only. I'm not there to tell them what to do, because — to be blunt honest — I enjoy pro sound, I really get off on it, but I'm not as knowledgeable as most of the salesmen who are in there on the pro stuff. Although I could go in there and I could sell. I could diagnose stuff a hell of a lot quicker than a lot of other people can. It's all pick-me-up education.

What will be seeing from K&L next?

Freedman: We'll be getting into pro more and more heavily. We'll probably be moving the heavier stuff into the pro recording side which we held back on for the last year or so.

What would you say that the professional side of recording in New England looks like?

Freedman: To be blunt with you, it looks like a nut house. You have all these semi-pro guys who are springing up, and each and every one has their eyes set on a studio.

INDUSTRY UPDATE

Scientific Audio Electronics (SAE) has made several staff appointments. **Warren Pompei,** formerly plant manager, has been promoted to Director of Sales. **Andrew McKinney,** formerly with the rep firm of Jack Carter Associates, has been named National Sales Manager. **Barry Thornton** has been named Chief Engineer. He is the founder of the Quitessence Group and has managed the PA/sound support systems for entertainers including Jethro Tull and Joe Cocker. **Don Jackson,** previously a systems engineer for Express Sound Company, has been named Professional Products Applications Engineer.

ARP Instruments, Inc. has acquired the Mu-Tron product line of sound modification devices formerly manufactured and marketed by Musitronics Corporation. According to **Bob Hoffman,** ARP Vice President-Marketing, ARP has purchased all trademarks, trade names, and rights to manufacture and market the line. The Mu-Tron effects line will be marketed through ARP's established network of certified dealers.

A. Dransfield and Company Limited of Hong Kong has invested $1 million in Bertagni Electroacoustic Systems. The investor received the rights to distribute and produce B.E.S. loudspeakers throughout the Far East.

Altec Lansing Sound Products Division has appointed **Allan L. Anderson** District Manager for the company's southwestern territory, responsible for the marketing and technical application of the commercial sound product line in southern California, Nevada, Arizona, New Mexico and the El Paso area of Texas.

James W. Stonebridge has joined Koss Corporation as Senior Buyer, responsible for purchasing component parts, packaging, plastics, adhesives, literature and tooling for the company's stereophone and loudspeaker products.

Emil Handke has been appointed Sales Manager of Sound Workshop. Handke was previously sales manager at dbx, Inc. He will be handling both the domestic and export markets for Sound Workshop. Michael Tapes continues as Director of Marketing and Paul Galburt as Chief Design Engineer.

The Institute of High Fidelity has named several new officers and board members. **Jon Kelly,** President of Audio-Technica U.S., has been appointed Vice President of the IHF, filling a vacancy created by the resignation of **Allen Novick,** formerly of TEAC. **Edgar W. Hopper,** Publisher of Stereo Review Magazine, was named Secretary of the IHF, filling a vacancy created by the resignation of **Kenneth Busch** of Empire Scientific Corporation. Named as Directors of the Institute to fill vacancies left by the appointments of Kelly and Hopper are **Jerry Henricks,** Director of Sales and Marketing, Hitachi Sales Corporation and **Bernie Mitchell,** President of U.S. Pioneer Electronics. All appointments are interim until April.

The IHF has also appointed Jon R. Kelly Chairman of the International High Fidelity Convention Committee. The Convention is scheduled for April 20-22 in St. Louis.

Jay Simmons has been named Field Sales Manager for Uni-Sync, Inc. Simmons was previously professional products sales representative with B&B Electronics in Denver. He replaces **Wayne Freeman** who has joined the rep firm of Carlile & Associates.

KLH has made several staff changes. **Scott L. Davis,** previously KLH President, has returned to the corporate headquarters of the parent company, Electro Audio Dynamics, as Vice President of Corporate Development and Legal Affairs. **Denis A. Wratten** has been named Executive Vice President of KLH, moving from another EAD subsidiary, Infinity Systems, Inc. **Frank Jones** has been named Vice President of Product Development of KLH. **Robert A. Coppola,** formerly Executive Vice President of KLH, has been named President of a new EAD subsidiary in Europe, intended to "strengthen the international business of EAD's domestic subsidiaries and to develop trade opportunities for the entire EAD group."

