SERVING THE CREATIVE AUDIO AND MUSIC ELECTRONICS INDUSTRY

SOUND ARTS

MERCHANDISING JOURNAL

MARCH 1978
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FRANK

Guitar Synthesizer Part I— A New Wave

Equalizers— What They Are & Who Needs Them

Western NAMM Sound/Image Product Roundup

Consumer Profile—The Sound Mixer/Musician

Finance America— A National Lending Program for Musicians

At Last—the Ultimate Musical Instrument

What is this incredible new guitar synthesizer that is suddenly finding its way into the hands of such superstars as Jeff Baxter, James Vincent, Steve Miller, and Jimmy Page of the famed Led Zeppelin? It is Roland's new GR-500 Guitar Synthesizer . . . both paraphonic and polyphonic. Polyphonic because full chords can be synthesized. Paraphonic because all five sections may be played at once. An auxiliary, external synthesizer can be controlled by the guitar. Chords may be sustained for any duration. You can bend synthesized tones throughout the scale. Any of the five sections can be channeled to three separate amps. The result of all this is that for the first time in history the guitarist commands even greater control of his music than the keyboardist!

The Roland Guitar Synthesizer consists of two units. The guitar looks, feels, plays and functions much like any solid, high-quality guitar with two pickups (one humbucking).

The guitar plugs into a synthesizer unit which can be placed on a stand or the top of an amp, near the guitarist. The connecting cable has 24 conductors, making it possible for the guitarist to control, from the guitar, many functions normally found on a synthesizer.

The second pickup on the guitar is a special **divided pickup**. It is actually six small pickups—one for each string—each sending its own signal to the synthesizer section. This enables you to select different sounds for each string.

Once the signals reach the synthesizer control unit, you can begin to synthesize sounds. Generally you work from left to right on the control panel when making adjustments.

The first section you come to on the panel is the **Guitar Section**. This controls normal guitar sound through equalization. Because the control unit has **three outputs**, it is possible to select which output any particular sound will be channeled to. Imagine you are using three Roland JC amplifiers . . . one on the left of the stage, one in the center, and one on the right. You would probably want the regular guitar sound to come from the center amp. You therefore set the channel selector of the Guitar Section to the output which feeds the center amp. Further adjustment of the sound can be made at the amp.

The next section on the control unit is the **Polyensemble Section**. It provides sounds similar to woodwinds, brass, soft reeds, and such solo effects as a harpsichord or human voice. The Polyensemble Section operates much like a conventional, polyphonic synthesizer. All the guitar strings will cause synthesized sound to be produced. Plus, you can bend strings as much as you like and the synthesized sounds will follow beautifully.

The **Bass Section** is next. Its controls, which include envelope adjustments, enable you to get sounds that closely resemble string bass, electric bass, strings or tuba.

The **Solo Melody Section** gives you solo sounds that can be shaped with square and sawtooth waves, VCF's, VCA's and envelope controls, just like in conventional synthesizers. Here again synthesized sounds follow your note-bending techniques with no problem.

The **External Synthesizer Section** lets you control other synthesizers. By connecting a larger synthesizer, you can create sounds that are truly incredible. Because Roland believes in compatibility between synthesizers, the possibilities are endless.

You really have control of your sound when you play a GR-500. The control panel is simple to operate, and all of the sounds are available at the flick of a switch on the guitar. For example, imagine a four-piece group—three guitars and drums. At the beginning, the lead guitarist wants a really raunchy sound. He flicks a switch on the GR-500, activating the Guitar Section. An adjacent, rotary knob changes the tone. This control is far more powerful than any tone controls built into a guitar before. It controls an equalizer on the control unit. When the guitarist comes to a solo, he wants a heavier sound. He flicks on the Bass Section. In addition to his raunchy lead, he's delivering a thundering bass in unison. The bass player plays a counterpoint and it really sounds very heavy. The lead player is using three independent amps with the GR-500 . . . the lead coming from center-stage. But when he activates the Bass Section, the bass sound comes from the far right . . . another instrument "seems to appear from nowhere."

