SERVING THE CREATIVE AUDIO AND MUSIC ELECTRONICS INDUSTRY

SOUND ARTS

MERCHANDISING JOURNAL

VOL. 2 NO. 5
JUNE 1979

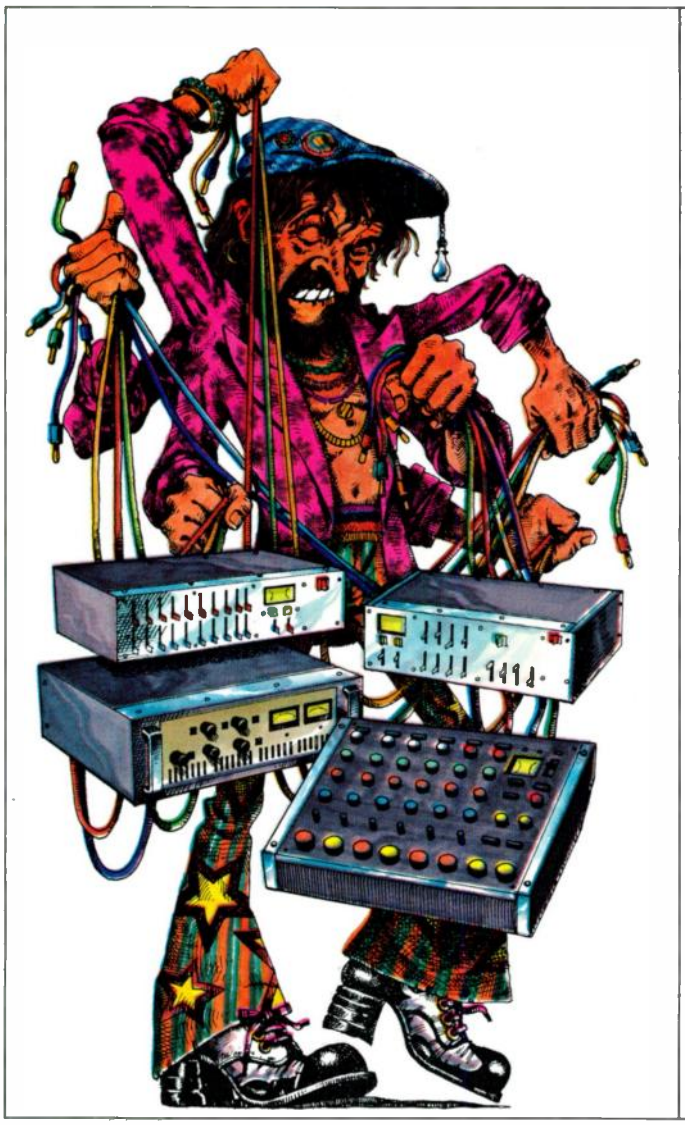
The Worst Sound Problem I Ever Had- And How I Coped

Do's and Don'ts of Equalization

Building Sales With In-Store Displays

Computer Technology and Music. 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 now.

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FOR THE SMALL STUDIO OWNER WITH BIG IDEAS.

If you're a small studio owner, you may have a problem: your ideas are far beyond your present equipment.

Maybe you're an engineer, dreaming of an automated 24-track studio. Maybe you're a producer, searching for "the next big thing." Maybe you're an artist, trying to land a record contract. What you need is something that will get you from where you are to where you'd like to be.

Sound impossible? Not to us. At dbx, we're committed to make professional recording technology available to anyone with the determination to make use of it. We make a line of rack-mountable signal-processing devices designed and priced especially for the small studio.

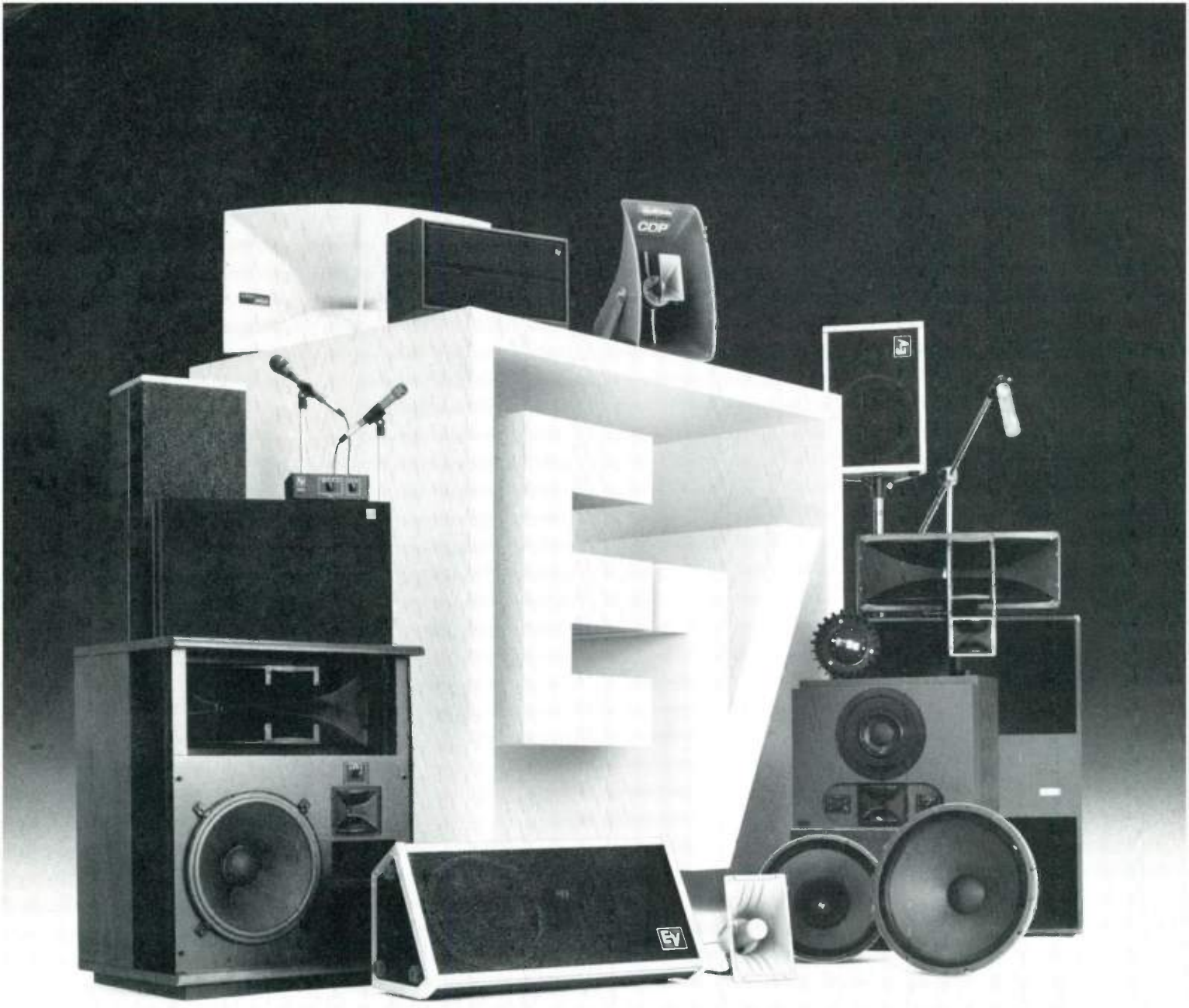
Our tape noise reduction systems, the 155 (4-channel, switchable), the RM-155 (8-channel, switchable) and the 158 (8-channel, simultaneous), all offer the same 30 dB noise reduction and 10 dB headroom improvement as our state-of-the-art units and are fully compatible with them. They enable you to make master quality tapes, instead of demos, on your present equipment.

Our 161 and 163 compressor/limiters feature true RMS signal detection, which closely resembles the response of the human ear, and feed forward gain reduction, which allow for infinite compression capability. The 163 employs "Over Easy" compression, the most natural-sounding you've ever heard, and its "one knob" operation is the simplest around.

We can't guarantee our products will make you a star. But if you've got the talent, they'll take you as far as you want to go. dbx, Incorporated, 71 Chapel Street, Newton, MA 02195 617-964-3210.

dbx
UNLOCK YOUR EARS

CIRCLE 99 ON READER SERVICE CARD



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

As a music dealer, you should be interested in the Electro-Voice story, because when it comes to music microphones, loudspeakers and music speaker enclosures, Electro-Voice is the leader.

The music your customers listen to at home was probably recorded using Electro-Voice professional microphones and mixed using E-V Sentry® studio monitors. The same technology that goes into these products also goes into all E-V music systems. Is it any wonder that Electro-Voice is becoming *the* asked for name in music.

If music is your business, it's good to know that the famous EVM® loudspeakers are not only standard in many manufacturers' "premier" 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.

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, stop by our exhibit at NAMM and talk to Chuck Gring, Electro-Voice Music Products Sales Manager, or contact us at Electro-Voice, 600 Cecil Street, Buchanan, Michigan 49107. Phone 616/695-6831.

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See us in Room 200, Second Level, Georgia World Congress Center, National Music & Sound Expo, June 9-12

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SERVING THE CREATIVE AUDIO AND MUSIC ELECTRONICS INDUSTRY

VOL. 2 NO. 5

SOUND ARTS

MERCHANDISING JOURNAL

JUNE 1979

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Cover photo by Doug Hanewinckel
Courtesy My Father's Place, Roslyn, NY

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SERVING THE CREATIVE AUDIO AND MUSIC ELECTRONICS INDUSTRY
SOUND ARTS
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Important Open Letter To Manufacturers

Re: Creative Audio And Music Electronics Market Study

As we all know, the sad lack of hard statistical marketing data has been much lamented and discussed within our industry. In an effort to begin to rectify the situation - to define the growth and promise of this burgeoning industry - SOUND ARTS has elected to undertake a Creative Audio and Music Electronics Market Study. Our plan is to solicit, tabulate and disseminate information from manufacturers that will indicate overall industry sales, categorical dollar representations and individual product unit sales.

One of the difficulties in providing this much-needed service is the understandable unwillingness of many manufacturers to make public their sales figures - either in dollars or units.

Most of you know all of us at SOUND ARTS and, we would hope, have a degree of confidence in our professionalism and integrity. We can't provide this service without your trust and honesty. We understand the various corporate philosophies regarding the privacy of sales figures and corporate volume. However, there are several ways we believe we can prepare this

study for dissemination and still preserve anonymity. Remember, we are not interested in providing information on any individual company's sales. Rather, we are interested in industry-wide categories of dollar and unit volume.

We are designing a questionnaire that includes all product categories within our domain, and plan to mail it to all manufacturers. Our plan is for this questionnaire to be answered on a standard typewriter and returned in a postage-paid envelope. No company name will be required. To further ensure anonymity, completed questionnaires will be received at a post office box and opened prior to our receiving them. All envelopes will be destroyed, and SOUND ARTS will thus receive only anonymously typed statistical information on returned questionnaires.

The success of this project depends mostly on two elements: honesty and trust. This survey may appear to be an impossible task, and - quite frankly - it might well be so. To ascertain the degree of cooperation available, we are mailing, during the first week in July, a copy of this letter with a yea-nay ballot whereby you, the manufacturer, can advise us of your feelings on whether we should proceed with the statistical questionnaire.

In closing, we want to reiterate our belief that information and education are the only bases upon which our industry can continue to grow at the healthy and progressive rate it has enjoyed. With your cooperation we will all have more knowledge of where we stand and where we are going.

We would like to tabulate and publish the survey by early fall. Your comments will be appreciated.

Very truly yours,

Vincent P. Testa
Publisher

Judith Morrison Lipton
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14 VANDERVENTER AVENUE·PORT WASHINGTON, N.Y. 11050

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A LETTER FROM THE EDITOR

A wing and a prayer and the seat of your pants make flying easy but eminently iffy. You can't always fix it in the mix and the unexpected always happens. Prayers turn to mist without a substantial underpinning of knowledge and preparation.

I asked a sound man I know how he had coped with his worst sound problem ever. His answer was direct, if tongue-in-cheek. "Shot the producer." The tactic may beat prayer, but it's of little practical help unless you're into shooting producers and/or piano players. David Hadler and Michael Berlin had a less violent and more positive approach, which they write about in this issue. It boils down to the necessity for broad-based knowledge with a heavy overlay of ingenuity controlled by the Boy Scout maxim of Be Prepared. Hardly a sexy mix, but an apt combination when you can't fix it and the show must go on. That's show biz—when winging it is a less than satisfactory endeavor unless that wing has something more substantial than a prayer on it. Ingenuity doesn't exist in a vacuum and it can only be effective when attention has been pre-paid to the just-in-case.

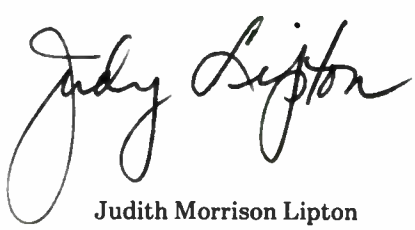
Ingenuity of course also plays a part in daily retailing. Allen Hester writes this month on setting up displays within the retail establishment. It's another case of success being dependent upon planning, knowledge—and a lot of ingenuity.

Ingenuity is on my mind because of course it is again the semi-annual season of shows in our ingenious industry when the manufacturers exhibit the product perceived as the best to promote. At this writing at least ten manufacturers have already taken their show on the road for press previews. Some of them have been interesting, all of them have been at least pleasant. Attendant irrelevancies included our staff being locked into the Giants Stadium grounds on the way home from Secaucus (don't ask for details, they're too absurd to be believed); and coffee at Regine's (to the tune of opera music rather than Gloria Gaynor).

The Consumer Electronics Show itself held its semi-annual press reception in New York, so heavily attended by the press that one manufacturer was heard to say to another, "I guess it's all press, and they're talking to one another." True in part. But at the product presentation press conferences, the product and marketing plans are the main focuses of attention. The plethora of presentations makes it difficult to cover them all, and in truth some manufacturers aren't ready for presentations until close to the show. Given the tyrannical demands of printing deadlines, a full and complete rundown in a monthly magazine requires an after-the-market article, which will be running in the August issue.

Meanwhile, it's the season of the show and to all attendees at AES, CES and NAMM, may all your quests be fulfilled.

Regards,

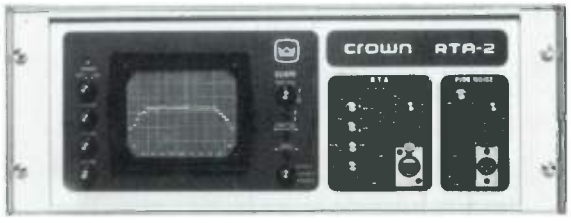


Judith Morrison Lipton

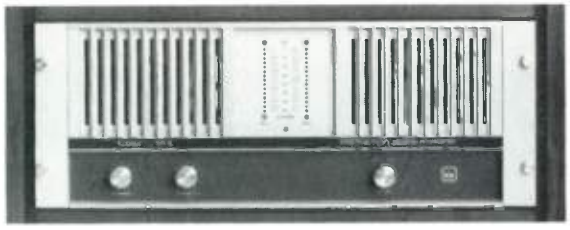
Did you notice?

Since June of 1978, Crown International, Elkhart, Indiana, U.S.A., has introduced the following audio products to the high end component market:

DL 2 A new concept in control centers that has the reviewers reaching for superlatives.



RTA-2 The versatile frequency-response analyzer for contractor's tool boxes, or dealer repair benches.



SA 2 The amp with the built-in brain for ultra-reliable power along with Crown sonic accuracy.

PSA-2 Professional power. Self-analyzing circuits protect against any overload.

STRAIGHT LINE ONE
The pre-amplifier for home use. Simple, accurate sonic control.

POWER LINE ONE
Companion amp for Straight Line One. Clearest, cleanest sound.

FM1 TUNER The all-American tuner. Includes digital control that's fun to use.

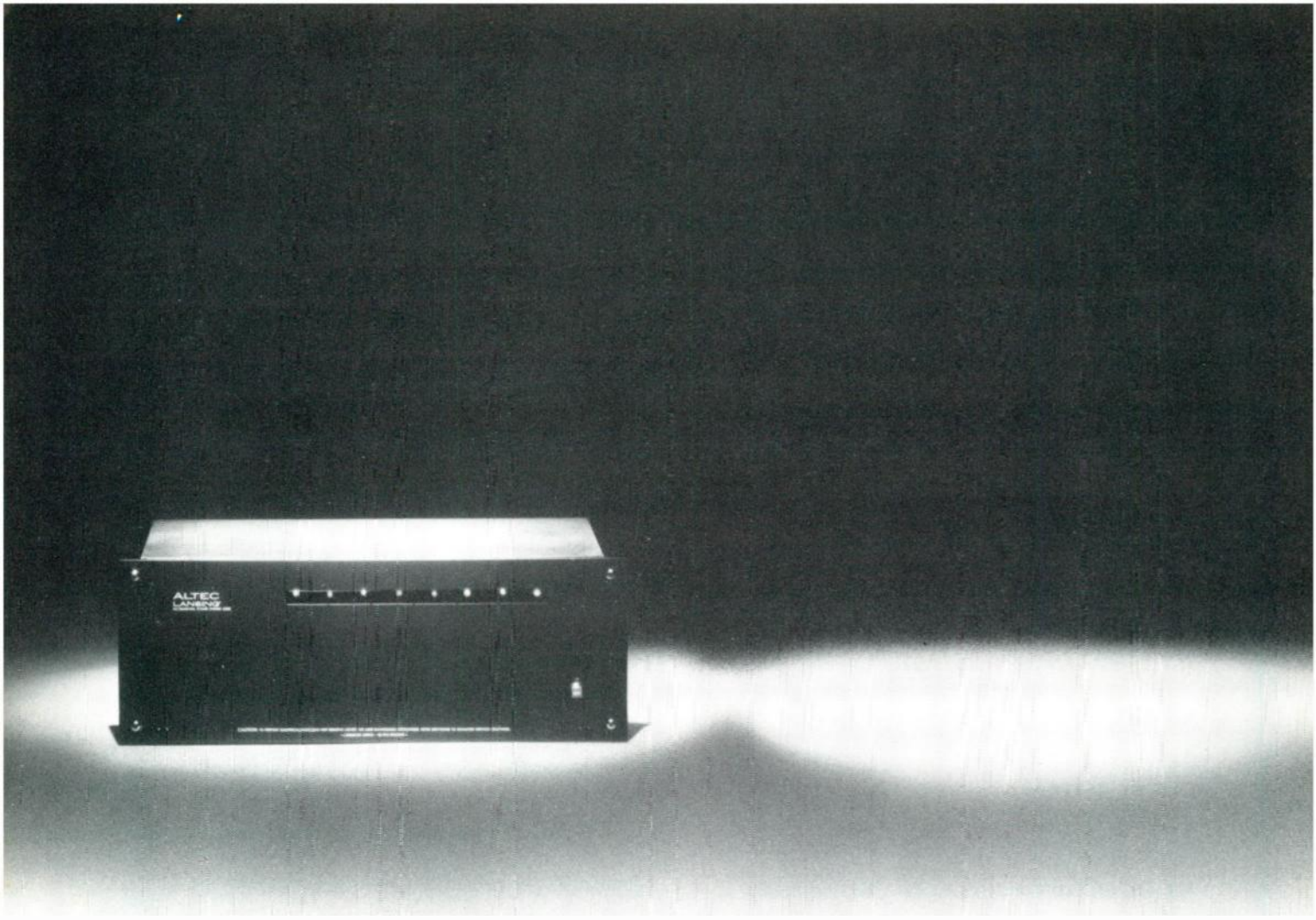
D75 The little-package amp that provides surprising power, super sonic excellence and Crown reliability.