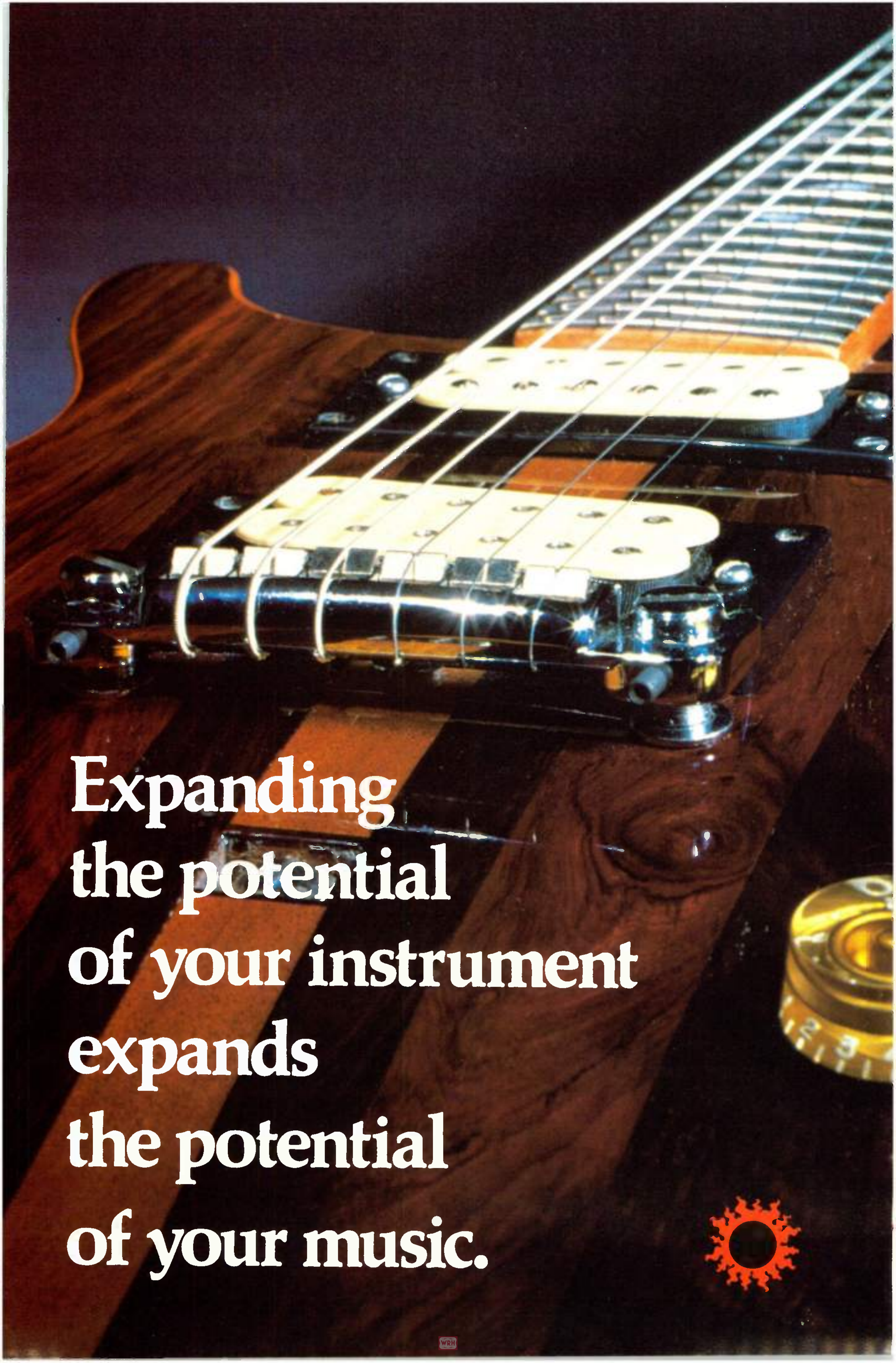
Ampex Corporation has upgraded its magnetic tape marketing effort in the Pacific Northwest. **Gary Pointon** has joined the company's magnetic tape division as consumer sales representative for the Seattle area. Two new independent representatives have been appointed for the Washington and Oregon territory. Walker Audio Associates will represent the line for high fidelity audio outlets, and J. Vahl Associates for mass merchandise outlets.