For the next part of the number, the thundering bass on the right disappears and a dreamlike chorus of sound from the Polyensemble Section comes from left-stage . . . blending beautifully with guitar sound from center-stage. All of these changes are made from the guitar.

There is one very important feature of the GR-500 that deserves to be emphasized. The GR-500 gives you **infinite sustain**. Some people spend the price of a GR-500 just trying to get that single effect!

At last . . . an instrument that requires no "inspiration from the gods." You are in complete control. You are the creator . . . the GR-500 definitely will set your creativity free. There is no other instrument like it.

RolandCorp US
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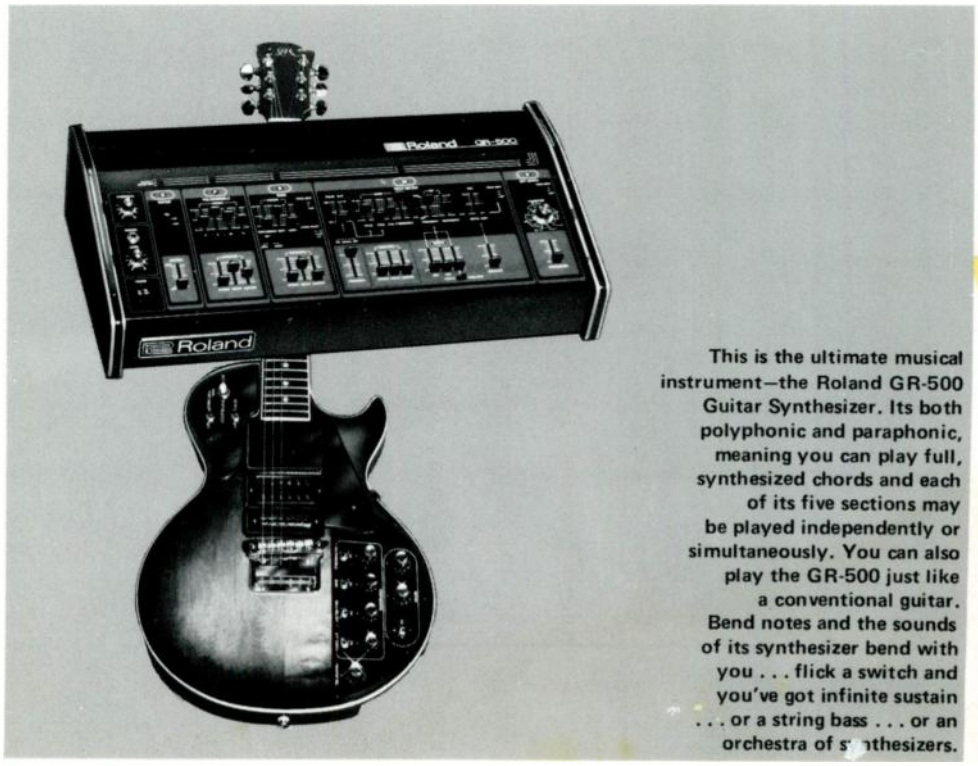


This is the ultimate musical instrument—the Roland GR-500 Guitar Synthesizer. Its both polyphonic and paraphonic, meaning you can play full, synthesized chords and each of its five sections may be played independently or simultaneously. You can also play the GR-500 just like a conventional guitar. Bend notes and the sounds of its synthesizer bend with you . . . flick a switch and you've got infinite sustain . . . or a string bass . . . or an orchestra of synthesizers.

10-096

UNI-SYNC INC

LIVE MUSIC

A JOURNAL PUBLISHED BY UNI-SYNC, INC., THE PROFESSIONAL PRODUCTS DIVISION OF BSR.

Deciding on a Mixer

The variety of mixing systems on the market makes it difficult to judge which is best for you. Features blend together forming a mirage of switches, controls and pots; all looking alike. Therefore, we've come up with the *sound artist's guide to mixing buys.* It will give you independence when it comes to buying a mixer. The following is an abbreviated version of *the guide,* which you can send for free of charge. We hope it serves you well and would appreciate hearing from you.