Innovation. High technology. American. That's Crown.

crown

1718 W. Mishawaka Road, Elkhart, Indiana 46514

CIRCLE 67 ON READER SERVICE CARD

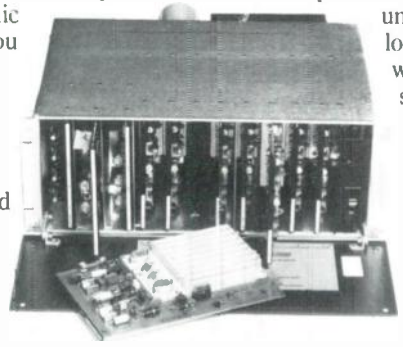


Introducing Altec Lansing's Incremental Power System. And Its Closest Competitor.

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

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

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



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

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

ALTEC LANSING
Altec Corporation

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

CIRCLE 76 ON READER SERVICE CARD

TEAR HERE

TERMS:

A CONTINUING INDUSTRY GLOSSARY

RECORDING

By Larry Blakely

Passive Filter: A filter that is made of passive electronic components (i.e., coils, condensers, and resistors). A passive filter will cause an insertion loss (loss of signal level, usually 3 to 6 dB) when it is inserted into the signal path. Passive filters were more commonly used prior to the 1960's.

Master Fader: Used to control the level (by attenuation) of an output bus or a group of output buses. A stereo master fader would have two fader elements (attenuators) controlled by a single fader knob. A quad master fader would have four fader elements (attenuators) controlled by a single fader knob, etc. The stereo master fader is often used for making fades at the end of tunes.

Sub Master: A fader that is wired in prior to the master fader. Submasters act as trim controls. If for some reason the signal level on a console were one or two dB too much on one channel, it could be corrected by adjusting the level of the submaster for that channel. A stereo output bus would have two submasters and a stereo master fader. A quad output bus would have four submasters and a quad (four channel) master fader, etc.

VU Meter: A meter calibrated in decibels to indicate level in dB. The typical VU meter displays a range of 23 dB (−20 to 0 and 0 to +3 dB). VU meters are commonly used on all mixers, consoles and tape recorders. The term VU means volume units. There are two common types of VU meters in common use: average reading and peak reading.

VU Meter (average reading): Indicates the average level of an audio signal. It is important to note that an average reading meter will not respond fast enough to indicate the peak level of the signal. Peak signal levels can be typically from 5 to 20 dB above what an average reading VU meter will show.

VU Meter (peak reading): Indicates

ELECTRONIC MUSICAL INSTRUMENTS & ACCESSORIES

By Wayne Howe

Class AB Amp: A combination Class A and Class B power amplifier where each section is turned on for slightly more than half of each cycle. When no input signal is applied, both positive and negative sections draw equal and opposite amounts of current. In this case, the output voltage of the amp is effectively zero. This is the most widely used low distortion power amp on the market today.

Class D Amp: A rarely used digital switching audio amplifier.

Class G Amp: A relatively efficient power amplifier which operates in a conventional mode for low power applications. When the input signal exceeds a certain level, the amplifier switches to a different set of output transistors to deliver the necessary high power peak. When the peak has subsided, the amplifier switches back to its lower power transistors.

Class H Amp: A relatively efficient power amplifier that uses the same output transistors but increases the supply voltage to handle high power peaks. When the input signal decreases to a normal level, the power supply resumes its normal supply voltage level.

Fundamental: In music, the fundamental is either the root of the chord or the basic pitch of the note being played. In electroacoustics, the fundamental is the lowest frequency component of the tone being used for testing. It is the oscillator frequency that is used as an imitation of a complex audio signal to observe the parameters of the equipment under test.

Harmonic: A vibration that occurs in mathematically-related integer multiples of the fundamental frequency or test tone. In music, these combinations of harmonics create what is termed the "timbre" or "texture" of different musical instruments. Amplifiers and other linear devices should pass the musical signal without adding harmonics. The purpose of controlling

SOUND REINFORCEMENT

By Glen E. Meyer

Older condenser microphones, especially those which incorporate vacuum tubes, tend to be rather fragile and careful handling is a necessity when using them.

Now, solid state technology, coupled with the perfecting of self-polarized electret diaphragm material, plus careful attention to mechanical design, have recently made available condenser microphones with greatly improved reliability. Some models are rugged enough for the most demanding public address and music application, and also have the other performance characteristics ideal for high quality sound reinforcement and recording.

Multiple-Microphone Interference: A sound phenomenon, usually undesirable, which occurs when two or more microphones, each receiving the same input signal at slightly different times, are fed into the same mixer, creating drastic and shifting voids in certain frequency bands. (This does not occur between microphones in separate channels of a stereo or multi-channel sound system.)

Pop Sensitivity: The measure of a microphone's reaction to explosive vocal sounds, like "P," "T," and "F." A microphone with high pop sensitivity will create a very disturbing low-frequency "boom" in the sound system. Microphone types vary widely in their pop sensitivity.

Shock Mount: A mechanical device which reduces the audible effects of mechanical shock. External shock mounts, usually in accessory form, can reduce noises from stage floors and microphone stands, but only internal shock mounting of the microphone generating element can effectively reduce all types of mechanical shock noise at once. Shock mounts are purely mechanical in nature and not electrical.

Microphone Impedance: Defined in an earlier section, it is a common

TERMS (CONTINUED)

A CONTINUING INDUSTRY GLOSSARY

RECORDING

the peak level of an audio signal. This meter responds only to the peak signal levels and not the average level of a signal. A peak reading VU meter is sometimes referred to as a PPM (peak program meter).

PPM (peak program meter): A peak reading type of VU meter.

Echo Device: A unit that will cause discrete delays and recycle them for an effect as though one were yelling in a mountain canyon.

Reverberation Device: Produces an effect like that of a large hall or reverberant area. The same type of effect can be obtained from a hard tile bathroom. These devices make a signal sound very alive and have a smooth decay time as the sound dies away.

Reverberation Time: The time it takes a signal to die away after the original signal stops. Technically, it is the time for the signal to decrease 60 dB after the original signal stops.

Echo Chamber: Often the term used to refer to a reverberation chamber. It also can be a room with hard walls, ceiling and floor with a speaker at one end and microphones at the opposite end, which would then be a reverberation chamber.

Reverberation Chamber: See Echo Chamber.

Digital Reverb: A device that simulates a reverberation chamber or reverberant room by means of a digital electronic process.

Digital Delay: A device that will delay an audio signal up to a given amount of time, usually in milliseconds. The amount of time delay is operator adjustable for most units on the market. This delay is done by a digital electronic process.

Analog Delay: Similar to the digital delay, but this effect is done with analog electronic techniques instead of digital electronic techniques.

Reverberation Plate: A reverberation device to simulate a reverberant room by electronically exciting or vibrating a large metal plate.

ELECTRONIC MUSICAL INSTRUMENTS & ACCESSORIES

equipment and distortion devices is to alter or add to the harmonics of the musical signal.

Overtone: A musical term for a harmonic, described above.

Harmonic Theory: The theory of musical acoustics that explains the presence and behavior of integer multiples of the fundamental frequency. Electroacoustic tradition typically labels the fundamental frequency as f_0 and the *first* harmonic f_1 as $2 \times f_0$ which is an octave above f_0. This reference to an octave above the fundamental as being the *first* harmonic is technically correct. However, common usage labels the fundamental frequency as f_1 and calls the octave above that f_2, the second *harmonic*. This popular usage would indicate that the first harmonic *is* the fundamental. Although this is technically incorrect, once the common usage is understood, it allows less confusion as to which harmonic is the correlative multiple integer of the fundamental frequency.

In other words, the traditionally correct method would state: The third harmonic, f_3, is four times the fundamental f_0 (2 octaves above the fundamental). In common usage, the same phrase would be stated: The fourth harmonic, f_4, is four times the fundamental f_1 (2 octaves above the fundamental). The common usage is easiest to understand and is used in the following definitions.

Overtone Series: A musical term for the harmonics of the fundamental. If we assume that the fundamental frequency f_1 corresponds to the lowest note C_1; then the second harmonic $f_2=2f_1$ is the note C_2 at one octave above the fundamental. The third harmonic $f_3=3f_1$ is the note G_2 at one octave and a fifth above the fundamental. The fourth harmonic is the note C_3 at two octaves above the fundamental. The fifth harmonic is the note E_3 at two octaves and a major third above the fundamental. We will continue discussing the overtone series next month.

SOUND REINFORCEMENT

characteristic of an electrical device expressed in ohms (Ω). Microphones are generally classified as either being high Z (10,000 ohms and up) or low Z (50 ohms to 250 ohms). High impedance microphones generally have higher output than low impedance types (about 20 dB). The biggest advantage a lo Z microphone has over a hi Z is that it permits the use of longer cables without experiencing high frequency rolloff. Only lo Z microphones should be used if microphone cables will be longer than 15 feet.

Input Overload Distortion: This is distortion caused by too great an input signal to an amplifier or preamplifier. This type of distortion most frequently occurs when microphones are used close to the sound source. This overload is best controlled with an attenuator.

Ribbon or ("Velocity") Generating Element: Ribbon microphones are similar to dynamics, except that a very thin metal-foil ribbon serves as both diaphragm and voice coil. In order to obtain adequate frequency response and output level, the thin ribbon must be exceedingly light. Older ribbon microphones could easily be destroyed by mechanical shock or a sudden blast of air which would stretch and destroy the fragile ribbon. However, the best current designs have been significantly improved for very satisfactory durability.

Microphone Pickup Patterns: A microphone's pickup pattern is three dimensional in character and shows how the microphone responds to sound from different directions. Omnidirectional microphones pick up sound pretty much the same from all directions. Unidirectional mics reject or reduce sound from the sides or rear.

Omnidirectional Pickup Pattern: The pickup pattern of an omnidirectional microphone may be represented as an inflated balloon with the microphone at the center. The polar pattern will be illustrated next month.

Today's hottest recording group.

Latest sales figures show that Maxell is the fastest-growing brand of recording tape in the country today.

It's not surprising.

Maxell cassettes are used by more people who own the finest tape equipment than any other brand.

Our open-reel and eight-track tapes have been accumulating some great performance records of their own.

Call your Maxell representative for an audition.

You'll find he, like our tape, is really worth listening to.

maxell

TROUBLESHOOTERS' BULLETIN

NON-ADHESIVE TRANSDUCER MOUNTING

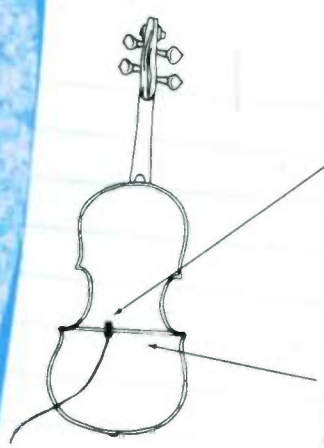
(1)

When we were faced with the problem of mounting transducers on a relatively large group of instruments that were both rare and beloved, we advised a method of mounting the transducer that used no adhesive whatsoever, and did not affect the tonal qualities of either the instrument or the Frap.

For the Emerson, Lake and Palmer Touring Orchestra, for instance, all the strings used the same transducer (Frap 3-D). In every case, when the pickup was placed directly over the sound post (the best location, we found), the instruments still sounded like the instruments - only louder. The transducer can be mounted by a compression method in order to preserve the finish of the instrument.

(2)

The first step is to scrape off the excess wax on the transducer and rub with sandpaper. When all the wax is removed, use a silicone rubber adhesive such as Dow Corning 3145 or G.E. Auto Seal in place of the originally waxed area. Place



(3)

Frap transducer. Sensitive part placed directly over the sound post.

Rubber band. In the case where the corners are worn or in the case of basses without violin corners, use an elastic band that will go around the instrument. It can fasten or unfasten with Velcro.

FIGURE 1

a piece of cellophane paper over the silicone rubber and press the transducer on the back of the instrument with the active part of the pickup being directly over the sound post. When dry, peel off the cellophane and a mold contoured to the instrument is made.

To mount the transducer, attach with rubberbands from corners, as with the shoulder rests on a violin. (See figure 1.)

ARNIE LAZARUS
PRESIDENT
FRAP

A LOUDSPEAKER CONTINUITY, POLARITY AND TRANSIENT RESPONSE TESTER

The only active element involved in this simple but useful circuit is a 1-1/2 volt flashlight battery. The basic concept is a battery and push-button momentary switch mounted in a mini-box. The switch and battery are wired in series and connected to a phone jack output. The jack can be used with a phone plug speaker cable for testing enclosed loudspeaker systems, or with phone plug to bare leads or alligator clips for testing raw transducers.

The most basic evaluation that can be made with the tester is a simple continuity check. If a speaker's status is unknown, applying the pulse from the tester can indicate whether or not the speaker is operating without setting up an amplifier and signal source. The excursion produced by the tester's pulse may also be sufficient to give an audible indication of a

rubbing or distorted voice coil or other cone motion problems.

Another important function of the tester is in determining loudspeaker polarity. The accepted standard is for positive voltage applied to the + or red terminal of a transducer, or the tip of a phone jack connection, to produce forward diaphragm motion. Not all manufacturers follow this rule, however, and custom enclosures are always in question. When more than one

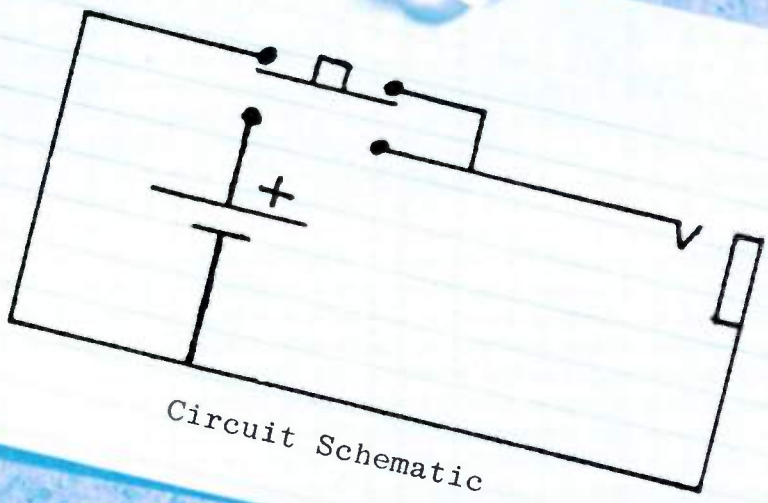
speaker is reproducing a particular instrument or frequency range, it is most important that all of the transducers are operating in phase, with the same polarity; and the tester is a simple way to check this.

The tester can also be used to make subjective evaluations of the transient response of loudspeaker systems. When the switch is closed, the signal output quickly goes from zero to the battery voltage level.

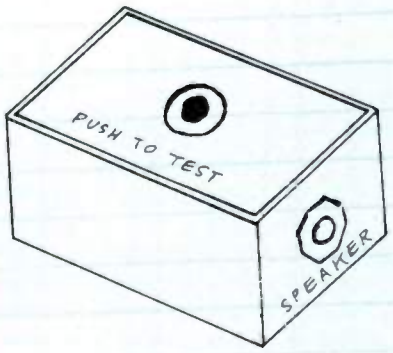
This instantaneous rise is an excellent approximation of a step function, the leading edge of a square wave. Similarly, when the switch is released, a pulse of the opposite polarity is produced, and the short circuit connection in the circuit simulates the high damping factor of most amplifiers.

This type of impulsive signal

is very revealing of the transient response characteristics of loudspeakers. Most high efficiency musical instrument and sound reinforcement loudspeakers will be highly damped, and produce a sharp snap or click. If a boomy, ringing sound is heard, with a long decay or hangover, it may mean that the system is tuned improperly if it is a vented system, or in need of additional damping in



Circuit Schematic



Finished Box

PARTS LIST: SPDT or DPDT momentary switch; 1/4" phone jack; 1-1/2 volt flashlight battery; battery holder; plastic or metal mini-box; miscellaneous wire, hardware, solder, etc.

the form of box lining, or other acoustical treatment to the enclosure walls, port, or the driver itself.

Don't be put off by the simplicity of this little box. Put one together and you'll find yourself grabbing for it often to make many basic checks on loudspeakers and systems.

MARK GANDER
TRANSDUCER ENGINEER
JBL

What is the variable speed control on the recorder and how is it used?

The variable speed capability can really be a key feature for the recordist, especially in the multitrack field. Here are six of the most common applications:

You can change the tone and timbre of an instrument or voice by recording several semitones down in tape speed, perhaps as much as 10 percent. On playback you'll find that the decay time of all sounds will be altered (shortened) relative to the sound had it been recorded at normal tape speed. This effect will also give the illusion of tighter timing, or "tightening the act." With the slower tempo, the opportunity for more precise timing allows for making an otherwise unacceptable take sound flawless. And if the speed variation is not so severe that it changes the basic instrument sound on playback, a mediocre lead guitar player can be transformed into a fleeting-fingered fretless wonder. This is accomplished by playing leads and licks at normal paces within the confines of the slower tempo.

Using that same method, but at an overdub tape speed of half or slower, you can make some instruments sound like others when played back at normal speed. A guitar can sound like a mandolin and a piano like a harpsichord.

Great pains may have been taken to tune all the instruments exactly with the pitchpipe or strobe-tuner and all the basic tracks may have been recorded. But then, when it's time to do the piano it's found to be a half-tone sharp. Well, instead of taking the piano to Instant-Tune, you simply use the speed control to match pitches. Of course, it doesn't have to be a piano, but that's usually the culprit in such a situation.

The vocalist is singing harmonies to a lead vocal or as a background chorus. There's another part floating around in his head, but it's too high for his voice. The solution is to slow the tape (lowering the key of the tune) until the harmony is within his range and to then record it. And again, the slowing of tempo affords more precise timing which is so critical in harmony. This might also be tried for a lead vocal, but it should only be a slight speed variation because, as mentioned above, the coloration of the voice will be different and you probably want to maintain a consistent lead vocal characteristic throughout the album.

When a piece is being produced to fit precisely in a time slot, the speed variance can give or take a few seconds for the correct fit in real time. Real time means the actual playback time in minutes and seconds at the speed of choice. For example, if you use 20 hours of recording time at various speeds including overdubs to produce a 4-minute piece, then the take time is 20 hours, but the real time is only 4 minutes.

At any stage in the recording process it might be decided that the piece "just doesn't work," though it's hard to pinpoint particular flaws. Before scrapping hours of effort, try a slight change in tempo by varying the playback speed. Many a producer has saved his efforts through this method.

These techniques often make the difference between the ordinary and the hit. They may be used in the home or in the professional studio.

Roy Brett Kamin
TEAC Corp. of America
Montebello, Ca.

What is a humbucking pickup? How does it differ from a single-coil pickup? Can they be used together?