TDK Electronics Corp has made several changes in personnel and responsibility. **Ken Kohda** has been moved up from marketing manager to Vice President and Assistant to President **Sho Okiyama. Bud Barger,** formerly eastern division sales manager, assumes the post of National Marketing Manager. **Sandy Cohen,** formerly western division sales manager has been named National Sales Manager. **John Schattin** has been promoted to Eastern Regional Sales Manager.

Jason Farrow has joined Sony as Advertising Manager for high fidelity and digital audio products. He was previously marketing communications manager for Harman Kardon and director of public relations for Acoustic Research.

John Pecoraro has been named Western Regional Sales Manager for the audio division of Akai America, Ltd. He was formerly sales manager for the west coast office of Lafayette Radio.

Advertiser's Index


Expanding
the potential
of your instrument
expands
the potential
of your music.

. from DiMarzio.

Model I™ Bass Pickup

The Model I fits directly in EB-0 basses, as well as in the bass position of the EB-I, 2, and 3. It produces the sound the owners of these basses have always wanted—powerful, yet crystal clear. Every note down to the lowest can be heard distinctly. A miniature switch is included with the Model One. Installation of this switch lets the Model One produce two distinct sounds. One is similar to that made by the Fender Precision, while the other has a Rickenbacker-type sound. The switch is optional; it can be omitted, with either sound available alone.

Acoustic Model II™

The Acoustic Model II is a humbucking, magnetic pickup for steel string acoustic guitars. A simple, sliding rail arrangement gives the Model II the unique ability to tune any acoustic guitar for harmonic balance—the player can emphasize overtones, or fundamentals, in any proportion he chooses.

DiMarzio Model P™ Bass Pickup

The DiMarzio Model P offers several major advances not previously available to Precision players. It has adjustable pole pieces for each string. It has a magnet structure which eliminates distortion and double-tones, and yet has noticeably increased output. The effect of this is to give the bass a brilliant sound, with punch and clarity on every note of the bass.

Accessories

Complete humbucking pickup hardware. 500K and 250K potentiometers. Five-way pickup selector switches. Double-throw miniature switches, identical to those included with our Dual Sound and Model One Pickups. Brass Hardware—Solid brass tunamatic bridges, Stratocaster replacement bridge pieces, brass nuts for Fender style guitars and basses, and Gibson-style guitars. Combination acoustic jack end-pin. Fret Wire. Creme neck and body binding material. Electric Bass strings in round and flat wound sets.

Acoustic Model™

The Acoustic Model is a high-efficiency contact pickup which can be used with round hole guitars, as well as banjos, mandolins, and upright basses. Low noise and high output make this pickup outstanding—no pre-amp is required.

DiMarzio®
Musical Instrument Pickups, Inc.