The Sound Artist's Guide to Mixing Buys

What is a sound artist? We've come to realize the existence of a new category of performer. This is the person who creates, through the use of various tools a sound that appeals to the audience: therefore the sound artist. How well the information communicates is left to the talented ears of this individual, and the manner in which these tools are utilized. The sound artist, with today's technology, has become an instrumental part of the performance.

Mixer is the Basic Tool

The basic tool of the sound artist is the mixing board; with it he can create a myriad of sounds from his fingertips. It not only has to sound good, be reliable and versatile, as well as having excellent specifications, but has to have just the right touch. For the person mixing relies upon his hands as much as he does his ears.

DESIGNED ON THE ROAD

It is for this reason that Uni-Sync, in designing the Trouper Series is extremely innovative in the choice of front panel controls. Michael V. Ragsdale, president of Uni-Sync as well as chief designer of the Trouper Series, has built, serviced and most of all, operated sound systems prior to forming Uni-Sync. The Trouper Series was designed on the road, the true proving ground of sound reinforcement equipment, from a practical viewpoint. For example, the use of slide faders as opposed to rotary pots was an extremely important decision based on ease of operation as well as visual and tactile indication of position. Rotary pots are hard to read under the dimly lit conditions of sound reinforcement, whereas a slide pot gives instantaneous recognition.

Live or Recording

One of the most important decisions to make is where your mixer will see the most use. Is it for a "Home Studio" or to be run live. Live boards have different gain and level structures. In a studio, you are dealing with a controlled environment, but live sound is just that; live and wide open. To handle that kind of sound you need to have a mixer that has been specifically designed for that purpose. This is the Trouper Series, designed and built for the road or permanent installation for mixing live sound...it is a live music mixing system.

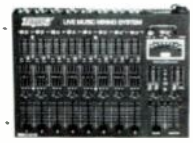
Inputs You Can't Grow Out Of

Next on our list of important decisions, is to determine how many and what type of inputs you need. Mixers come in various configurations, application determines the need. The Trouper I, for example, has on each channel: low Z balanced and high Z inputs, and an in/out jack. This allows for maximum flexibility.

Now, how many inputs do you need? Most mixers come in fixed quantities; for example, six, twelve, or sixteen. Once you grow out of it, you have to buy a new board. Not so with the Trouper Series. The basic mixer is an eight input/output control module that is expandable through the addition of a ten input expander module, that simply plugs in. You never grow out of a Trouper.

Build Yourself a Custom Board

If you had the freedom or ability to build a mixing board perfectly suited for your needs, what would you put in it, how big would it be? The Trouper Series gives you this freedom at an affordable price. Our mixers are big boards in little packages, giving you the opportunity to custom design a system that is tailored for your specific needs. You build what you want, not what someone else thinks you need.



Dollars Per Input

An excellent way of determining the value of the mixing board being considered, is to divide the cost of the board by its total number of inputs. This gives you an objective analysis of the mixer, and by comparing and contrasting features per dollars, you can arrive at a decision. For instance, the mixer at $100 per input may have far greater features than the one at $85, and would be a more valuable purchase.

Mono or Stereo

The Mono/Stereo issue is one of the most controversial at hand today in the retail sales of mixers. Most installations and gigs are best handled in Mono. But many groups today, want the added flexibility of a Stereo board. We are presently introducing the Trouper I Stereo, which is probably the most flexible and versatile mixer on the market for its price. At $898 (suggested pro net), each channel features a house pan pot along with an echo pan pot enabling you to pan the echo to or away from the house signal. A little imagination can create some very interesting effects.

The choice for Mono or Stereo is based on budget and application. Practically speaking, Mono will satisfy most of your needs.

Send For Your Free Guide

That's the abbreviated version. If you'd like the complete guide fill out the attached coupon and send it in to us right away. You may want to get some of the other Trouper Series goodies like T-shirts or director's chairs for a comfortable place to mix from. We're looking forward to hearing from you.

Thanks,
Larry Jaffe
Marketing Manager

Dear Larry,

I'm interested in what you have to say. Why don't you send me a copy of SOUND ARTIST'S GUIDE TO MIXING BUYS. While you're at it send the goodies I've checked off.