In essence, without being purely technical, a humbucking pickup generally consists of a magnetic arrangement wherein two coils are placed in close proximity to each other in a single magnetic structure and the two coils act on each other to effectively cancel out 60-cycle hum—60-cycle hum being the loudest prevalent hum you have to deal with in high-impedance situations. The advantage, therefore, over most standard high-impedance magnetic coils is very simply that in terms of background noise, humbucking pickups are very quiet in relation to the noise a high impedance single-coil pickup produces. You can reduce the noise in a single-coil pickup by completely shielding the insides of the electric guitar, but the effect is still not quite as great as utilizing a humbucking pickup.

They can be used together. There's no reason why any pair of pickups with relatively similar outputs cannot be used on the same instrument. The differences in physical construction are not as important as the differences in output when using pickups together. However, if there is too great a difference in outputs, you may not be able to balance them within the same instrument adequately. The only other problem is a psychoacoustical effect. Namely, that when you have an instrument that consists solely of single-coil pickups, the noise is not as apparent as when you can switch between a quiet pickup and a noisy one. And then it appears that the single-coil pickup is the noisier one because you have something on which to base a comparison. This is the only reason they should not be used together.

However, the nature of the humbucking pickup also involves some cancellation of various frequencies which sometimes gives the effect that a single-coil pickup has a greater range of sound than a humbucking pickup. Most people do not consider this a defect in the humbucking pickup, however, but in point of fact there is a difference sonically in this respect.

Steve Blucher
DiMarzio
Staten Island, N.Y.

How should a noise reduction unit be interfaced with an echo unit (in both P.A. and recording applications)?

The type of noise reduction unit employed will determine where it is placed in relation to an echo unit. Most

echo units have a limited dynamic range (the difference in level between clipping and residual noise). To get the most out of this dynamic range, the first step would be to make sure that the signal coming into the echo unit is as high as possible without clipping the echo. A compressor and limiter may be used here to keep the signal within the dynamic range of the unit. A Noise Gate, dynamic filter, or downward expander should be connected to the output of the echo unit and adjusted to remove the residual output noise when no signal is present. Two-way noise reduction devices (Dolby, dbx, MXR) can be used with the encoder (compressor) preceding the echo unit and the decoder (expander) coming directly after the unit.

If the echo unit is used on stage, it is usually in series with other effects and between the instrument and amplifier. In this case, the dry signal is mixed with the delayed signal at the echo unit and any excessive expansion or gating of the echo signal will also affect the dry signal. When patched into a mixing board (either P.A. or recording) the echo signal is usually mixed back in at the board and gating the echo signal off will not affect the dry signal.

Keep in mind that any noise reduction technique affects the musical dynamics of the signal in the echo, thereby altering the unit's decay time or reverberation characteristics.

Richard Neatrour
Chief Engineer
MXR Innovations, Inc.
Rochester, NY

How does a bass-reflex enclosure increase loudspeaker efficiency?

Using a direct-radiator loudspeaker in a vented-box or bass-reflex enclosure can be said to increase efficiency in the sense that additional bass output is obtained. In a sealed-box system the springiness, or compliance, of the loudspeaker driver combines with the compliance of the air trapped in the box and reacts with the moving mass of the cone and voice coil. This creates a system resonance, below which the response rolls off. The energy from the rear of the driver is wasted, dissipated inside the enclosure. In a vented box, this back wave from the driver is used to excite a resonance between the compliance of the air in the box and the mass of air trapped in the port. When properly tuned, this second resonance in addition to that of the driver extends response to a lower frequency, though the response rolls off faster below cut-off.

Actually, the efficiency is an intrinsic quality of the driver, determined by the motor strength, effective piston area, and moving mass. This efficiency predicts the driver's output in the mid-band region independent of enclosure mounting. Modern loudspeaker theory has derived equations that describe loudspeaker systems in terms of their efficiency, box volume, and low frequency limit. These are all related by an efficiency constant depending on the type of system. A vented box has a higher efficiency constant than a sealed box, meaning that for a given box size and low frequency cut-off the vented system can be more efficient. A more practical case is that for a driver of known efficiency, a vented system will have a lower cut-off than a sealed system in the same box size.

Mark Gander
JBL
Northridge, CA

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CIRCLE 65 ON READER SERVICE CARD

Synthesizer filters

By Craig Anderton

So far we've covered a lot of ground in describing synthesizers, and since I don't want to spend any additional column space to re-cap what we've gone over, I suggest you take a quick glance over the last few issues of SOUND ARTS to refresh your memory concerning such concepts as voltage control, how synthesizer modules form a complete synthesis system, and so on. This month, we'll take a look at the *voltage controlled filter*, which is a very important modifier that allows you to change the timbre of the basic waveforms emanating from the oscillator.

WHAT IS A FILTER?

A filter is a modifier that selectively passes some frequencies, and rejects others. However, this process can happen in many different ways.

A *low pass* filter passes *low* frequencies (see, not all electronic music terms are mysterious!), but rejects higher frequencies. For example, if the sound coming out of an oscillator was overly bright and trebly, processing with a low pass filter would be ideal, since this type of filter removes treble. But there are additional considerations.

First of all, you need to specify exactly where the passing of low frequencies ends, and the rejecting (or *rolling off*) of high frequencies begins. This point is called the *cutoff frequency* of the filter; a cutoff frequency of 1 kHz would pass frequencies below 1 kHz, but above that point response would drop rapidly. For real-world instrument use, a 1 kHz cutoff is pretty low, and would take away much of the "sparkle" of a tone, turning it into a much "mellower" sound.

Another consideration is that no filter has a perfect cutoff. In other words, a 1 kHz cutoff does not mean that we wouldn't hear a 1.002 kHz signal; rather, the rolloff is relatively gentle. The characteristic of a filter to gently attenuate past the cutoff frequency is called the filter's *slope*. Some filters have a 12 dB per octave slope, which means in the case of a low pass filter that each higher octave will be 12 dB softer than the previous octave. Other filters typically have 24 dB per octave slopes, which means each higher octave will be 24 dB softer than the previous octave; therefore, a 24 dB per octave filter has what is called a *steeper* slope. This is the type of filter that was first popularized by the Moog series of synthesizers. Usually the slope of a synthesizer filter is a "given," but some designs allow for variable slope options.

There are other filter responses in addition to the lowpass kind, and these other types of filter responses are sometimes available on synthesizers. A *high pass* response is the opposite of a *low pass* response; it passes *high* frequencies and rejects *lower* frequencies. Everything is pretty much the inverse of the low pass type . . . the slope of the filter makes lower frequencies progressively softer, for example. Another type of filter is the *bandpass* filter. This type of filter passes only a narrow band of frequencies, rejecting all those that are higher or lower. However, this time the term "cutoff frequency" is no longer as appropriate, so the center frequency of a bandpass filter is called the *resonant* frequency. Signals both above and below the resonant frequencies are rejected.

A *notch* response is the inverse of the bandpass; it rejects a specific band of frequencies, but lets higher and lower ones go through pretty much unimpeded. This filter mode isn't found too often on synthesizers, but notch filters are used extensively in recording studio equalization.

THE VOLTAGE CONTROLLED FILTER (VCF)

In previous columns we've discussed other voltage controlled synthesizer modules, and noted how voltage control extends the operational flexibility of a module. Filters may also be voltage controlled to create many differing and dynamic types of effects. In the following discussion, we'll assume we're using a low pass filter; however, other types of filters can also be voltage controlled.

The characteristic of the filter that is affected by the control voltage is the filter's cutoff frequency. Therefore, with no voltage injected into the filter, the cutoff frequency is at a minimum. Injecting more voltage raises the cutoff frequency until you reach the highest frequency which the oscillator is capable of reaching.

To visualize the effects of a VCF, imagine what would happen if you fed a bright, brassy waveform that has a high harmonic content into the voltage controlled filter. At minimum voltage,

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An ARPography: 1969–1979

1969: Under the name Tonus, Inc., the ARP *2500* is introduced. It is the first synthesizer designed for live performance use, featuring slider-controlled matrix patching and rock-stable oscillators and filters.

1970: ARP introduces the world's first portable synthesizer, the *2600*, also known as "The Blue Meanie." The 2600 is discovered by Stevie Wonder and Pete Townshend, and becomes the most popular studio synthesizer of the 70's.

1972: The ARP *Odyssey* is introduced. Through many years and many design updates, it sets the standard for rugged, compact, versatile synthesizers, becoming "the most copied synthesizer in music" according to *Rolling Stone*.

1974: The ARP *String Ensemble* is introduced. The addition of "systems interface" jacks to the String Ensemble allows it to be used as a polyphonic synthesizer, more than two years before a self-contained polyphonic synthesizer is built.

1975: The *Axxe*, the first synthesizer under $1,000, is introduced. The Axxe opens up the amateur and educational market with its low price and high performance features.

1976: The polyphonic wave begins with ARP's introduction of the *Omni*, a symphonic electronic instrument that produces strings, polyphonic keyboard effects, and bass synthesizer *simultaneously*. The Omni sets the music world on fire, outselling every other synthesizer on the market.

1977: In one revolutionary stroke, ARP opens up the incredible sound possibilities of synthesizers to electric guitarists. The ARP *Avatar* boggles the mind and creates a brand new market.

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ARP enters the sound modification and effects market with the purchase of the Mu-tron product line.

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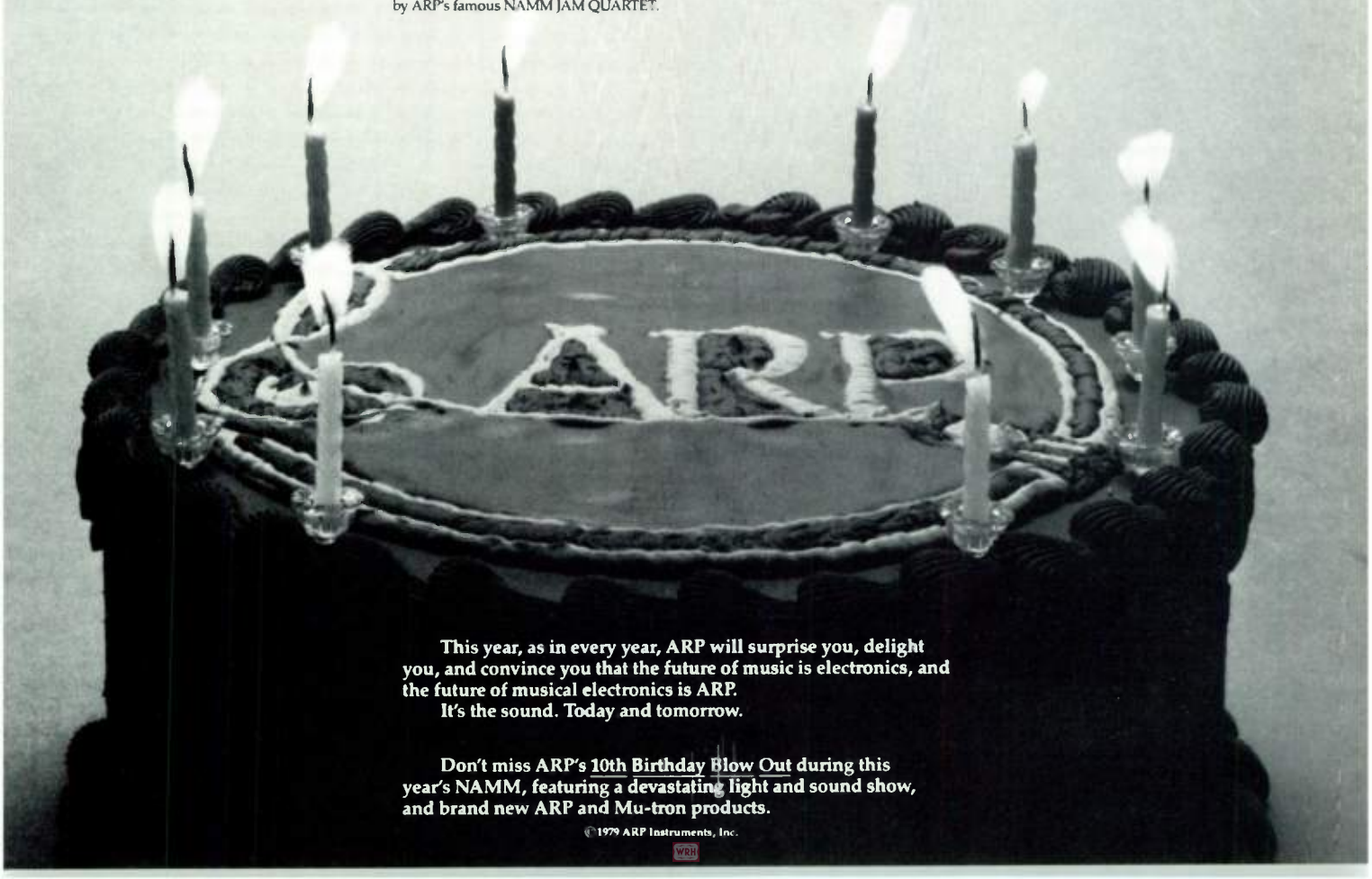

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the tone would be very muted; in fact, if the cutoff frequency of the filter is lower than the oscillator frequency by a significant amount, you may hear nothing at all. (By the way, this is one of those things to look for when someone says their synthesizer "isn't working"—it may be that the filter cutoff control is misadjusted.) As you increase the voltage, you hear more and more of the harmonics of the signal, and the overall tone quality becomes brighter. Eventually, when the cutoff frequency is at its maximum, the sound is essentially the same as if it hadn't been filtered at all.

One reason the filter is such an important part of a synthesis system is that real-world, acoustical instruments often exhibit changes in harmonic content that can be electrically simulated with a VCF. For example, wind instruments tend to have a lower harmonic content towards the beginning of a note, but the harmonic content increases as the player pumps more air into the instrument. This same phenomenon also occurs with bowed instruments. Therefore, if the synthesizer player wishes to simulate a wind or bowed instrument effect, using some type of filter is mandatory to give an accurate simulation.

FILTER "Q", OR RESONANCE

Another characteristic of the synthesizer filter is called resonance, which is set with a front panel control called either *resonance* or *Q*. This varies the intensity, or sharpness of sound, of the filter effect. Unfortunately, this effect is rather hard to describe in words, but here goes . . . . Think of a wa-wa pedal, with its distinctive, sharp sound; this is an example of a relatively high resonance type of filter. The type of bass and treble tone control filters on your hi-fi, on the other hand, produce a very gentle filtering action, and these are considered low Q type filters. Occasionally you'll find filters whose resonance is also voltage controlled.

INITIAL FREQUENCY CONTROL

You may not always want the filter to go from lowest possible cutoff to highest possible cutoff frequency every time you use it, so the *initial* frequency control injects a variable amount of voltage into the filter to establish an initial cutoff frequency. Then, other controllers can either add to, or subtract from, this initial voltage to change the cutoff as desired.

KEYBOARD "TRACKABILITY"

Raising the Q of the filter to a very high level gives an almost "whistling" type of effect; in fact, with many filters, setting the Q to a *very* high amount will actually turn the filter into an oscillator. As a result, we can think of the filter as being a tuned device, and requiring the same type of tuning stability and accuracy as an oscillator. For example, let's say the resonant frequency of a bandpass filter is 440 Hz, which corresponds to an "A" 440. Doubling the control voltage to the filter should therefore double the resonant frequency to 880 Hz, or exactly one octave above "A" 440, if the control voltage follows the convention we explained for voltage controlled oscillators (e.g., exponential response).

By feeding the filter with the same control voltage that feeds the voltage controlled oscillator, the filter will "track" the oscillator, meaning that it will follow the oscillator in a highly predictable way. This is very important; let's illustrate why with an example. Suppose you have a 500 Hz note going through a low pass filter, with the cutoff frequency set to 1 kHz. Now you change to a 1 kHz note—and if the filter *doesn't* track, its cutoff frequency is now the *same* as the note you're playing. Play a 2 kHz note, and the filter cutoff frequency is no *lower* than the note you're playing. But with a properly tracking filter, when you play a 500 Hz note, the filter cutoff will be 1 kHz (an octave above); when you play a 1 kHz note, the filter cutoff will be 2 kHz (still an octave above); when you play a 2 kHz note, the filter cutoff will be 4 kHz; and so on. This preserves the timbre of the various notes on the keyboard as you change notes during your playing.

One final note of interest: Even only a few years ago, voltage controlled filters were tremendously complex devices. Now, things have gotten to the point where there is a company that builds virtually complete, high quality voltage controlled filters in IC form . . . so times do change, and for electronic musicians, often these changes are for the better.

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SPICED JUST RIGHT

The Do's And Don'ts Of Equalization

By Chris Coffin

One of the oldest audio processors in the world is the equalizer. It has evolved from simple passive networks or tone controls to multiple band graphic and parametric equalizers. As technology has advanced, so has the user's responsibilities in maximizing the use of different equalizers. Not only does the user have to concern himself with network saturation; he also has to think about the proper gain staging of active equalizers. In this article we will discuss some of the do's and don'ts of equalization. We will also examine a few of the various applications for equalization in the studio and on the road. Since it is impossible to cover all equalizers in one article, we will deal with two popular types—the multiple band graphic and the parametric equalizer.

One of the most important things to maintain when equalizing is headroom. All electronic devices, whether active or passive, will take only a certain amount of input level before they reach their point of clipping or 3 percent distortion. Headroom is the difference between a unit's nominal operating level or point of unity gain and its maximum output. For example, if a unit has a nominal operating level of 0 dB and a maximum output of +18 dB, the unit has 18 dB of headroom. With passive equalizers, since there is no gain available through the device, the only thing you have to be careful with is that your program peaks do not exceed the clipping or saturation point of the device. With active devices, however, you have the ability to boost a frequency above the average program level. For example, assume an active equalizer with 18 dB of headroom at a nominal operating level of 0 dB. If you boost 4 dB at 3 kHz, you now have only 14 dB of headroom at that frequency. If you then dropped the input level of the equalizer 4 dB, you would regain your 18 dB of headroom at 3 kHz and increase your noise floor a little, but you would no longer have to worry about your 3 kHz clipping or causing audible distortion. The process of raising and lowering levels to maintain headroom with the

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Chris Coffin is a design engineer with Express Sound. He has done the monitor mixing for Bob Dylan's world tour and has worked on Francis Ford Coppola's new film, "Apocalypse Now."

least noise is called *gain-staging*.

Gain-staging is an important part of any audio system. In order to use an audio processor to its fullest, the audio system itself must have a good gain structure. A term often used in conjunction with gain-staging is unity gain. Unity gain is when the input of the device is equal to its outputs. Gain-staging begins at unity gain, since it is at this point that most devices are at optimum operating range.

Let's examine a simple audio system consisting of a mixer, equalizer and power amp. The nominal output of the mixer is +4 dB, yet the input of the equalizer is 0 dB. To maximize our system at this point, a resistive pad of −4 dB should be placed between the mixer and the equalizer. If the output of the console were simply turned down, the noise floor would increase. With a pad nothing is lost except the excess gain. Some equalizers have input pads in them. This is helpful, especially when using an equalizer with higher output consoles.