1388 Richmond Terrace Dept. MRBC
Staten Island, NY 10310 USA
(212) 981-9286

CIRCLE 97 ON READER SERVICE CARD

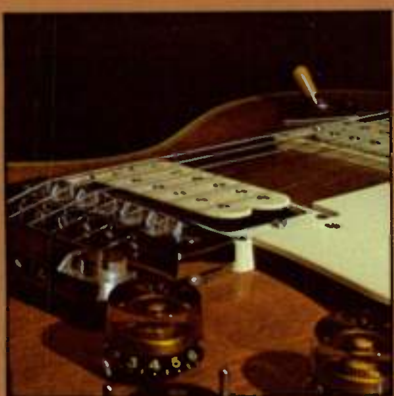
Creative Expansion

Super Distortion™ Humbucker

The SDHP is the pickup that first put the word 'hot' in the guitarists' dictionary. For the rock guitarist, the SDHP does it all —at full volume, it provides the most powerful, sustained sound your guitar is capable of. At lower volume, the SDHP becomes clear and warm, useful for all playing styles. This is accomplished without squeal or unwanted hum, and without any loss of frequency range, as is common in other "hot" pickups.

PAF™

The PAF is a special pickup, with a special sound. It is the sound of the "patent applied for" pickup made in 1957-1960. This sound is not one of sheer power, but of a warm tone, and a singing sustain. It has only been available until now to the limited number of players able to acquire old instruments.



Dual Sound™ Humbucker

Everything written about the SDHP is also true of the Dual Sound—it's the same pickup, with the added option of a second sound, available at the flick of a switch. This allows the guitarist to go instantly from the sound of the Super Distortion to a clean, bright sound similar to that found in Stratocasters and Firebirds. No increase in noise is experienced, and, unlike other companies' 'split coil' pickups, the Dual Sound pickup is humbucking in both modes.

FS-I™ & Pre B-I™

The FS-I & Pre B-I are direct replacement pickups for Stratocasters and Telecasters. The Pre B-I is made for installation in the bridge position of Telecasters, as shown in the insert photo. The FS-I can be used in all positions on Stratocasters, but is especially recommended for use in the bridge position. Both pickups mount with no modification to the guitar. The FS-I and Pre B-I enhance the basic qualities of the guitars they mount in by offering increased output and greater midrange response.