- ☐ Uni-Sync Trouper Series T-Shirts @ $4.99
- ☐ Trouper Series Director's Chair in natural wood (great for mixing in) $32.00
- ☐ Trouper Series Catalog (no charge)
- ☐ Name and address of my local dealer

Thanks, I'm enclosing a check for $ _____

California residents add 6% sales tax

Mail to:

UNI-SYNC

742 Hampshire Road, Westlake Village, Ca. 91361

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CIRCLE 85 ON READER SERVICE CARD

You can pay for someone else's studio or you can invest in your own.

Our new Studio 8000 gives you that choice. And you won't have to sell your soul to get it.*

The eight tracks give you room to spread out your music, and your own studio gives you the option of turning on the equipment whenever you turn on to a good idea... 24 hours a day.

Here's what you get. The TASCAM Series 80-8: half-inch, 15 ips. One speed, one format saves you money but gives you a final product: professional master tapes, faster and easier than any recorder/reproducer you ever sat behind. Add the DX-8 for up to 30dB of noise reduction.

The Model 5A Mixing Console gives you 8-in and 4-out, and has been studio proven in both mobile and fixed installations. In short, it's uncomplicated and tough. Add the 5EX for 12 inputs and even greater flexibility. The Model-1 (8-in, 2-out) gives you those necessary sub-mixes, without affecting your primary mix.

And for absolute quality stereo mastering, plug in the 25-2. DBX is built in, and so are speed, simplicity and accurate final editing capability.

So why go on paying for time in a studio that someone else owns? Especially when the total dollars involved wouldn't buy much more than a new car at today's prices.

See the Studio 8000 at your authorized TASCAM dealer.

STUDIO 8000

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A new generation of recording instruments for a new generation of recording artists.

*$10,460 suggested retail price as shown. Rolling consoles shown are optional. Prices subject to dealer preparation charges where applicable.

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SERVING THE CREATIVE AUDIO AND MUSIC ELECTRONICS INDUSTRY

MARCH 1978

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VOL. 1 No. 2

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THE STAPLES

COMING NEXT ISSUE!

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Cover illustration by Sheryl Stern

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A LETTER FROM THE PUBLISHER

Last month most of us witnessed the tremendous increased interest in 'active and creative' audio equipment and accessories at the Winter CES show. For the first time major companies like Sansui were botiquing their *hi-end* professional consumer products. Likewise, newer manufacturers like A.B. Systems, Unisync and Spectro Acoustics presented their sophisticated product lines of power amps, signal processing devices and recording equipment. Something new and creative seemed to be happening in the audio industry.

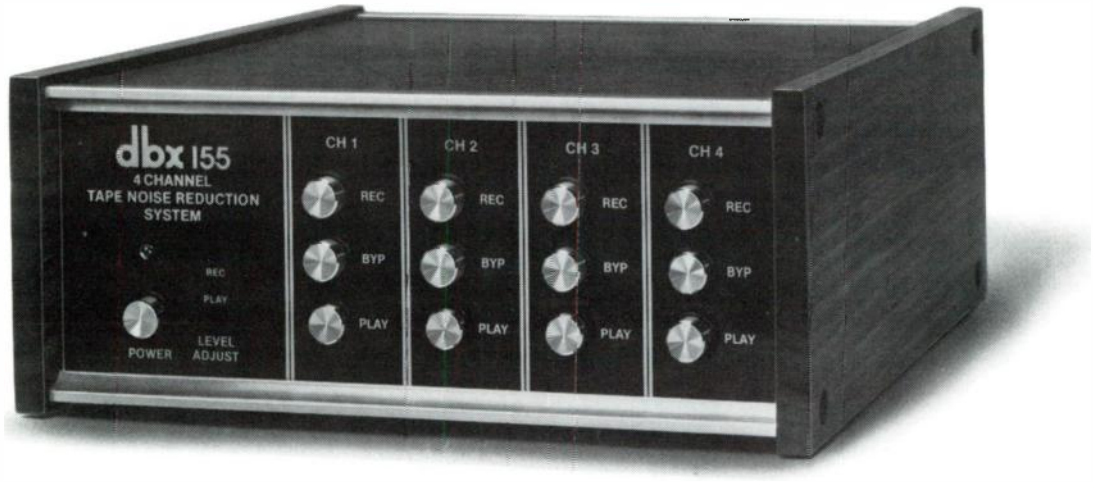
Having just returned from the Western NAMM Show (National Association of Music Merchants), I must say that the evidence of a new industry emerging was chiseled in stone. At NAMM—the dbx's found the Peavey's the ARP synthesizers found the Teac's and a new industry of electronics manufacturers was solidified—an industry that's not hi fi (low-end passive equipment i.e. tuners, record players etc.), but *creative audio* equipment (i.e. mixers, tape machines, equalizers, etc.) for the hi-end active user. An industry that is not primarily—'drums and bugles'—but *synthesized* drums, keyboards, and guitars; sound reinforcement mixers, power amps, electronic signal processing devices, effects pedals etc. An industry association has been formed for this new market. It's called CAMEO—the Creative Audio and Music Electronics Organization (see Billboard, Dec. 17, 1977 cover story).