Now, after padding the console output, the equalizer is getting 0 dB in and sending 0 dB out (unity gain). However, the power amp wants to see +4 dB. A line driver may be used to raise the gain. Often, the equalizer will have its own output gain controls. As long as these controls are placed after the final summing amp of the equalizer, they can be used to raise the output level without any effect on the headroom of the equalizer. By dropping the level from the console to the equalizer, then raising the level after the equalizer for the power amp, the gain structure of the system is optimized. However complex the audio chain, the gain structure should be set before other adjustments are made.

It is advisable never to cut one frequency and boost the adjacent frequency unless they are more than one-half octave apart. This will help to avoid unwanted phase distortion.

Equalization has some negative properties that must be thought out and fully dealt with. For example, any time a signal passes through a filter and is altered, a certain amount of phase shift or phase distortion occurs. The audible signs of this are loss of definition and articulation. The more EQ you use, the more audible this becomes. Ringing can also occur when too much EQ is used in adjacent filters. The filters will find a common resonant mode and begin to interact or modulate each other, causing ringing or false tones to appear. The most common problem with equalization is that too much is used. Too much cut or boost causes problems with gain-staging. The noise increases, phasing problems become very audible, ringing may occur and any frequency balance improvement is negated by these other problems. The most important rule, when it really gets down to it, is to use your equalizer sparingly. It's a tool which will achieve an improved sound only if used properly. Have your customer maximize his microphone choice and placement, his power amp and crossover settings, speaker placement and room acoustics before he begins to equalize. If he goes as far as he can without EQ and then brings it in, he'll get much closer to his best sound.

MULTIPLE BAND EQUALIZERS AND PARAMETRICS

Two of the most common audio equalizers are the multiple-band equalizer and the parametric equalizer.

Before we examine their different features let's review some terms used when dealing with equalization.

Bandwidth. The arithmetic difference between the upper and lower cut-off frequencies of an equalizer or audio system. The range in which the equalizer boosts and cuts.

Center Frequency. The frequency at which the maximum amount of cut and boost are available.

"Q". The slope of the equalizer's curve. A high "Q" indicates that the boost applies to only a small number of frequencies adjacent to the center frequency. A broad "Q" would affect more adjacent frequencies.

Multiple-band graphic equalizers come in many different forms. The most common of these are 1/3 octave, 1/2 octave and 1 octave, with from ±8 dB of boost and cut to ±15 dB of boost and cut. There are some companies who manufacture cut-only active equalizers which are very useful, especially when equalizing sound reinforcement systems. The distance between center frequencies will be determined by the type of EQ you are working with. For example, a 1/3 octave equalizer will have center frequencies 1/3 octave apart beginning between 32 and 40 Hz and going to 16 kHz to 20 kHz. The bandwidth on this equalizer is set by the manufacturer. This is one reason why different manufacturers' equalizers sound different. The narrower the bandwidth, the more exact the equalizer. But sometimes a softer sound may be desired, so a wider bandwidth is needed.

Multiple-band graphic equalizers can be used in many ways. Because of the actual graphic display of frequencies and their relative levels, you know where and how much equalization you have at any one time. One-third octave equalizers are very versatile because of the large number, typically 27, of filter centers available. Both narrow boosts and cuts can be achieved and wide sloping curves can be created by using multiple center frequencies.

One octave equalizers are good for minor program equalization and instrument equalization. The reason for this is that their bandwidth is wider than that of a 1/3 octave or 1/2 octave equalizer, and therefore produce a softer sound. Because of its simplicity, the graphic equalizer may be used for many applications.

Parametrics require a little more thought in their use. Parametrics typically come in either two-, three-, or four-section versions. Each section has a variable frequency center, variable bandwidth (Q) control, plus the cut or boost control. Usually the center frequencies of each section will overlap to provide more versatility. Parametrics are usually tuned more to the ear than graphics. For example, if you wanted to bring up or brighten the high end of a piano, you would first begin by arbitrarily setting the center frequency and setting a narrow bandwidth. You would then add some boost and would sweep the center frequency until you found the point at which you wanted to make a change. Then, you could narrow or widen the bandwidth, add more boost or cut, until your desired sound was achieved. Parametrics and 1/3 octaves are sometimes used together to provide an extremely versatile system. We will further discuss both of these equalizers in the following section on applications.

EQUALIZER APPLICATIONS: SOUND REINFORCEMENT

Because of the rapid advancement of stage monitoring systems, requests for multiple-band graphics are growing in P.A. sales. Since feedback is the continuous regeneration of a resonant frequency through a speaker-to-microphone loop, equalization can be used to notch this frequency peak. Once it is attenuated, the overall pro-

gram level of the monitors can be raised. One-third octave cut only equalizers are great for this application, since they have a narrow bandwidth. After being on the road with several systems, I have also found parametrics to be excellent in feedback tuning. With a parametric, you can narrow the bandwidth down to 1/6 of an octave and achieve a narrower notch than the 1/3 octave EQ. Also, because the parametric has a variable frequency center, you can tune to the exact problem frequency instead of the approximate frequency you get with fixed equalizers.

Single octave graphic equalizers are great for rough tuning of house systems and disco systems. Because of their limited center points and broad bandwidths, single octave equalizers can be tuned by ear without the use of critical acoustic measuring systems. These equalizers are also good for keyboard and guitar systems.

ROOM EQUALIZATION

In both studio control rooms and large concert sound systems, room equalization is often used. Proper room equalization is done by using real-time analysis. By reading the real-time analyzer, critical adjustments can be made to the frequency response of the system in its acoustic environment. One-third octave equalizers are used here also, since multiple points are often needed to properly set up the system. Once this has been done, the equalizers should be covered and/or locked up in order to avoid any tampering with these settings.

STUDIO EQUALIZATION

In any studio control room, you will usually find at least four or five different types of equalization. Input or channel equalizers are found in the console. These may be parametric, fixed frequency and bandwidth, switched frequency and set bandwidth or any combination of types. These equalizers can be used for fine tuning of individual instruments, equalization of reverb or echo returns to the console and several other functions. Outboard or auxiliary equalizers consisting of one octave, one-third octave, parametric, notch filters, hi and lo pass filters and others also can be used for major or critical adjustments. Even though parametrics and multiple-band equalizers can be used as notch filters, a single frequency notch filter can be even more accurate for equalizing out things like undesirable 60 Hz hums and buzzes. So, the more sophisticated the outboard equipment is, the more precise the equalization choice can be. Hence, the less you have to use.

GETTING IT JUST RIGHT

All in all, equalizers are very valuable tools. They are like a spice, just the right amount and you've achieved perfection, too much and you've spoiled your creation. Work with your customer and qualify his needs. Maybe his problem isn't that he needs EQ, but that he needs a better microphone. Maybe his crossovers need to be adjusted, or maybe he needs some help with the acoustic environment he is working in. He should first get as close to the right sound without EQ. Work also with your customer on things like his gain-staging. Look at his system and find the best equalizer to achieve the best headroom and signal-to-noise combination. Explain to the customer that over-equalization is a problem. If these practices are followed, both you and your customer will be happy, and just watch those hit tapes and concerts come down the line.

The new M12. It eliminates the three deadly sins of live performance mixing.

If a mixer can't deliver a complete range of functions, high-quality electronics and rugged reliability, it can't bring a live concert to life.

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CIRCLE 68 ON READER SERVICE CARD

MXR on the use of multiple effects.

The use of special effects has become a major part of today's music. Contemporary musicians are able to reach new levels of self-expression by combining instrument and effect in the development of their technique. If multiple effects are used, not only is this expression enhanced, but a new world of totally unique sounds is provided.

Although many of today's knowledgeable musicians use multiple effects, this technique has not been exploited to its fullest. One reason for this is that the artist usually has a musical rather than technical background, and is more familiar with artistic expression than the concepts of signal processing. At MXR we feel that our knowledge as designers may be very useful to the artist. The information here has been gathered by our engineering and marketing staffs through customer feedback, research, and personal use.

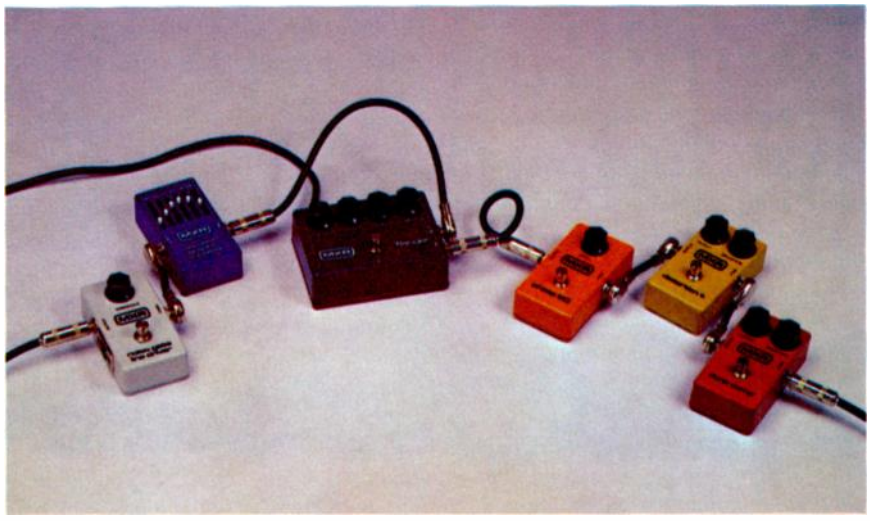
The purpose of this paper is to be a comprehensive yet understandable guide for the musician who is using or planning to use effects devices. We feel that MXR products provide well designed, legitimate effects that easily interface between your musical instrument and amplifier, whether employed singularly or in multiples. The following information, however, applies to any device of the same generic type. These devices are organized into groups according to their effect on the signal.

In a hypothetical situation, where all types of devices are used simultaneously, the following groups may be distinguished. Note that the order given is for the general case. Later we will discuss some alternatives and specific applications which you can try. The first group is composed of level boosting devices and includes preamps and distortion units. They are generally placed in the front of the effects chain because they are capable of providing a higher signal level for the successive devices to work with. The second group, the dynamics dependent devices, vary their effect in relation to the level dynamics of the incoming signal. This group includes envelope filters and envelope generators. Next are compressors and limiting devices which provide a more uniform signal level. Next are phase shifters and flangers which alter different frequencies in differing amounts. Delay lines, including tape delays, analog delays, digital delays, and doublers, are the next group. Noise control devices such as noise gates and noise filters would be next and would help to decrease the overall system noise. Equalizers belong in the last group and help to tailor the entire tonal character of the signal.

Let's examine a typical situation in more detail. It is beneficial to use a preamp or compression device such as our Dyna Comp first. The compression device sets up a more constant level for the next device, which might be a distortion device. Many additional harmonics are provided by the distortion device for later stages. Next a phase shifter would produce its characteristic frequency cancellation notches. If desired, a flanger would be used next to generate many more notches, especially at higher frequencies. Use a wah-wah pedal towards the end of the effects chain since it has the capabilities (in certain frequency ranges) of overdriving devices placed after it. If you are using a delay line, it should also appear near the end, regardless of what type it is (tape, analog, or digital), since it could potentially add noise to the signal. Use a volume pedal next. It can serve as a manual noise gate, a dynamics controller, or an envelope control. A Noise Gate/Line Driver, used at the end of the chain, can create a threshold between unwanted noise and the musical signal. It can also serve as a signal splitting device for remote recording or P.A. amplification. The last device is usually an equalizer. This contours the sound or may be used to compensate for the acoustical response problems at a club, auditorium, or studio.

The goal is to achieve a desired effect, impression, or sound, without losing control and without introducing unnecessary noise. An important idea to keep in mind is to make connections as short as practical, in the interest of a cleaner signal as well as neater physical organization. For instance, if you are using four different devices simultaneously, chances are it would be to your advantage to have them relatively close together. More noise can be introduced by the cables than by the circuitry within the devices. Most MXR effects devices have been designed so that when they are switched "out" the input and output jacks are effectively tied together to help maintain a clean signal when bypassed. The exceptions are our graphic equalizer pedals which are always "in" and our Noise Gate/Line Driver whose line driver function is always "in."

Buffering devices (preamps, compressors, etc.) establish a strong or uniform driving signal. In the beginning of a long chain (more than four devices), they can prevent the effects of "loading." The symptoms of loading are loss of highs and lows and the lowering of the "signal to noise ratio." The resulting sound is one of lost crispness. All MXR products have a relatively high input impedance so that more of them can be used with less loading effects.



Whether on stage or in a recording studio, a proper arrangement of signal levels must be set at all places along the signal path. In order for any electronic device to operate at its optimum it must be used within the specific level range for which it was designed. MXR devices have been designed to accept a wide variety of

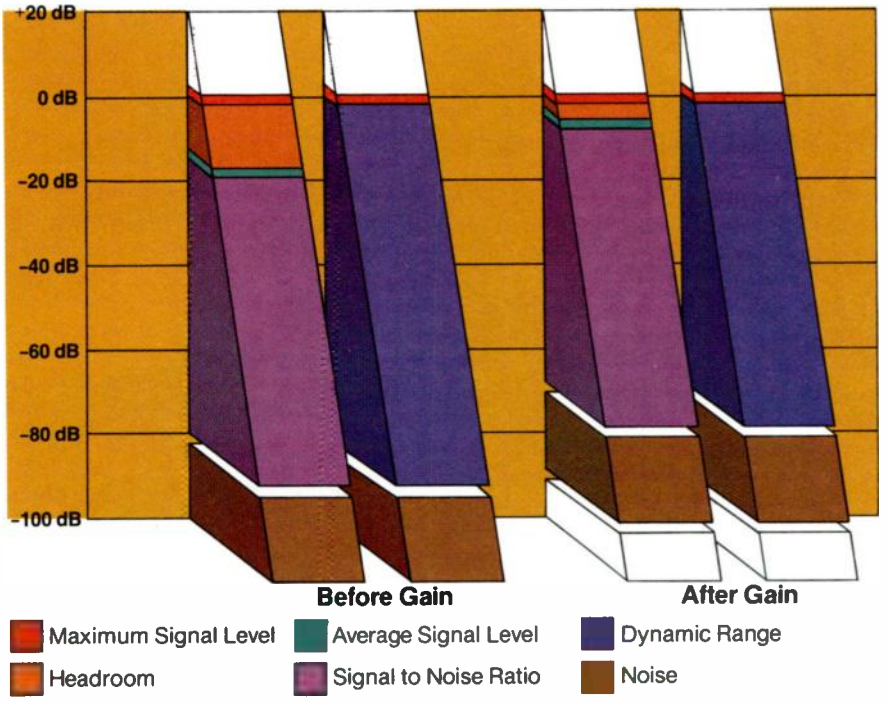
input levels. These levels must be considered, however, since they can vary greatly, depending on where the signal is coming from. Keep the signal, at all points, below the clipping level and above the noise floor. Maintaining adequate "headroom" refers to having enough room at the signal peaks to ensure that they are not clipped or distorted by the unit. Maintaining a high "signal to noise ratio" refers to having enough signal level to prevent unwanted noise from being heard.

Until now we have been primarily concerned with dynamics and level, but frequency response is also important. It is generally not advised to limit the frequency response in the beginning of the effects chain. An exception to this is the use of an equalizer at the beginning in order to purposely alter the apparent tonality of a particular instrument. An example of this is a guitarist who wants to make his guitar with humbucking pickups sound as if it had single coil pickups. Equalizers can also be used at the end of the chain to filter out unwanted high and low frequency noise.

Something to consider when using several devices is the level difference which can result from the device being switched in and out. Usually there will be a gain or output level control on the unit which enables adjustment of the modified signal's level in relation to the original or dry signal. A recommended method for achieving uniformity between signals is to play a note or chord with the device "out" then switch it "in" and play the same note again. Rotate the output control to the place where the signals are in their proper relationship according to your needs. Remember that the first devices have the potential of overloading all the rest, so be aware.

Another factor to consider is the harmonic character of the material and how it affects subsequent signal processing. To illustrate a point, consider a situation in which we can use either a compressor or a distortion device to feed a flanger or phase shifter. Theoretically, a compressor should not change the harmonic character of the signal, while the distortion unit adds many harmonics. The thickened frequency spectrum of the distortion unit offers more signal for the flanger or phase shifter to act on. The result is a more intense sound.

A compressor or limiter should generally not be used at the end of a signal chain. By definition, a



compressor brings up the lower level signal to a predetermined level. If that signal is the noise floor, then that is what will be amplified. What will result is a high concentration of circuit noise. Even though most of the higher quality units have been designed for low noise, it is still in your best interest to keep this in mind. By using a compressor or limiter before a volume pedal an interesting technique can be developed. This technique is particularly appropriate for steel guitar and electric piano. The compressor boosts the softer notes, giving the instrument greater sustain, and the volume pedal permits the artist to fade the music in and out smoothly for a fluid and gliding sound. The artist is also afforded the benefit of always knowing what level the signal is at before going into the volume pedal.

If you are using a dynamics dependent device you would not want to preceed it with a limiter. These devices are dependent upon the unaltered level dynamics of the incoming signal to control their effect and therefore will seem more subdued if preceeded by any device which alters the dynamics of the signal.

One interesting thing to try with multiple effects would be splitting the signal and processing it through different effects chains and monitoring amplifiers. Another thing to try, which is especially effective with keyboards, is to have a phase shifter sweeping with a narrow width and fast speed, running into a flanger, sweeping with a wide width and slow speed.

Once again the general rule is to run gain devices ahead of effects devices. However, always consider what the desired effect is to be. Combining knowledge and imagination can certainly help musicians (and engineers) to get more out of their art and the most from their electronics.

MXR is helping musicians to achieve their goals and enrich their music by designing professional products which are functional, durable, and superior in performance. See your MXR dealer.