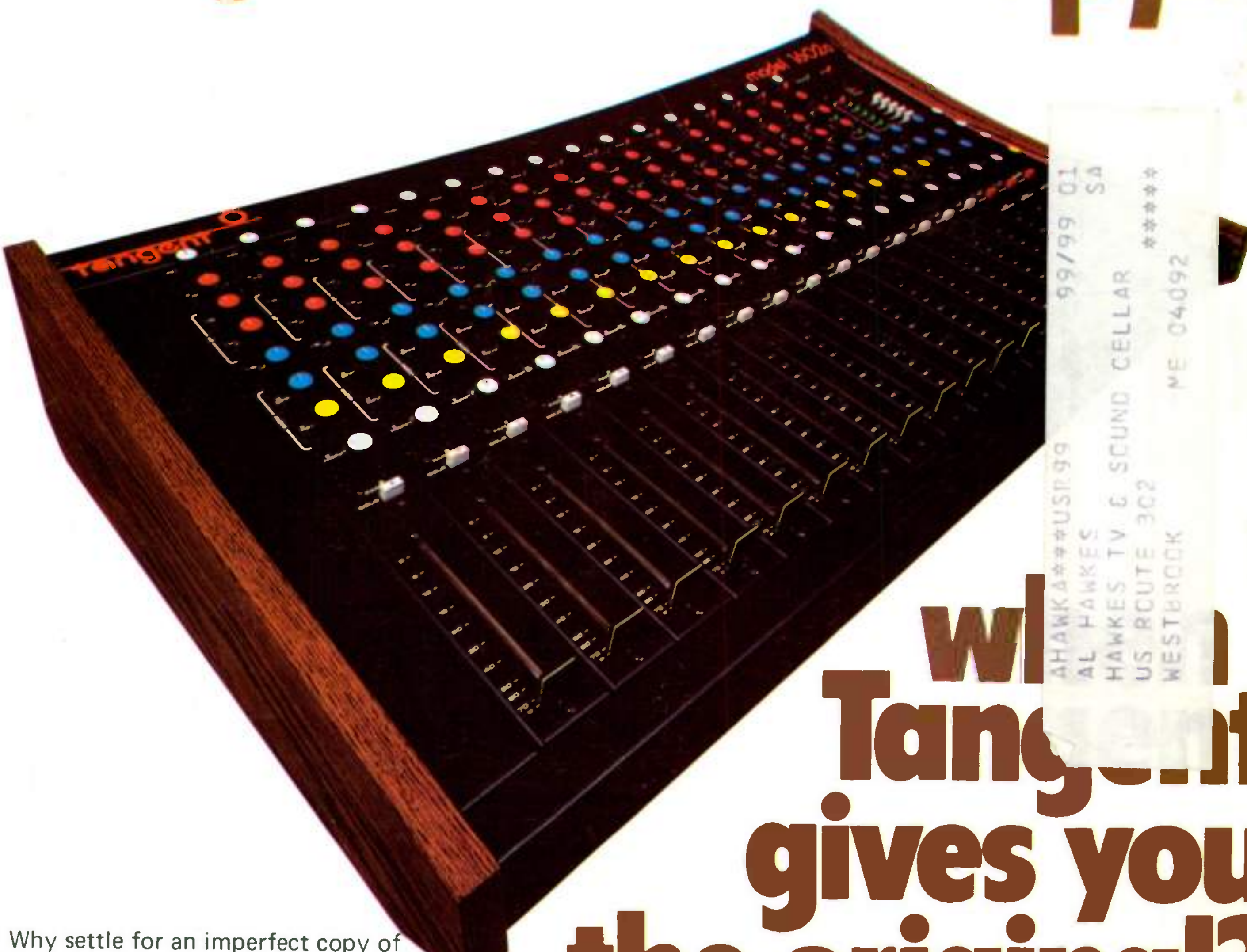
Super II™

Any guitarist who has had trouble cutting through, or has a guitar that sounds dull, will appreciate the Super II. It offers slightly less output than the Super Distortion, but has increased treble response. Combining this performance with low cost gives a pickup ideal for guitarists looking to upgrade their present equipment.

SDS-I™

The SDS-I is a Stratocaster replacement pickup like nothing you've ever seen or heard before. It is the first Strat replacement pickup with fully adjustable pole pieces. It is the first pickup of its kind powerful enough to let Strats compete with the hottest guitars on the market. The power of the SDS-I is not weakened by the double tones and out-of-tuneness common to other Strat-type pickups, because it has 50% less string pull.

Why settle for a copy...



when Tangent gives you the original?

Why settle for an imperfect copy of your sound, when Tangent will give your audience the original?

Tangent's crystal-clear transparency allows your original to flow cleanly to your listeners, with only the coloration that **you** add.

And beyond this foundation of solid quality, Tangent's "a" series mixers give you these features found previously only on recording consoles:

SOLO

Listen to any input by itself, or preview an entire group. Pushing ANY solo button automatically puts that channel into the headphones, no matter what signal was there before. Electronic FET switching makes this possible.

MODULARITY

Tangent consoles are totally modular for servicing ease. Take a spare module on the road for no down time.

100mm SLIDERS

Tangent uses 100mm long-throw faders for extra control and visual feedback. Compare these to the competition's usual 45mm or 60mm length.

THREE SENDS

Effects, Reverb, and Monitor sends on each channel act as three independent mixers-within-a-mixer. Compare Tangent's three send busses to other mixers having typically just one or two.

CHANNEL PATCHING

Each input has a pair of access jacks for patching external effects into a single channel or sending a direct feed to a multi-channel tape machine.

LOTS OF EXTRAS

Tangent also has totally balanced inputs and outputs, buss access jacks for slave mixer expandability, and an optional reverb that is one of the smoothest sounding spring units available.

Compare these features to those on consoles costing twice as much and you'll see what a value Tangent is.

As for comparing the quality, well, you just can't get better than the original.

Write or call for your nearest dealer.

tangent
musical engineering
2810 south 24th street
phoenix, arizona 85034
602-267-0653

CIRCLE 98 ON READER SERVICE CARD