As the Publisher of MODERN RECORDING I have watched (and I would like to hope I have helped) a 'new breed' consumer—the professional consumer. The musician today considers his sound reinforcement electronics, his home recording electronics and his musical instrument electronics a necessary extension of his talents. The audio enthusiast, today, performs *with* the equipment as his means of creative expression, therefore the sound mixer, the recording enthusiast, and the engineer/technician emerges. Many of you have heard me talk (and in some cases argue) about this new breed of consumer for quite awhile now, since my days as founder and developer of RIA (the Recording Institute of America), and then and now with MODERN RECORDING. MODERN RECORDING has proven beyond a doubt, the existence of this *professional market.* Our growth in circulation since July is unlike any other special interest publication in the field.

Now I need your help. We at MODERN RECORDING want this industry to grow—to grow as an industry concerned not only with sales but with a sharp eye to the needs of the consumer. Remember, as a creative, professional consumer, his needs and wants are much more serious than the 'hi-fi buff' or the 'music player'. He also is considerably more knowledgeable and sophisticated as a result of his serious interest or vocational aspirations.

The person selling your equipment to this consumer needs help in communicating. We think SOUND ARTS MERCHANDISING JOURNAL offers him that help in a unique and refreshing way. It is unique from other trade books in both the

HUSH UP YOUR 4-TRACK.



Introducing dbx professional four-track tape noise reduction for under $500.

You've just settled on a TEAC, Tascam, Otari or Dokorder four-track tape deck for that studio you always wanted to have. You've chosen the mikes, the carpenter is almost finished (or maybe you even built it yourself). Your console's ready to be wired into place.

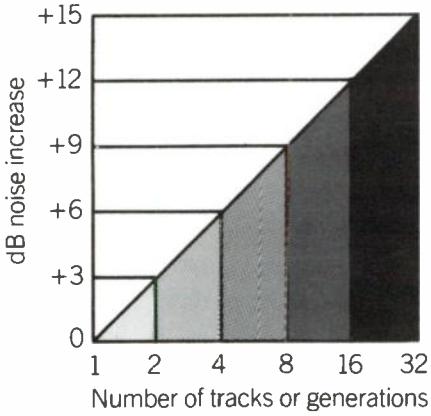
But...haven't you forgotten just one important item?

Noise reduction.

Because every time you (or your group) want to bounce a track, you're adding at least three dB of tape noise (see chart). So the great artistic result you plan to end up with, might end up sounding like a rainstorm.

Fear not. Help is at hand. It's the new dbx 155 four-channel tape noise reduction system. You can add it for far less money than you ever imagined possible. Here, on one compact chassis, is a complete dbx noise reduction system. But the best part is, it will give your tape deck an extra 10 dB of headroom, and reduce tape noise by 30 dB. That means no audible noise whatsoever will be added to your tracks. And, because dbx tape noise reduction operates by linear compression/expansion, you won't have to get involved with tedious level calibration, either.

Additive Noise Chart



All you need do is press the playback buttons to hear noise-free, full dynamic range reproduction of your music.

The new dbx 155 also has user-changeable modular circuit boards, so in the unlikely event that one processor fails, the other channels remain operational. You can even keep a spare on hand.