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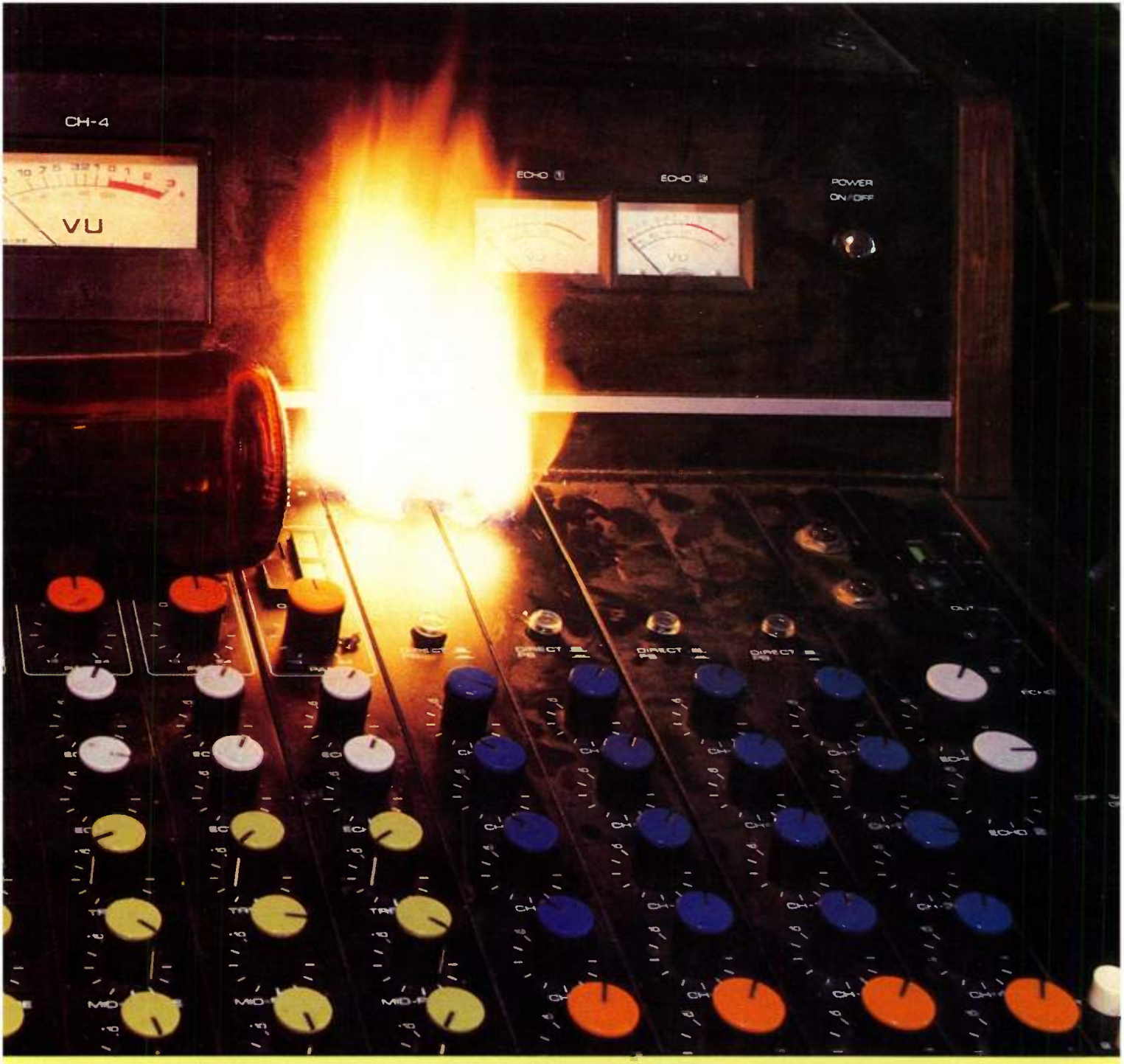
The Worst Sound Problem I Ever Had- And How I Coped



By David Hadler & Michael Berlin

Time heals all wounds, but even nine years can't erase the memories of our "worst sound problem." The lessons we learned remain and can be communicated to both retailers and their customers. One spring day in 1970 especially remains on our minds. Back in those days, back in the wonderful pioneering days of early rock and roll tours, there were only a few companies with the ability to produce reinforced sound for large shows: Hanley, Clair, Tycobrahe, McCune, and ours, the Festival Group. Everyone had his own way of doing things, nothing was standardized. Interfacing two systems was difficult. The monitors consisted of a few side fill speakers if any monitors were used at all, and the existence of a separate mixer for monitors was an impossible dream. It was time of custom-made equipment, inexperienced technicians, and disasters on a grand scale. Like one night in the big apple.

The Felt Forum at Madison Square Garden was packed for the first of two shows by a then-popular rock group. Outside, the line began to form for the second show, and inside, a reporter began to interview the band. The warm-up act had been great, and the audience was ready for the stars of the show to take the stage. The sound for the opening act had been fine; there had been no problems at all.

The PA consisted of JBL drivers and enclosures, Crown amplifiers, and a custom-built 20 x 6 mixing console. The manufacturer of the console we were using must, for friendship's sake, remain anonymous.

The lights dimmed and the reporter took his place on stage, ready to review the performance. It was not an ideal time for our equipment to malfunction, but then there are so few good times for disaster to strike. Everything went fine at first, but the situation rapidly deteriorated. Perhaps it was the large integral power

supply that freaked out, or perhaps it was an unstable design, but by the third song, we knew that we were losing the mixer. The sound was all wrong, the vocals sounded terrible. The input EQ was producing high levels of third harmonic distortion, and switching inputs didn't help. Suddenly, the whole board went gonzo berzerko with distortion. Odd order harmonics were all over the place. We saw our relationship with the client dying along with the board. About that time the manager wandered over to casually inquire if there was a problem.

One of the problems that touring sound companies face is the repair of equipment on the road. Competent people are hard to find, and it was usually not feasible to send a repair technician out with every system on the road. Standard procedure was to find a local electronic engineer who could do the emergency repairs with the tech manuals for reference. Unfortunately, most custom-built equipment, including our board, did not have an intelligible manual, and the designer of the piece was the only one who really understood what was going on inside the piece.

The problem with this particular piece of custom equipment was its lack of modular design. All twenty inputs and the power supply were on a single chassis, and each input pot and its EQ were encapsulated in epoxy. Access was, to say the very least, limited. To add to the servicing problem, the construction of the console did not contain a mother board. This being back in the days when integrated circuitry had not yet overtaken the performance specs of discrete components, it meant that beneath the calm facade of a well engineered piece of electronics lay a mass of wires so thick it could choke a goat. Transistors, capacitors, and resistors stuck out of the spaghetti like mushrooms growing haphazardly.

The board had just come back from the manufacturer, where it had been overhauled. Everything was supposed to be fixed, and it had worked perfectly during the sound check. We were afraid to mess with the thing because we didn't know what was what inside. Besides, the last time someone had tried to fix the thing on the road it had revolted, sending out such high levels of 100 kHz signal that it had set fire to every speaker cone in the system. It was obvious that the mixer was not serviceable, and we began to wonder what we could plug into its place as a substitute. We had no monitor mixer, and no backup equipment. No one in town could help us out, and the first show was coming to an end anyway. It had progressed, as concerts often do, in the face of great adversity and even greater distortion. As the guitar player walked off the stage he shouted angrily, "We got the only PA in the world with a fuzz-tone built in."

The manager was furious by this time and we were afraid for our lives. The band was even more mad. They threatened us with torture and we became even more afraid. We were unprepared for this contingency, and had to do something by the time the second show began. If necessity is the mother of invention, then fear of death is the mother of fast action. In just 20 minutes we had located the bare minimum as a replacement for our board, and installed it. It was a six-channel mixer suitable for home use. It certainly wasn't the most advanced piece of equipment at the show, but it put out a fairly good signal. We grouped the mics into the six inputs, adjusting the amp levels and microphone placement to even out the volume differences within each group. There could have been problems if the demands of the show had been greater, but as it happened, no one needed much variation in individual microphone levels. The second show came off fine, but we have never been the same. We always carried a back up mixer after that.

Getting the system up for the second show is not the end of the story. Two people remained to be dealt with. The first conversation was with the manager of the band, and the second was with the manufacturer of the board. Neither situation was very pleasant. Asking questions is as difficult sometimes as answering them.

It was quiet in the dressing room after the second show. The band had not been very friendly since the failure of the board. We could hardly blame them; we should have been prepared. The manager opened the door and stalked into the room. "You'll never work for me again," he said. "You're finished—through."

"The first show was bad," we said, "But the second was all right; besides no one in the audience even knew what happened." It was true; the audience had loved both shows. Fuzz tones were big back then, and distortion was hip. (At both shows many people had come over to the booth to tell us what a great job we were doing with the sound; others came by to thank us for the lights.) "Why worry about it, it won't happen again."

"Why worry about it? Why worry about it? I'll tell you why worry! What about the reporter? What about the mess that came out of the monitors in the faces of the band? How were they supposed to sing without hearing themselves? I have a mind to drown you in that bowl of cole slaw."

"You wouldn't drown a man for poor taste in equipment, would you?" He would have, but in the end the tension began to show and he broke into hysterical laughter. Soon we were all tied in fits. Laughing until our sides split. Not really caring about the consequences and not wanting to think about sound, we proceeded to drink away our shame. Replacing it with relief that the ordeal was over, we slept all the way back to Philadelphia.

The next day, reflecting on the adventures of the night, we felt lucky to have escaped with our lives and equipment before the band had a change to reconsider the situation. We never did sound for them again, but overall the business prospered. This was partly because of the lessons we learned about being prepared, and partly because we returned our board to the manufacturer, and left it.

The manufacturer was really understanding, and it probably was the safest attitude for him to adopt. We could have minimized the disaster by having backup equipment, but the board itself was the root of the problem. We explained the deficiencies of the board to the builder in great detail, telling how hard it was to patch around the problem without a patch bay, and how difficult it was to service inputs encapsulated in a sea of wire. We showed him the necessity of modularity in the design, of isolation of the power supply, and of documentation. Then we put his board on the floor of his shop and walked out, never to see it again. The unnamed manufacturer must have learned something from this episode, because his products are now used throughout the sound reinforcement industry. His company is a giant by industry standards.

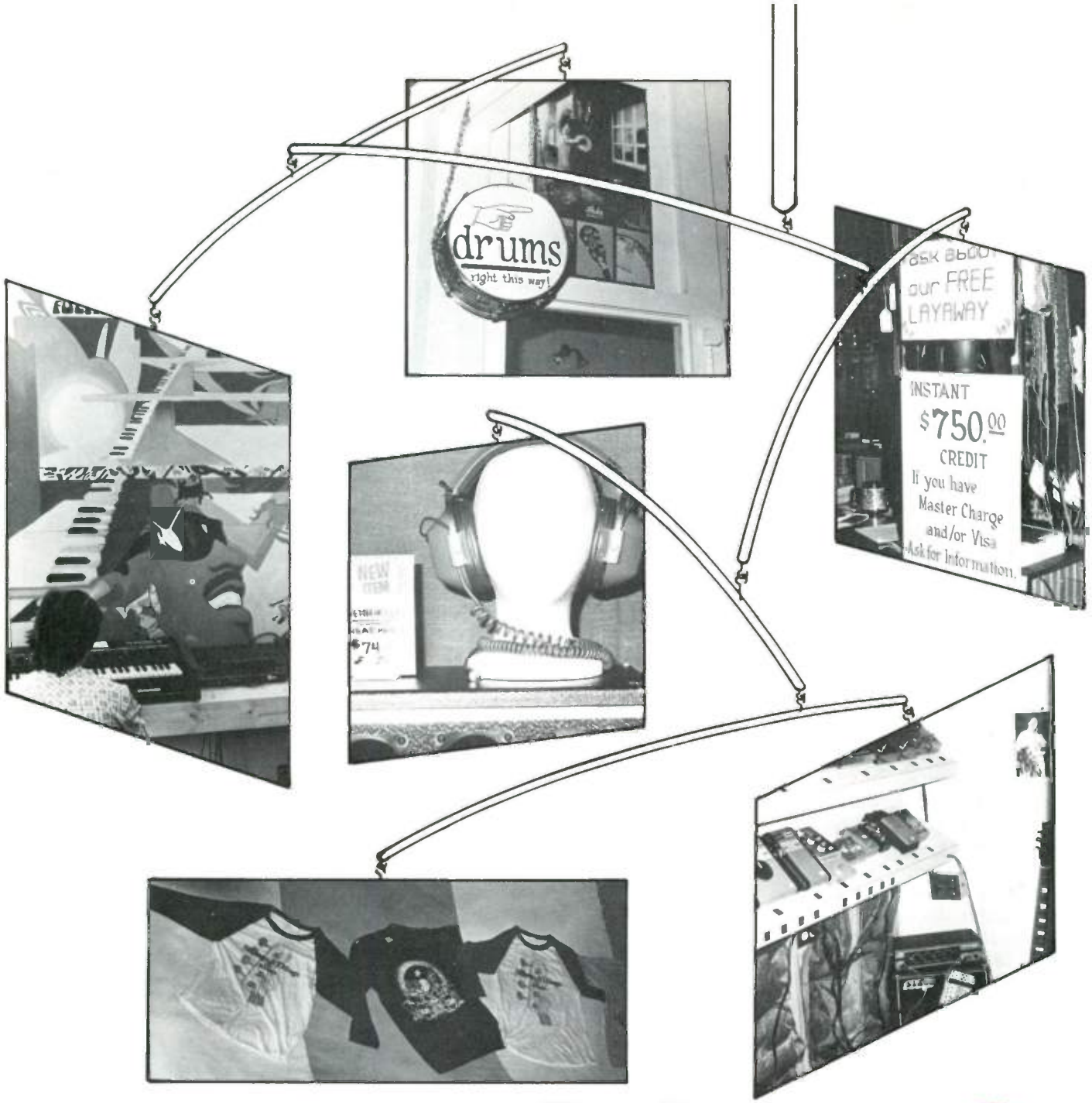
David Hadler is with AudioTools in Denver, and has worked as audio engineer for the Doobie Brothers, J. Geils Band, Traffic, Jefferson Airplane and others. Michael Berlin, also with AudioTools, has worked with F.M. Productions and Feyline, and has toured with Crosby, Stills, Nash and Young, George Harrison, Bob Dylan and others.

We learned some lessons, too. In addition to the obvious possibility that custom equipment may have design problems, you should consider the realities of touring. Tours take you far away from the designer of the board and his spare parts inventory. Use standard equipment that any competent technician can understand and repair with local supplies of electronic parts. Have a patch bay or some way of bypassing the potential problems you may encounter. It makes the location and replacement of the problem circuits much easier. Some of the problems you may encounter are sinister; they cannot be solved unless they are isolated from the rest of the system.

Most important of all, carry a backup for your vital equipment, especially electronics. There is no way in the world to absolutely guarantee that equipment will function as planned. Good maintenance and the use of dependable equipment will decrease the probability of failure, but what will you do if some idiot pours a bottle of Southern Comfort into your amp rack? Your backups don't have to be top-of-the-line, just good enough to get you through a bad night without catastrophic loss of sound. Carry extra amps, mics, and an extra board at the very least, not to mention a few extra drivers. These days many touring systems have a monitor mixer which can be substituted for the house board in a pinch, but by the same token more musicians than ever will not stand for having to perform without a mix suited to their particular needs. So have at least one extra mixer that can work on the stage or in the house, or risk facing an enraged employer.

In the end, the entire episode made only positive contributions to everyone involved. The unnamed band got a little extra publicity and a sympathetic review from the press for their stiff upper lip and good humor in time of crisis. In a couple of years they were history anyway. The sound company went on to a long string of successful tours for all of the heavies of the day. The crew had been seasoned by the incident at the Felt Forum, and we had learned the value of anticipating trouble rather than waiting for it to happen. The promoter made a bundle; they usually do.

Overall, it seems rather funny now, after nine years. We don't laugh too hard about it, though. It still hurts a little when we laugh.



BUILDING SALES: IN-STORE DISPLAYS

By Allen Hester

Mom's Music, Anyplace, USA: The minute you walk into Mom's, you can smell the pianos, all shiny and new. There they stand in a long line down the south wall of the store. Opposite them is the glass display counter, some 20-odd feet in length, filled with sax reeds, violin bows, and mouthpieces. Above and behind the counter (out of reach to you or me) hangs a row of new guitars, mostly low-end acoustics, with a few electrics thrown in for good measure. Near the front window, one lonesome set of drums and a four-channel high-impedance P.A. tell the outside world: This is a music store. But do people passing by really get the message?

There are so many storefronts and billboards and advertisements that most people filter out the bulk of the advertising barrage without even realizing it. There is a simply not enough room for all of it.

As a retailer, you are aware of this, no doubt. But how do you combat the built-in insensitivity to worn-out marketing techniques? Do you try to display your merchandise in a different way? Do you strive to give your store a little atmosphere of its own instead of relying solely on the display tools provided by manufacturers?

There are many things that can be done, and most of them are cheap and easy to do. Most importantly, though, good displays can save your sales staff a lot of legwork, and at the same time inform your customers of things they want to know. Taken together, those two objectives are all the justification you need for going off to the art supply house for display-making materials.

The simplest thing you can do is invest in a set of oversized marking pencils ("Magic Markers") and several sheets of colored poster board. Use a pencil and a T-square to outline your lettering, then color in your message. Big block letters work best; they are the easiest to read, and there's really no need to get fancy and sacrifice legibility in the process.

Hand-lettered signs can be used to inform customers of store policies such as lay-away and credit plans that they might otherwise be unaware of. Many times customers hesitate to ask about financing, but if they see the sign they feel more comfortable about asking questions. Posters can also be used to direct customers to different areas in the store, or to inform them of upcoming events concerning them and your store, such as concerts, clinics, and so forth.

Another thing you can try is to contact an outdoor sign company about using some of its discarded signs. Even better yet is the local record shop: They get huge posters and outdoor signs from the record companies to help promote new albums. You can usually get the old ones for free if you go pick them up. Here at Strings and Things we used one such sign in our keyboard department with great success. Customers can see this sign from across the store, from which point the keyboards themselves aren't even visible. But the customer gets the idea that there must be something going on "over there where the Stevie Wonder sign is." Likewise we use a similar discarded poster of Kiss in our drum shop behind a display of Pearl drums, the brand used by that group's drummer.

Drum shop manager Dave Patrick decided he needed some kind of sign that let the customers know how to reach the drum shop once they got inside the front door. (The drum area is set off from the rest of the store and is accessible by a hallway at the west end of the main sales floor.) So Dave took an old chrome snare drum, hand-lettered a sign on the head of the drum, and hung it from the ceiling so everyone could see it. This is a good eye-catcher, and it is obviously a home-made thing, which helps give the store a more personal atmosphere.

Does your store sell headphones? If so, you might like this idea. Go to a drugstore or discount department store and buy a few of those styrofoam heads that are used to display wigs. These things make great headphone displays because they are unusual-looking, and thus they draw attention very fast. Besides, these styrofoam dummies provide a way to display the headphones "in use;" they bring the headphones to life in the mind of the customer. Next thing you know, the customer wants to try them on, too!

However, novelty ideas such as these, no matter how clever, don't do the whole job that a good display should do. The eye-catcher needs to be supported by informative literature and/or signs that tell the customer something about the product he is looking at. What does it do, how much does it cost, where was it made, who uses it, and so forth. If the customer can stand there and read about it, then the salesman's job is easier because he doesn't have to spend time giving the customer a lecture on the origin and use of the product.

One of the things people want to know first is how much something costs. Cost alone often determines whether or not a customer is interested; sometimes cost is not a factor. However, if all the merchandise is clearly priced, then budget-minded people will gravitate toward those things they feel they can afford.

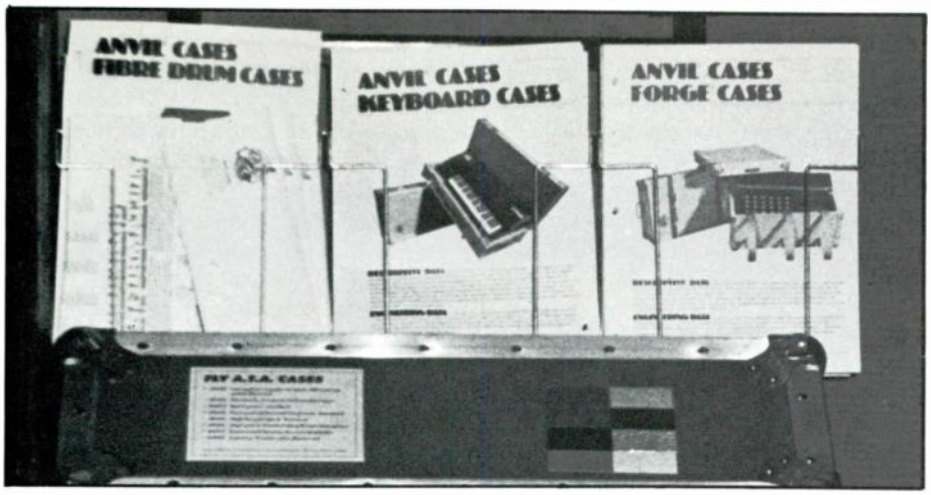
There are dozens of different kinds of price tags, ranging from the plain white ones to classy ones to those gaudy day-glo orange ones marked "Special Sale," "New Item," "Warehouse Clearance," and so forth. Get a good supply of several kinds; they are all useful at one time or another. Prices change so quickly that you have to keep a daily check on prices to make sure they are correct. Also you should check to make sure that all the prices are printed clearly so as to avoid any misunderstanding with the customers. Most office supply houses have some kind of adhesive-backed plastic card holders suitable for holding small placards in a display. Office supply catalogs also feature clear plastic literature racks that work very well with catalogs and brochures provided by manufacturers. These racks can then be personalized by affixing your store logo or emblem to them. These save the salesman time because he doesn't have to go scrounging around in the back somewhere for a catalog that the customer can take home.

Speaking of time-savers, how do you display effects pedals? How many times have you spent half an hour with a customer who wanted to try every pedal in your store? It gets a little frustrating after about the fifth pedal, especially if you have to walk back and forth to the counter, check each pedal for fresh batteries, and hook everything up while the customer fidgets.

One way you can overcome this, although it is a pretty involved project, is to build a permanent display of effects. At Strings and Things, there is a small room set aside for this. In it there are three shelves of effects, permanently mounted and connected to a power supply. In front of each effect is a button; push it and the effect becomes operational.

Customers can try any combination

Allen Hester works at Strings and Things in Memphis.



of gizmos without taking away valuable time from the salesman. Even more helpful is the fact that the room is insulated. Customers often become irritated at the screaming and howling of someone else's rock guitar; likewise a customer may get offended when you ask him to turn down while he is trying out a new pedal. So the sound-proof booth lets both customers shop in the best possible atmosphere. This effects booth is by far the most costly and time-consuming display idea yet presented, but if you sell a lot of effects, it might be worth looking into.

Regardless of what you sell, you need to print up some T-shirts. People love to wear them; they are profitable on the front end, and they carry the name of your store to a lot of places. The first T-shirts that Strings and Things ever printed up were line drawings of vintage guitar headstocks, surrounded by a laurel wreath. People like to wear their favorite instrument around on their chest, and don't mind if your logo is on the back.

What about point-of-purchase displays provided by manufacturers? These displays are often included in the shipments of merchandise you receive, as are posters and T-shirts. You can forget the T-shirts; your employees will descend on them like hungry wolves. But use the posters and POP displays.

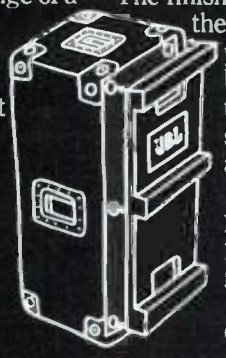
Many manufacturers have good displays, but sometimes you have to ask for them. Your sales rep may forget to mention them, so the burden is on you to see that you get all the help that a manufacturer can offer in the way of displays. You may find that some of the manufacturer-supplied displays do not fit the mood of your store; indeed, there are some that work well in a piano and organ store that don't go over in a rock shop. You can decide, though, only if you have the display in hand in the first place. If nothing else, you can make another front for the display out of poster board and staple it to the display face.

Good displays inform and enlighten the customer. They also save the salesman time and make the job of selling easier. Personal touches in your store give it atmosphere, hopefully one that relaxes the customer and makes him feel at home. If your customers are relaxed and comfortable, they are more likely to buy from you.

NOW SOUND AS GOOD AS YOU PLAY. JBL'S NEW CABARET SERIES.

Both you and your audience deserve to hear exactly what you're playing. Which is exactly what JBL's Cabaret Series is about.

These new systems have the power-handling to reach every listener with every note. Your audience hears the lowest bass. All the vocalist's lyrics. The full range of a multi-octave lead.



And it's heard accurately. Undistorted. Uncolored.

There are three Cabaret Series sound systems: lead instrument, reinforcement and stage monitor.

The lead instrument system features JBL's patented interlocking stackability. All three systems feature JBL's famous K Series musical instrument loudspeakers. All systems are fully portable. And built to endure the punishment of the road.

Their housings are made of the best birch ply available. The front covers fit flush to protect the drivers and grilles during travel. The finish is extra-rugged. Even the voice coils have unique protection: To minimize bounce during transit, they're automatically shorted when there isn't a plug in the input jack.

With these JBLs you don't have to invest your sound system budget in traveling cases.

The Cabaret Series offers enormous value in other ways. For example, one JBL column probably puts out as much sound as four of the systems you're now using!

And JBL quality assures you of avoiding costly "down time." It's the same quality standard that so many pros rely on.

Hear JBL's new Cabaret Series soon. Because your sound system should be as good as you are.

JBL

FIRST WITH THE PROS.

See and hear JBL's Cabaret Series at NAMM, Booth #4002.

CIRCLE 71 ON READER SERVICE CARD

a chip off the old circuit block

part III

digital audio applications

by Mike Beigel

An audio signal, once converted to digital format, enters the "world" of computer processing. Virtually any conceivable operation can be performed on the digital audio signal, given a computer that is fast and powerful enough. That sounds ominous and expensive, but computers (and digital circuits that perform computer-like functions) are increasing in power and decreasing in price.

Without delving into the complex techniques and methods of computer processing, I will describe some of the functions that have been developed in digital audio processes. Many of these attainments are still confined to university and commercial development labs, but they will inevitably appear in product designs in the near future.

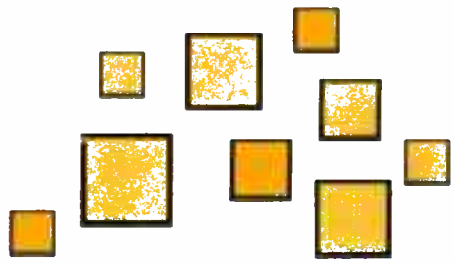
DELAY AND REVERBERATION PROCESSORS

The simplest and most inexpensive processors are the "digital delay lines." They consist of analog-digital conversion hardware, digital "memory" for storing and delaying the digital audio signals, and various means for producing musically interesting effects with the delayed signals.

The digital hardware depicted in figure 1 simulates a delay system of a more familiar nature, shown in figure 2. One of the various properties of digital systems is that they can simulate many different processing structures. In this case, the way the system behaves is not the way it's wired up. Figure 1 is the way it is "wired up," and figure 2 (The Virtual System) is the way it acts.

In commercially available delay processors, the system may not be all digital inside. The technology has not stabilized, and many economic trade-offs are needed to provide acceptable audio quality at a reasonable price. For example, a digital delay line using only ten binary digits (with a 60 dB dynamic range) may be the only purely "digital" part of the system. It is surrounded by a compander to increase its dynamic range, and all the mixing and feedback are performed by traditional analog circuits. (See figure 3.)

To simulate natural room-reverberation, a complex array of delay, feedback and mixing elements is necessary. Processors of this type are generally too complex to use the mixed digital-analog technology just described. They rely on sophisticated (and currently expensive) digital systems to perform all the delay, mixing and recirculating functions. These processors represent the highest forms of technology available in commercial equipment, and a look at figure 4 indicates why.



HARMONIZERS

A musically valuable effect is the transposition of a musical sound (instrument or voice) to a different pitch, allowing automatic "harmonizing" with the original musical signal. The principle is not too complicated: The musical signal is stored in a digital memory similar to an ordinary delay line. It is then "read" to the output at a different clock rate than was used to store it. This would be like tape-recording the signal, and playing it back twice as fast, or twice as slow. The problem: If you play it back twice as fast, it takes half the time; or if you play it twice as slow, it takes twice the time. Clearly, a phenomenal job of "splicing" is needed to either fill in the gaps and provide extra sound when the tape runs out, or to discard extra

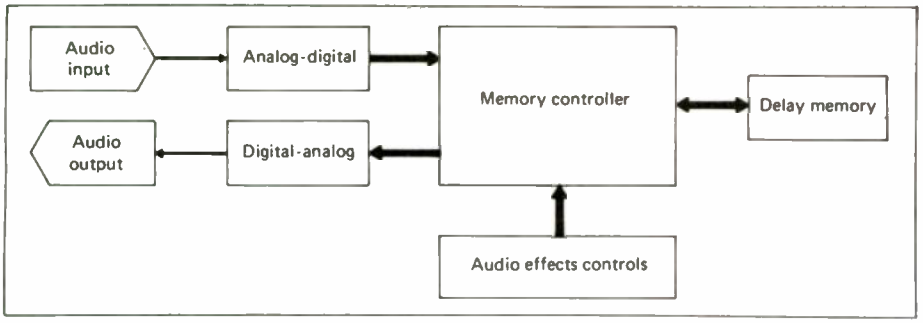


Figure 1 Digital Delay System

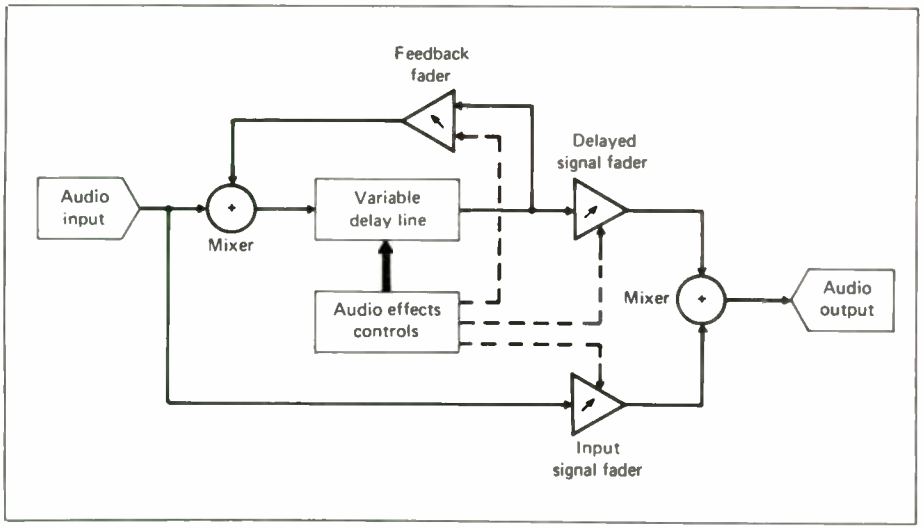


Figure 2 Digital Delay "Virtual System"

tape when the tape is too long.

The harmonizer therefore acts as a very sophisticated tape-recorder-splicer-timekeeper; to adjust the delay length in proportion to the harmonizing ratio, and to re-circulate or discard audio signals to create the impression of a continuous "harmonized" output. This "splicing" occurs at a rate just below audio frequencies, and the sophistication of the splicing technique determines how natural the output will sound. As yet, there is no "perfect" method for harmonization, but there are harmonizers presently available with quite acceptable musical quality.

DIGITAL FILTERS

Musical signal processing relies heavily on the principle of filtering: selectively amplifying certain audio frequencies while attenuating others. Digital systems can be constructed to simulate all types of filtering operations; including high-pass, low-pass, bandpass, phase-shift and more complex varieties. Surprisingly, the structure of a digital filter stage is almost exactly the same as that of a digital delay line. (See figure 5.)

The digital input signal passes through a delay, and the signal is fed back and forward through multipliers which act as variable attenuators. The main difference between a digital delay line and a digital filter is the length of the time-delay. In the delay line, the delay time is usually between one millisecond and one second; whereas in the digital filter it is almost nonexistent: a couple of microseconds. Enough time, though, that by properly selecting the multiplier ratios and cascading filter stages, any mathematical wizard can obtain any filter characteristic desired.

Digital filtering, while quite versatile (and unbelievably complicated), is not yet ready for widespread commercial application. The digital circuitry required is still on the leading edge of technology, and many subtle problems prevent the ultimate goals of superb audio quality at a reasonable price.

DIGITAL TAPE RECORDING

Recording and storing audio signals has always been one of the weakest links in the chain of audio signal processing. Tape recording has problems with noise, distortion, wow and flutter, print through, frequency response, etc; and disc recording has another set of limitations related to the physical characteristics of the disc-stylus system. Digital technology provides a virtually uncompromised solution to these problems.

Once an audio signal has been converted into a sequence of digital numbers, these numbers (zeros and ones) are recorded on the tape. The recording process is optimized for the rapid transmission of a fast sequence of zeroes and ones, instead of a continuum of analog values. Since digital codes are being transmitted instead of analog voltages, the possibility of distortion is virtually eliminated. The digital codes may even contain "error detecting and correcting bits" so that, even if a transmission error occurs, it can be detected and corrected at the

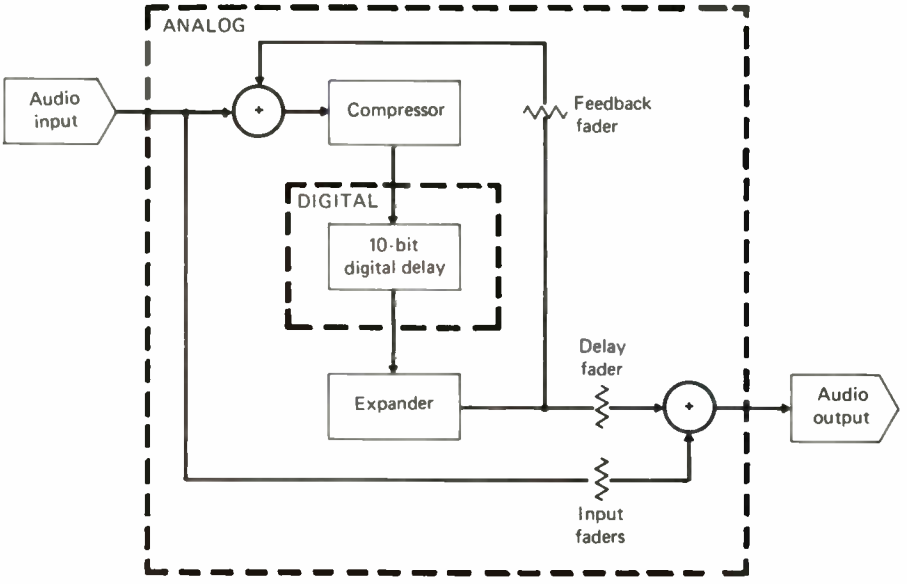


Figure 3 Digital Delay with External Analog Processing

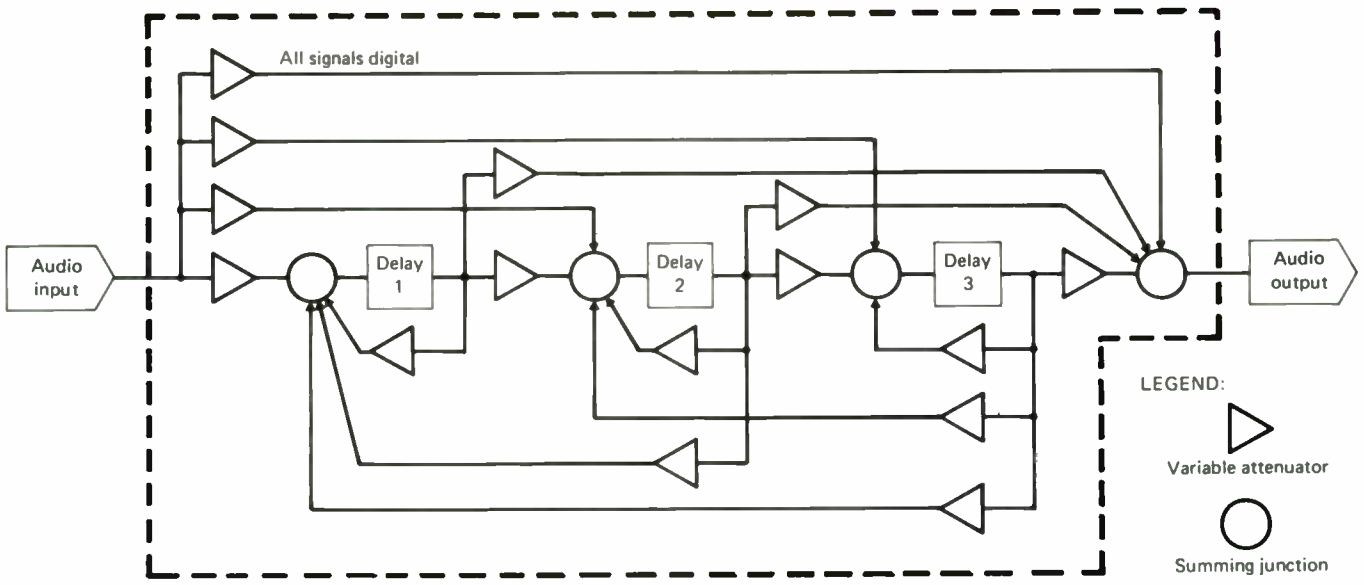


Figure 4 Simplified Digital Reverberation Processor

receiving end.

Wow and flutter are eliminated by a digital storage system that receives information from the playback head at whatever rate it arrives, but "clocks" it out to the digital-to-analog converter at a precisely regular rate. (See figure 6.)

WHAT THE COMPUTER DOES

Webster's Unabridged Dictionary defines computer as "an electronic machine which, by means of stored instructions and information, performs rapid, often complex calculations or compiles, correlates, and selects data."

First, you must accustom yourself to the idea that the device which fits the above description by Webster or his assistant also fits in the palm of your hand. And that the associated electronic devices necessary to make the computer perform as a "computer system" can fit on a circuit board the size of this page. And that the newest of these computers can handle 16-bit data (the optimum word size for high quality audio signals) at processing rates fast enough to deal with "real-time" audio signals. (Real-time means that the signal processing occurs fast enough so that the audio "output" is available with no perceivable delay.)

The computer is programmed to perform its rapid and complex calculations on audio signals. Not only audio signals from an outside source, but also internally generated audio signals. Music and speech can be synthesized as well as modified. They can also be analyzed. Analysis functions include automatic recognition of the spoken words, "pitch" detection, even automatic transcription of music into a visual form. Frequency spectra (the visually and mathematically represented "signatures" of different musical and acoustic signals) can be generated from the analysis of musical sounds, and then re-synthesized to accurately simulate these sounds.

These computer functions have not reached the marketplace, but prototype systems already exist at such institutions as Bell Labs and some universities. Having seen, heard, and worked with such systems, I am convinced that they will revolutionize every aspect of the process of making music. Scientific papers describing these systems have appeared in the *Journal of the Audio Engineering Society*, and new publications such as the *Computer Music Journal* devote themselves exclusively to coverage of this emerging discipline.