Visit your dbx professional dealer now, for a demonstration of our new 155 tape noise reduction system. Discover how you can put an end to tape hiss, without putting an end to your bankroll.

dbx

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

CIRCLE 88 ON READER SERVICE CARD

audio and musical instrument field because we believe the dealer, and sales person himself is unique. He has to be because the product is and the end user is. The one cmmon bond is that the consumer wants to buy and the salesperson wants to sell.

In order for SOUND ARTS MERCHANDISING JOURNAL to be effective it must be a 'shared' experience. Both the manufacturer and the dealer/salesperson must feel that this book is exclusively for them. We want it that way.

We feel we have delivered, in the premier issue, more helpful educational merchandising and sales information on creative audio and music electronics than other trade publications have done in a year's worth of issues. That's only because we're specialized. We need more. We need your input. We want to know what you liked and what you would like to see. Help us help you! Reach the people who reach the people.

Thank you.

Cordially,

Vincent P. Testa
Publisher

FORUM

We thought we would publish the first wave of communication we received upon the announcement of the 'birth' of Sound Arts. *In future issues "Forum" will be an open communication line for you—and we want your input. Any questions, problems, opinions—whatever! Send it along for consideration to: Sound Arts, c/o Forum, 14 Vanderventer Ave., Port Washington, New York 11050.*

* * * *

We wish you the greatest success in this new venture. Best Regards.
Frank Karoly
Shure

The growth of this market is directly proportional to *knowledge* and a book of this type will definately be a giant step in this direction. *Sound Arts Merchandising Journal*, I applaud you!
Larry Blakely
dbx

Good luck with your new publication!
Dane Butcher
Symetrix Pro Audio Products

Best of luck in your new endeavor and all best wishes for 1978. I look forward to seeing the premiere issue of *Sound Arts* at the West Coast NAMM Exhibit.
Bob Hoffman
ARP

Steve Caraway—Congratulations! I hope you like New York. I think you'll like working with Vinny. Best of Luck!
Curt Pickelle
Altec

Thank you for your notification on the birth of your new publication, *Sound Arts Merchandising Journal.* Thank you for your interest in our products and the best of luck to your new magazine.
Ted Nakamichi
Nakamichi

More than any other, we need this!
Larry Jaffe
Uni-Sync

Analog/Digital Associates wishes you the best of luck with *Sound Arts Merchandising Journal.* We hope it is a big success for us all.
Gil Pumar
Analog/Digital Associates

The customers are reading magazines and the dealers are not. The first job the manufacturer has to do is to educate the dealer. The word is *help!*
Hartley Peavey
Peavey

In the complex field of sound, what is needed isn't just a textbook, but a living journal that is constantly focusing on problem areas and new developments. This is what Vinny Testa promises, and judging from his *Modern Recording*, he's the man who can do it.
Paul Ash
Sam Ash Music

Good Luck in your new venture!
Buck Munger
Norlin

Congratulations of the birth of *Sound Arts!*
Sally S. Darby
Burwen Research/KLH Corp.

Thanks for your note and the initial issue of *Sound Arts*. I'm extremely impressed with Issue No. 1. It really doesn't look like a first effort. We are proud to be represented in it. Importantly, we believe your underlying concept has great validity. Your new magazine gives us an ideal forum for presenting this essential information.

Keep 'em coming; we're with you all the way!

Jon R. Kelly
Audio-Technica

TEAR HERE

TERMS:

A CONTINUING INDUSTRY GLOSSARY

RECORDING

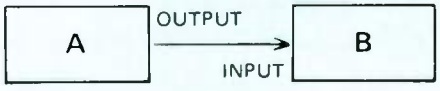
By Michael Tapes

Last month we started discussing how one knows whether or not one piece of equipment will work in conjunction with another. We defined how to determine proper matching of impedance. But does impedance matching insure compatability? No!

We must also know the operating levels at which the devices work, both input levels and output levels. The *input level* can be defined by 3 specifications: input sensitivity; nominal input; and maximum input. The *input sensitivity* tells you the lowest signal level a device will accept and still produce its rated output. The *nominal input level* is the average or normal signal level at which the device "likes" to work. The *maximum input level* is the highest level that the device will accept before it *clips* or badly distorts.