In commercial products, microcomputers have not yet reached their full potential of processing audio signals. The more powerful processors are still too new to appear in product designs, and smaller processors are capable only of performing control and

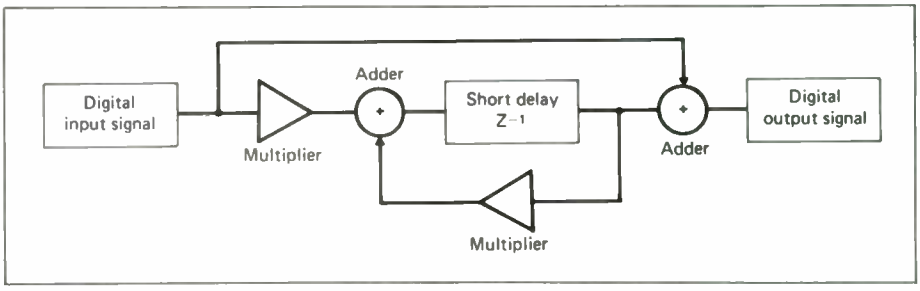


Figure 5 Digital Filter Stage Multiplier

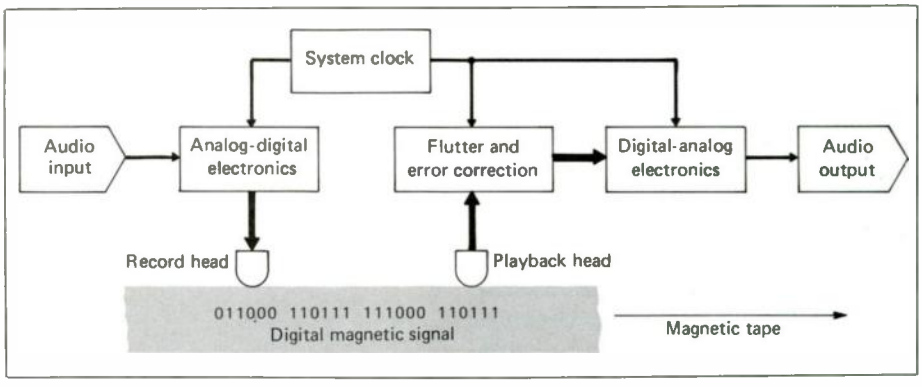


Figure 6 Simplified Digital Tape Recording System

We're shipping them at last.

The Gizmotron® turns an electric guitar into an entire group of string instruments.

In the past two years, the Gizmotron has received more international advance editorial publicity than any other musical product in history.

It has even been on TV.

The editorial coverage was well deserved, as the finished product demonstrates.

The brainchild of Kevin Godley and Lol Creme, the Gizmotron is simple in concept, and difficult in production.

The idea of an automatic bowing device for stringed instruments (in this case, the electric guitar) has been around since the 1400's. The device has eluded engineers, including the great Leonardo, who worked on the problem for years.

The fact that the Gizmotron now exists is owing in part to an advanced technology which had to be created for it, and the use of sophisticated space-age materials which assure maximum quality and trouble-free service.

The Gizmotron is covered by extensive international patents.

We started shipping the first Gizmotrons in May.

They are heavily back-ordered, and are being shipped on a first-come basis.

And they are about to be heavily advertised, now that they are available.

Almost every major artist who records with an electric guitar has either ordered a Gizmotron, or is now recording with it.

The Gizmotron carries a suggested retail price of $250.

Think of this.

There are more than 3,000,000 electric guitars out there, just waiting for the Gizmotron.

Gizmo Inc.

For further information write Gizmo, Inc., P.O. Box 139, Dept. B. Rosemont, New Jersey 08556 or call 609-397-2000. Visit our booth 6001 at the NAMM Int'l Music Expo, June 9-12, 1979, Atlanta, Georgia.

CIRCLE 64 ON READER SERVICE CARD

WRH

sequencing functions in predominantly analog equipment. However, anybody who has played some of the newer programmable synthesizers will testify to their usefulness.

The machines used for digital audio processing at the present time are special purpose, computer-like systems designed to do specific jobs with very high speed and efficiency. The system at Bell Labs, for example, uses special processors for digital oscillator banks, filter banks, delay and reverberation and high-speed interconnection of the system. These special processors are all controlled by a microcomputer, which, like the conductor of an orchestra, instructs each "instrument" to play the right notes in the right way at the right time.

The Bell Labs system can simulate woodwind, brass, string and percussion instruments; generate "electronic" sounds which are really beautiful; sing phrases in harmony; and provide quadraphonic spatial phase-shifting. It allows the user to record, edit and correct musical compositions with push-button convenience. It is also a laboratory for musical research and development.

Bell Laboratories All Digital Synthesizer

GET READY

Digital techniques will enter into every aspect of musical and audio products and processes, and the full impact of the digital computer revolution on the music business is hard to foresee. We know that the music business is technologically "backwards" compared to economic areas of higher volume, or more heavily supported by Research and Development funding. So we can project the progress in music and audio by looking at the more advanced "high-tech" fields in our society.

innovate your sound

with sunn®

For more than a decade we've pushed the limits of electronic amplification and sound reinforcement equipment. We've established ourselves as the sound innovators. Our commitment is to develop imaginative concepts that give you the opportunity to discover your own unique sound. But it's really more than that. Our amps and sound systems *are* different. They're built for you. But, we go at it like they're built for us.

That's why we applied C-MOS (Complimentary MOS) circuitry to our Beta amplifiers. C-MOS was first used in the computer industry. We had no idea there was music in C-MOS until we tried it. The result is a unique overdrive circuit that is rich in harmonic content, much like tube distortion. But C-MOS does more than tubes. Distortion can be added in any amount, at any volume, so you don't destroy your ears to get that hot sustained sound you want. Yet in the clean mode the Beta is clearer, tighter than tube sound, so the characteristics of your instrument shine through.

Most amplifier people use standard speakers. We don't. We make our own so we can capture the subtle sounds that make each of our products unique. And that is what makes your sound unique. Because you can count on those differences to bring out your best. Play the fantastic Beta Series at your Sunn Sound Dealers.

Sunn. The Sound Innovator.

Beta Series Self-Contained Amps

Sunn endorsees include Chris Squire of Yes (shown above), John Entwhistle with The Who, Steve Buslowe with Meat Loaf, John Deacon with Queen, Joaquin Lievano and Jamie Glaser with Jean Luc Ponty.

Sunn Musical Equipment Company A Hartzell Corporation Company Amburn Industrial Park Tualatin, Oregon 97062

CIRCLE 66 ON READER SERVICE CARD

The SOUND SH

Only three years have passed since **Music Man** introduced the Sting Ray bass guitar, the precision-looking electric bass which has an oval pickguard and a rather odd headstock design. This bass became popular almost overnight; professionals from every corner of the music business found the Sting Ray bass a useful instrument, probably because there is more to the bass than what you see at first.

For one thing, the Sting Ray has an on-board preamp which powers the separate bass and treble controls, thus providing a very broad tonal range. Other features, such as a low-impedance output, individually adjustable mutes on each string, and a heavy cast bridge have all helped make the Sting Ray a popular bass. That this bass is so popular is no accident; the Sting Ray was designed by Leo Fender, the man who designed the Precision bass guitar that has long been the most popular bass ever made.

Now Leo and his constituents at Music Man have another bass to add to their line—the Sabre. Those of you who are familiar with the Music Man line know that there are two electric guitars currently offered—the Sting Ray and the Sabre, the Sabre being the more expensive of the two. Likewise the Sabre bass will cost a little more than the Sting Ray model, but the Sabre has some added features that parallel those on the Sabre 6-string.

For instance, the Sabre bass has two pickups and a three-way selector switch instead of just one pickup. The on-board preamp is there, as well as a bright switch and a phase reversal switch. The tonal range is even wider here than on the Sting Ray bass; the punch is there; the clarity and high end definition is there; the only thing that isn't there is a dead spot on the fretboard. Dead spots were inherent to bass guitars until Music Man eliminated them by using a preamp to smooth out the response of the pickups.

This Music Man stuff is great as far as I am concerned. Granted, they are not as "pretty" as other brands, because most of the bodies are finished in a clear acrylic finish with no paint and no frills. But when it comes to playing ease and sounds, the Music Man instruments offer more than a lot of others.

Most importantly, though, the instruments are priced right. They are well within the reach of the young musician, and the price looks even better to the crusty old studio veteran who bought his last guitar in '65 and thinks everything is too high nowadays!

I look for these new basses to catch on fast, and from all indications there seems to be little doubt that musicians have taken to the Music Man line in a very big way.

CIRCLE 1 ON READER SERVICE CARD

The **Teac** Corporation recently introduced some new cassette decks, the A-500 and the A-510, which are made to be used with ferrichrome or cobalt tapes. The two new units are similar in that they both have left and right record controls that are clutched, mic/line switching, memory, Dolby noise reduction, three-position bias and EQ, and record, rec/mute and pause buttons.

OPPE

By Charlie Lawing

Both the A-500 and A-510 have erase and record/playback ferrite heads and FG servo-controlled motors. The difference in the two units is the bar graph meter found on the A-510. This meter has a peak hold and auto/manual reset switch which enables the user to hold peak readings for three seconds or hold them indefinitely and reset automatically or manually.

As far as metal tape goes, Teac dealers should know by now that Teac is offering to refit the high-end model C-1 cassette deck with the proper bias circuitry and erase head to make the unit compatible with metal tape. The record/playback head in the C-1 is already capable of handling the metal tape.

CIRCLE 2 ON READER SERVICE CARD

Many of us can remember the days when a velvet-voiced crooner stood in front of an old ribbon microphone with his arms outstretched and eyes tilted upward, cooing out a chorus of "Deep Purple." Or something like that. Anyway, about the most rugged treatment a mic got in those days was when the singer tilted the mic stand and leaned over to the front row audience—ever so gracefully.

Nothing could be further from reality today. Microphones are something akin to billy-clubs in the world of rock and roll. Singers twirl them, throw them, bang their stands on the stage, and occasionally just drop them in a moment of frenzy. Needless to say, the old ribbon mics wouldn't last one gig, much less a major tour.

Thus we have the "new breed" of microphone, battle-tested in the concert halls by carpenters, discus throwers, and baton-twirlers. Any product that emerges reasonably intact from such grueling test conditions is deemed fit for consumption by the stalwarts of rock and roll.

It is in this spirit of extreme hardship that **Electro-Voice** has designed the DO56 omnidirectional microphone. This mic is intended for hand-held broadcast and sound reinforcement use, which means it must not only perform for the rockers, but for the humble newsman in the middle of a government revolt as well.

In order to achieve greater stability, E-V has isolated the main acoustic cavity and the diaphragm/voice coil assembly from the mic case itself. By doing so, the possibility of a capsule/case collision is virtually impossible, even under the worst conditions.

The case itself is made of steel and aluminum, chosen, according to E-V, for the best possible hand-held balance. The new Memraflex grille screen is designed to resist denting, and the silver tone beige finish looks good.

As if that wasn't enough, the frequency response has been doctored up, which should come as great news to lousy singers! The DO56 has a slight emphasis in the 2 kHz-20 kHz range, which enhances the vocal qualities of the mic, and the frequencies below 200 Hz have a slow rolloff that reduces rumble. Also, the DO56 has a high-density "Acoustifoam" blast filter that helps eliminate "P-popping."

CIRCLE 3 ON READER SERVICE CARD

New from **Acoustic** is an eight-channel stereo sound reinforcement mixer, the Model 924. The mixer is housed in a wood and fiberglas road case for protection while traveling.

Each of the eight channels on the 924 has the following features: low and high Z inputs; lighted overload indicators; three-way gain switch (−40, −20, −0 dB); individual bass, mid and treble EQ (+ or −18 dB); reverb and/or effects send; monitor send; pan pot; and main level fader.

The master control section of the 924 has a master level fader; master reverb effects control; and an eight-segment LED light bar display for overload indication.

On the monitor channel you will find a mas-

ter monitor level fader and a monitor reverb effects return control. In addition, the 924 has two nine-band graphic equalizers (+ or −18 dB), one on each master channel.

The reverb is of the internal spring reverb variety, which can be overridden on each channel individually by any external effect, such as echo or digital delay devices.

Distortion of the 924 is rated at 0.1%. The maximum gain for a 1 volt output using a low Z input is 66 dB; with a high Z input, the maximum gain is 46 dB. Headroom above normal for each individual channel is 40 dB; for the master channel the headroom is rated at 18 dB. Shipping weight is 55 lbs.

CIRCLE 4 ON READER SERVICE CARD

Ahh, here's the kind of low-cost goodie that sends your customers into ecstasy: a line output converter merchandised by **Musical Research Laboratories**. This is a simple looking device that has a variety of applications, most of which your customers may not have even thought of, but will immediately be anxious to try.

The LOC-1 contains a small circuit which requires no battery replacement. It is designed to be plugged into the extension speaker output of any amplifier and provide a line level signal suitable for the line input of a tape deck or any device with line inputs.

In sound reinforcement and studio work, close-miking techniques are often employed. However, this technique is not without its inherent problems: mic distortion, altered frequency response, ambient sound leakage, and cone distortion, not to mention the high cost of mics suitable for this technique.

The LOC-1 eliminates the need for close-miking by presenting an accurately converted signal from the amp which can then be fed directly into the console.

If the amplifier being used does not have an extension speaker jack, a simple "Y" cord can be plugged into the speaker jack, one side of which is used for the speakers and the other for the LOC-1. In addition, the LOC-1 can be used without the speakers on the amp at all, but in such a case it is recommended that a "dummy" load be presented to the speakers to avoid damage to the amplifier's circuits.

Rock musicians often prefer the sound of a small tube amp that is driven into distortion, but cannot use that amp on stage due to inadequate volume. By using the LOC-1, the small amp signal can be preserved intact and used to drive a larger power amp for plenty of additional volume.

In video applications, video taping crews who cannot get a good audio signal can use the LOC-1 to provide a signal from the P.A. which will drive VTR's and movie camera line inputs, as well.

Many other recording applications are possible with the LOC-1; and its sister product, the LMC-1 (a line-to-mic-level converter), is equally useful.

The best thing about this device is the price: only $9.95, and it could encourage the average guitar player to further investigate sound reinforcement techniques.

CIRCLE 5 ON READER SERVICE CARD

ATS Cases has introduced a new "affordable alternative" line of transportation cases. Combining the style and weight advantages of one-piece construction with the strength

and reliability of solid core plywood, the cases feature a plastic exterior laminated to a plywood core and polyester foam lined interior, tongue and groove valance closure, steel corners, full length piano hinge, and for guitar cases, both contour cuts and plush velvet lining. All ATS cases are available in three colors: blue, silver gray, and black.

CIRCLE 6 ON READER SERVICE CARD

One of the most respected names in audio is **Crown**, makers of the near-legendary DC-300A power amplifier. For many years, that was the amp to be reckoned with by other manufacturers of high-quality amplifiers.

This year Crown has a new amp on the market, the SA2 Distinction Series stereo power amp. The SA2 is rated at 220 watts per channel minimum RMS into an 8 ohm load, with a THD rating of less than 0.05% (20 Hz-20 kHz). Under a 4 ohm load, the output per channel is 350 watts RMS.

The two channels in the SA2 are essentially separate, each having its own power supply and circuitry. Also, the SA2 has a cooling system which will keep the amp cool even in rack-mounted enclosures. A two-speed rear-mounted fan and thin aluminum heat-sink fins are employed, in conjunction with a removable dust filter and air-flow grilles on the sides and top of the amp chassis. Air can even escape through the front panel grillework if the amp is otherwise enclosed and unable to "breathe" any other way.

The SA2 has a front-panel display of two LED output level indicators, both of which have a range of 42 dB, much higher than conventional VU meter displays. These LED's also indicate, by means of color-coded lights, the status of the amplifier at all times: green if operation is safe; red if overload or clipping is occurring; and amber if the amplifier has disconnected itself from the speaker load to prevent overload damage.

Additionally, the SA2 has three rear-panel switches which perform important functions: One switch overrides the seven-second turn-on delay; another switch converts the amp to mono mode; and a third switch enables the user to defeat the low-frequency protection built into the amp. In normal operation the SA2 monitors low frequency signals and filters them out to prevent unnecessary speaker overload; however, the user may want such sub-audio frequencies (especially in a disco setting), in which case he can have them by flipping the switch.

For all the features mentioned above, the most unique aspect of the SA2 has yet to be mentioned. Believe it or not, this amp has a built-in computer circuit which monitors the amount of stress under which the power transistors are operating. The computer takes a "look" at the load being presented to the amp, then scans back through the "history" of the amp's power transistors, and decides in an instant whether or not the amp can handle the load being presented.

This unique computer feature is a result of new testing technology developed by Crown to measure the safe operating area of power transistors. That SOA had previously been limited by designs, even though engineers knew that many power transistors could perform well beyond their existing limits. Those old limits were designed to deal with "worst-case" conditions, and were therefore very low.

To test their design, Crown engineers fed a pink noise signal into the amplifier circuit under a 4 ohm load. Test instruments indicated an instantaneous peak power of 900 watts per channel; under a 2 ohm load the amplifier circuit developed 1500 watts per channel. Crown concluded that the SA2 will provide sufficient power to handle any music peaking requirements without overloading or limiting.

CIRCLE 7 ON READER SERVICE CARD

By Joel A. Knutson

DEALER DOSSIER

AVC Systems
Minneapolis, Minnesota

AVC Systems, Inc. of Minneapolis has become a major force in the upper midwest professional audio market since assuming its present corporate identity and structure in 1975. Surprisingly diversified for a company of its size, AVC Systems handles a broad range of equipment and product lines. The firm lives by a systems approach for meeting the needs of clients in a half-dozen distinct professional audio markets. AVC's progress is due to the work and creativity of its two young owners, Douglas Ordon, president/treasurer, and Jon Bormann, vice president/secretary.

AVC Systems is a relatively mature company in professional audio.

Bormann: That's true in a couple of ways. AVC Systems has been in business about 20 years, which I guess makes it a kind of pioneer firm in the professional audio industry. But the company has existed in its present form only since 1975, when four of us purchased it from the original owner. Within a year, there were just the two of us: Doug and I. At that time, we decided to make a commitment to the business and to make as many contributions as we could to the audio industry. At the end of that first year, we had four staff members. Since then, as a company, we've grown up fast.

You claim that AVC Systems in unique in the upper midwest pro audio market. Why?

Ordon: I guess our primary justification for that claim is that there is simply no other company serving the market the way we do. The pro audio market has become so varied in recent years. We're dealing with recording studios, broadcasting stations, performing arts theaters, corporate theaters with extensive audio-visual capabilities. We also serve live performing groups and discos. Each of these market segments has its own specific needs and problems in terms of audio. We address all of these markets, so that gives us a unique breadth.

Bormann: The other dimension we try to work with at AVC Systems is depth. We offer a very broad line of equipment and services, but all of that wouldn't amount to much if we didn't have the depth to back it up. One of our unique attributes is that we can not only sell . . . but also design, install and service highly complex audio systems. We start by working with the client to develop a concept that will get the job done for him. By the time it's all over, we've taken him every step of the way . . . right down to the location of the light switch on the wall. That's the kind of comprehensive, in-depth service we feel we have to offer. And we do it for each of those audio market segments Doug mentioned.

Ordon: You see, we're not only a sales and service organization, but we're a stocking company as well. That's something else that sets us apart. Most of our competitors do not stock product. They might have a few samples, but that's about it. Our business is based on a stocking program. We make it a point to have most of the merchandise in stock. And because of the technical nature of our company, we can show the customer not only the equipment he wants to see, but the technical measurements on that equipment versus a competitive product.

Let's talk a bit about the various markets you've referred to.

Bormann: Well, recording studios would have to come first. That market is a great source of dollar volume for us. I'm talking now about a music studio or a voice recording studio.

Ordon: Our primary service area is Minneapolis-St. Paul. There are prob-

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CIRCLE 54 ON READER SERVICE CARD

ably 100 or so established recording studios in the Twin Cities area . . . and hundreds more free-lance recording operations. Then, too, we serve the entire upper midwest. I guess our major achievement in the recording studio market would be the work we've done for Sight & Sound in Omaha. We sold, designed and installed a studio for them about two years ago, and it worked out so well that we just finished putting another one in for them. We have several other installations outside the Twin Cities. Des Moines and Fargo, North Dakota, for instance. We regularly serve a five-state area which includes Minnesota, Wisconsin and the Dakotas. Then there are five other midwestern states we get into from time to time. Chicago is included in that market. We have exclusive distribution rights for various products in this entire ten-state region.

What would be the range of investment for the recording studios in your service area?

Bormann: Anywhere from $3,000 to $3,000,000. But the average range of equipment we would sell and install would probably fall between $5,000 and $80,000. Sales of $15,000 to $20,000 are common for us. A large sale would probably be in the range of $50,000 to $60,000 . . . and our large sales are getting larger all the time.

Do you find the same kind of growth potential in the theater market?

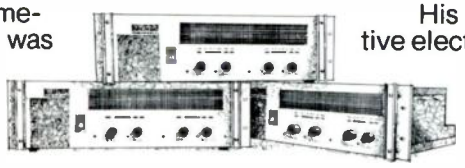
Ordon: Definitely. In the theater area, we have two major categories: performing arts and corporate theaters. Corporate theaters are getting pretty sophisticated. They're usually set up for 16 mm film and 35 mm slide projections, and need lots of audio to back up the visual portions of their sales and training programs.

Bormann: We've sold and installed audio systems for many of the major corporations here in Minneapolis-St. Paul: Control Data, 3M, Northwestern Bell, General Mills. Some of these installations are very innovative and futuristic.

Then there's the performing arts theater. These are really fun to work on, and again, we're unique in this market in that we're one of the few companies in the nation with experience in recording *and* theaters. Part of our experience stems from work with recording studios and multi-channel sound systems, and part of it comes from our background in unique theaters like the Tyrone Guthrie here

WE HAD TO LET OUR CHIEF AUDIO ENGINEER GO.

We felt he was on the verge of something. He wasn't sleeping nights and was often found in corners talking to himself. Our chief engineer expressed a strong desire to go away for a few weeks to clear his thoughts. We let him go.



When he returned, he was grinning from ear to ear and began to explain...

He said he felt the audio industry—specifically pro audio amplifier design—had reached such a level of technology that *everybody* had quality specs. Manufacturers were developing "super-specs" for the sole sake of the specs themselves. A new direction was desperately needed. But where? How?

Our engineer had the answer. Why not develop an amplifier design that not only had incredible specifications, but considered total efficiency as a prime design philosophy?

His brain began to work overtime...creative electronics began to take place. He developed a high-turbulence flow-through ventilation system, direct-mounted power transistors for cooler operation, a unique PowerLimit circuit, error-free DC and sub-audio protection and functional LED power level indication. He also included a horizontal connector panel with balanced 3-pin XLR-type inputs and outputs. The importance being they all pull together into the first real complete "common sense" amplifier design.

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in Minneapolis. This theater is world famous, so it was a privilege and quite a challenge to design a sound system that would complement the theater space and the fine performers who work in it.

Ordon: Theater sound relies not only on excellent reproduction . . . all the framework for tightness and realism . . . but it also relies on effect. If you're going to create a special environment in the theater, you must understand how you can best distribute signal sources to speaker locations. And

that's a thing we have a handle on. We understand what the creative needs are for the producers, directors and actors who will be working with the system we install.

Broadcast audio: How big a part does it play in your business?

Bormann: It's an area we've been concentrating on much more heavily in the past year or so. It's a growing field with us, and it overlaps into other areas, in that it uses large numbers of tape machines, which are a primary line in our stocking system.

Ordon: There's also a return to theater in the broadcast market, too. In the old days of radio, there was a sound effects man, clamoring around for every effect he could possibly come up with. Now, with the kind of equipment we can provide, one man can do what it might have taken two or three men to do in those early days.

Up to now, we've been discussing fixed installations. But what about portable audio gear . . . sound reinforcement systems?

Ordon: In sound reinforcement, we address a highly specialized market. We serve primarily the true professional . . . the person who makes his or her living playing music.

Bormann: These are performers . . . most of them groups, actually, who need a high-quality, sophisticated sound system and who can pay in the neighborhood of $10,000 for it. Often, the group just starting out will buy sound equipment at the music store. When they've become established, started filling a few halls and want to move up to a better quality system, they come to us.

While we're on the subject of music, let's talk about discos. Is this a growing area for you, or do you see the disco market shrinking?

Bormann: The disco market seems to be alive and well. In this market, we have a lot of point-of-purchase sale

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items. For example, a club owner may come in and buy a mixer or a light controller to add to his existing system. Or it can be a total system sale, ranging from $15,000 to $25,000 and up. We have one salesman who concentrates on disco, and this area accounts for a sizable cash volume on sales of single pieces. But we also routinely handle design and installation of complete disco systems.

Where does instrumentation fit in here at AVC Systems?

Ordon: Because of the direction we've taken in the market, we've built our company on heavy service. We dedicate 40 percent of our space to service and engineering. We can spread our experience over the broad-based market that we serve, and buy just about every new piece of test equipment that

comes out. That's allowed us to do things faster and better.

As a result, when we go into the marketplace, people ask our advice. They want to know what they can buy to perform this test or that test on their equipment. When that first began happening, we took a long hard look at the testing equipment on the market, selected what we thought were the best lines, and began selling instrumentation.

It's worked out very well, both for us and for the client. It's to our advantage, and to his, that he progress technically, so that he can handle his own service needs . . . or at least, most of them. That's where the instrumentation package comes in. And it *is* a package, because we provide not only the equipment, but also the training our clients need to use it properly.

If it doesn't have an ANVIL®... you don't have a case.

A lot of musical instrument and amp cases are going around these days trying to pass themselves off as ANVIL® cases. And a few dealers are adding to the confusion by using the word "anvil" to describe cases that may look like ours on the outside but don't even begin to measure up on the inside.

The fact of the matter is simple: If it doesn't have the ANVIL® on it – it doesn't have ANVIL®'s experience, reputation and quality construction in it.

Sturdy, dependable ANVIL® cases are the industry standard – always have been. They're designed around the delicate equipment they carry and are perfectly balanced for easy handling. They're dent-proof, scuff-proof, and slippery-fingered-roadie-proof. And they're available in just about any color you can name.

So if you want your guitar, synthesizer, cello, drums, amps and sound-reinforcement gear to get to the gig in one piece – depend on ANVIL®. We can make a strong case for just about anything.

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Any discussion of instrumentation leads naturally to engineering. Is engineering a corollary to service, or is the relationship just the opposite?

Ordon: The two are very different, but they complement each other. Engineering, as we practice it, relates to the custom design of a system. Service is after the fact . . . once the system's been installed. Engineering is a totally separate area. Designing the complete system, including the hook-up and any special adapters that are necessary to make it work better . . . that's engineering. It's much different from just delivering a stock piece of equipment and plugging it in.

We've got to be concerned not just with electrical engineering, but with mechanical engineering as well. We do everything we can in-house, but we will go to outside sources when the situation dictates.

We try to take a total concept approach, which involves planning, engineering, installation and followup. As in the case of instrumentation, we like to present it all as a package.

What trends do you notice emerging in the professional audio business?

Bormann: Of course, we notice some trends in the business. We try to stay on top of them . . . set them, if we can. A great advantage we have is that our business is so broadly based that uptrends and downtrends tend to

ROAD TOUGH? PROVE IT.

It's six long feet to the floor. What will happen when our great sound hits bottom? How long will it still sound great? We had to find out. So we picked an ATM41 Dynamic and an ATM91 Fixed-Charge Condenser out of stock, tested them, and started in.

Each was dropped seven times on its side from six feet onto the office floor. Nothing much was happening. So we repeated the series, this time dropping each microphone on its nose. Seven times from six feet. Still no problems. They looked good and sounded good, but *we* were getting tired.

So we moved to an unyeilding slate floor. Here it took three more drops on its side from six feet, and three more on its nose from four feet to finally affect the ATM41. A truly remarkable record!

But what about our ATM91 Fixed-Charge Condenser? It should have given up long before a dynamic. But quite the contrary! The ATM91 withstood four side drops onto slate from six feet, three drops right on the nose from four feet, and another six drops on the nose from *six* feet and still tested OK for sound! Granted it looked anything but new, but it *still performed*

Our little test left us arm-weary but convinced that the ATM Series microphones could easily earn their "Road Tough" name in the field. That's the testing which really counts. Try us.

Part of the secret of ATM toughness is this 3-layer windscreen. An outer heavy wire, a finer wire screen just inside, and an inner layer of woven bronze. All soldered to each other and to a solid brass ring. There's nothing else like it on any microphone.

This ATM91 survived 27 drops from as high as 6 feet!

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equalize each other. That makes for a more stable financial environment. For instance, one aspect of the business, discos as an example, might be heating up, while some other portion of the trade is cooling off. As a result, we're not really subject to a lot of major ups and downs.

At this point, we're serving a wide open market that really hasn't been too well served in the past. Then, too, the audio market itself is growing rapidly ... especially in production and recording studios. We've been successful in aiming at that market in the past, and we intend to target in on it even more in the future. Another fast-expanding market is broadcast stations. We expect the activity in this area to be at a very high level over the next couple of years, because AM stereo is soon to become a reality ... and in order to be competitive, most AM stations will want to go with it.

Ordon: We're looking at some new markets, too, and I'm speaking primarily now of industrial accounts. The new market is industry. Industry requires the testing instrumentation, it requires the production studio and the communications equipment and expertise we can provide.

We've already entered this market successfully, but the possibilities for expansion are endless.

Speaking of expansion, how has your staff grown since 1975?

Ordon: From four to a total of nineteen. Six in sales, three clerical, three in service and seven in engineering.

How is your sales staff structured?

Ordon: We have one person assigned to broadcasting facilities, one to portable sound reinforcement systems, and one to discos. Three sales people handle recording studios and performing arts theaters. Those are areas where Jon and I get pretty deeply involved, although the two of us really cover all the bases when it comes to the various client categories.

Do you find it's a problem to attract knowledgeable sales people?

Bormann: Not really. Our reputation and the quality of the work we do seem to attract good people to us. These are people who already have a thorough knowledge of their particular field: broadcast engineering, for instance.

The man who heads up our broadcast sales effort was formerly chief engineer for one of the largest radio stations in the Twin Cities. That kind of thing . . . the extensive background . . . is true of all the members of our sales staff, whether their particular field is recording studios or portable sound reinforcement.

Ordon: That applies to our service and engineering people, too.

What about training for your staff?

Ordon: Since they already have the basic knowledge when they join the company, training is largely a matter of orienting them to the way we do things at AVC. We're probably more thorough in many respects than most audio-related firms. That's a point of pride with us. So we need to make sure our people know that's the way we want to continue conducting our business. We want them to be thorough, too. We also want them to work closely with audio equipment manufacturers, to let the makers know about the needs of clients in the field. Most important, I guess, is that we want to instill confidence in our people, whether they're in sales, service, or engineering. We want them to know that we'll provide whatever backup they need to get the job done right.

Do you use much in the way of ads and promotion?

Bormann: Word of mouth has been very effective for us. But we do use some promotional materials, notably brochures and a rather impressive audio-visual production. We use that slide-tape piece at the various seminars we conduct. Recently, for example, we did a seminar for broadcast engineers. We've also done sessions on speakers, and we're planning a seminar for recording engineers. Of course,

the seminars themselves are not only educational, but an effective promotional tool.

You seem to have an extensive selection of equipment lines.

Ordon: We carry over 70 major lines. We have to, because of the broad range of markets we serve within the industry. A few of our primary lines would be Tascam, BGW, UREI, JBL, 3M-Mincom, Scully. There are many others, of course.

How do you handle the customer's need for financing?

Bormann: We've developed a variety of financing programs including leasing, long-term loans and short-term notes. We like to feel we can arrange a plan that will best suit our customer's need. In addition to financing, we offer other kinds of help. For instance, it's not unusual for us to conduct a feasibility study for a prospective customer, or to help him set up a bookkeeping and scheduling system. Like many of the things we do, this is approached as a "package program."

What's the physical layout of AVC Systems?

Ordon: In total, we have 10,000 square feet. We've allotted 4,000 square feet to engineering and service. The display and office area has 3,000 square feet, and so does the warehouse area. By the way, we recently separated our offices from the display area, and that's a welcome change.

Any other changes?

Bormann: Yes, we recently remodeled the entire facility, adding a new phone system and expanding our office area. There's more work in progress: we're now completing the expansion of our engineering and service area. We feel that, with more space and equipment, these two departments will be able to give us the additional backup we need as our business continues to grow.

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- Power amp input

THE BASS

- 200 Watts RMS @ 1% THD, 4 Ohms
- Two channels with pre and post gain on each channel
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- AUTOMIX function selects either or both channels with remote footswitch
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- Channel LED indicators
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THE STANDARD

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- LED channel indicators
- Master reverb
- Preamp and line outputs
- Power amp input

THE CENTURION

- 130 Watts RMS @ 1% THD, 4 Ohms
- Two channels with pre and post gain on each channel
- Individual channel equalization
- Parametric equalization
- AUTOMIX function selects either or both channels with remote footswitch
- LED channel indicators
- "DDT" compression circuit with LED indicator and in/out switch
- Preamp and line outputs
- Power amp input

These features give you an indication of the Mark III Series' versatility and performance, but specs and features mean little unless you can plug-in and experience for yourself. See your Peavey dealer for a demonstration, he'll show you why unbelievable values are still a reality with Peavey.

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INDUSTRY UPDATE

Bob Angona has been named Assistant to the President for the Audio Products Division of Osawa and Co. (U.S.A.). According to the company, the current "executive management staff will remain intact."

The American Music Conference has named **Gene C. Wenner** President. Wenner is an arts administrator and music educator, and has been Arts Education Coordinator for the U.S. Office of Education.

The Village Recorder has established a research and development department to develop and market products its staff engineers devise. Two devices already planned for public sale is a direct box and a frequency counter.

KEF Electronics, Ltd. has expanded its U.S. marketing with the addition of a 10,000-square-foot west coast warehouse in Inglewood, California to mesh with the current facility on the east coast.

Norm Schneider, JBL professional products representative for the southeastern United States, has been awarded the JBL "Rock" for outstanding sales performance and dealer service during the past year. Schneider's company is Technical Systems Reps, based in Chamblee, Georgia.

Jesse Maxenchs has been appointed Marketing Manager for Broadcast Products of Orban Associates. Pro audio marketing activities will continue to be under the direction of **John Delantoni**, general manager, and **Michael Bernard**, pro audio sales manager.

Gabriel Nakash has been appointed to represent Fender/Rogers/Rhodes in southern Illinois and Indiana. Nakash has previously served as district and regional sales manager for Norlin Music.

Cowan Associates has been named to represent the Sonus Line in New England. The principals are **Jeffrey** and **Douglas Cowan.**

Herbert Paige has assumed responsibilities as Acting President for James B. Lansing Sound, Inc. and will continue to serve as President, U.S. High Fidelity Group, for Harman International Industries. **Sterling B. Sander** has resigned as President of JBL. **Bernard Girod** has been appointed to the newly-created position of Executive Vice President for Finance of the company. **Steve Rand** has been appointed Vice President for Domestic Sales. Rand was previously with Marantz/Superscope.

Ken Sacks, previously with TEAC, has been named Executive Director of CAMEO (Creative Audio and Music Electronics Organization), becoming the first full-time executive director of the organization. **David Schulman** will continue to serve as legal counsel and advisor to the board of directors. **Larry Blakely**, vice president of CAMEO, will assume the position of president previously held by Sacks, until the next election this month. The new location of CAMEO's national headquarters is 5430 Los Angeles Avenue, Simi Valley, California 93063. CAMEO is presenting a series of twelve seminars at the NAMM June Expo this month.

Charles Harper has been appointed Vice President, Consumer Sales for Altec Lansing International. Harper was previously with Harman/Kardon and KLH.

Soloca, Inc. has acquired the Coupland Synthesizer Division of International Micor, Inc. **Rick Coupland**, inventor of the Coupland digital synthesizer has left Micor in order to assume duties as President of Soloca. The company plans to have the synthesizer in production by the third quarter of 1979.

Horst Ankermann has been appointed President of Sennheiser Electronic Corporation. Ankermann first joined Sennheiser in 1954 as a research and development engineer, and later served as Vice President-Engineering.

The third annual NAMM Management Institute will be held August 1 through 4 at the Harrison Conference Center in Lake Bluff, Illinois. The Institute's theme will be "The Decade Ahead: What's Going to Happen to the Retail Music Business in the 1980's." Enrollment is limited to 80 individuals, all of whom must be retail store owners.

Furman Sound has appointed **Larry Earl Collins** Marketing Director. Collins was previously Vice President of Rothchild Musical Instruments. Among several duties, Collins will be assisting in setting up a new co-op advertising program.

James Coleman Guthrie, Jr. has joined Sony Industries' Professional Audio Division as Technical Field Sales Manager. Guthrie is responsible for the professional applications and field sales of Sony's wireless microphone system as well as professional and consumer microphones and stereo headphones. He was previously manager of the professional products division of the Paul Seaman Company.

Nasty Cordless Inc. has moved to a new 20,000-square-foot building in Oakland, California. The company will be handling its own marketing through in-house personnel, with Morley Pedal Company no longer handling distribution. **Peter Kalmen** has been appointed Sales Director and **Byron Stone** Marketing Director.

The Board of Directors of the Institute of High Fidelity has approved a resolution to investigate an affiliation with the Consumer Electronics Group of the EIA.

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Burns Audiotronics is the new and exclusive representative and distributor in the United States of Beyer Dynamic headphones, microphones and all accessories. And we handle Beyer products exclusively.

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Here are our plans:

- The Beyer line will be made available to a limited number of highly qualified professional sound distributors and music stores who have demonstrated their ability to effectively serve the users of Beyer products.
- A substantial advertising and publicity campaign aimed at the professional audio and music markets will be launched June 1st.
- To back-up the ads, we will provide our dealers with a substantial in-store display and literature program.
- We will maintain a large and complete inventory for fast delivery of your orders.
- We are setting up a major service facility with factory-trained technicians and the latest test equipment to assure rapid turnaround of your repair orders.
- We're setting up a stringent quality control program so you'll rarely need the service department.

If there's anything else we can do for you, or if you need the name of your local representative, give us a call.

Beyer Dynamic

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