The output level is defined by 2 specifications: nominal output and maximum output. The *nominal output* is the average or normal level produced by the device when it is operating properly. The *maximum output level* is the highest level the device can produce without distorting.

How does this relate to interfacing some gear? Let's assume the situation in the following diagram; we want to



feed device "B" from device "A". Device "B" has a *nominal input level* of +4dBm and an *input sensitivity* of −6dBm. Device "A" has a *nominal output* of −10dBm with a *maximum output* of 0 dBm. Can these devices work together? The numbers seem to indicate that they can since "A" can produce the −6dBm necessary to "drive" "B" properly. However, when "A" is operating at −6dBm, there is only 6dB to spare. This is known as the

ELECTRONIC MUSICAL INSTRUMENTS & ACCESSORIES

By Fred Ridder

Magnetic Pickup: The signal-generating device used in electric guitars and bass guitars. The steel strings of the instrument vibrate within a magnetic field while a coil with many turns of very fine wire senses the variations in the magnetic field caused by the vibrating string and generates a voltage proportional to the vibrations.

Humbucking Pickup: A type of magnetic pickup which splits the pickup coil into two halves wound in opposite directions. This feature means that external magnetic fields such as the hum field around an AC cable or power transformer will cancel out in the two oppositely-wound coils rather than being picked up and fed to the amplifier as in a conventional, single-coil pickup design.

Pole Pieces: Small metal pieces which protrude from the top of a magnetic pickup and serve to focus the magnetic field on the strings. In some pickups the pole pieces may be individually adjustable by screw threads to produce a balanced output level from each string.

Contact Pickup: A type of musical instrument pickup which is designed to be attached to the body of an acoustic instrument to pick up the vibrations of the instrument itself. Contact pickups are generally either dynamic or piezoelectric designs.

Dynamic Pickup: A type of contact pickup which uses a permanent magnet and a coil of wire which move in relation to each other to generate an output voltage in the same manner as a dynamic microphone.

Piezoelectric Pickup: A type of contact pickup which uses a piece of piezoelectric material to produce an output voltage proportional to the vibration stress applied to the pickup. The piezoelectric effect (generating a voltage in response to a physical stress) is a special characteristic of cer-

SOUND REINFORCEMENT

By Bob Heil

Attenuate: To weaken a given electrical signal by reducing its level (usually by means of a volume or loundness control).

Audio Frequency: Any frequency corresponding to a normally audible sound wave, usually 20 to 20,000 cycles per second.

Audio Oscillator: A nonrotating device for producing an audio-frequency sinusoidal electric wave, whose frequency is determined by the characteristics of the device.

Baffle: The panel upon which a loudspeaker is mounted, most commonly the front panel of an enclosure.

Bass Boost: An accentuation of the lower audio frequencies in the amplitude-frequency response of a system or transducer.

Bias: An alternating current of high (beyond audibility) frequency, linearly added to the signal.

Crosstalk: Undesired energy appearing in one signal as a result of coupling from other signal paths.

Distortion: Any difference between the original sound and that reproduced sound. Distortion takes on many forms, and although it can never be completely eliminated, it can be reduced to a minimum in a good recording or reproduction system.

Distortion, Intermodulation: Nonlinear distortion of a system or transducer characterized by the appearance of the output of frequencies equal to the sum and differences of integral multiples of the two or more component frequencies present in the input wave.

Distortion, Phase Delay: Of a system or transducer, the difference between the phase delay at one frequency and the phase delay at a reference frequency.

Dynamic Range: The difference, in decibels, between the overload level and the minimum acceptable signal level in a system or transducer.

TERMS: (CONTINUED)

A CONTINUING INDUSTRY GLOSSARY

RECORDING

headroom. Although the 2 pieces will work together, device "A" would not have adequate *headroom* to allow all the musical peaks to pass through unclipped. The amount of headroom needed in a given situation is dependant on what is being recorded and how its *dynamic range* must be preserved in the final product. If the final product will be used for television background music in a soap opera, we can compress the music to 'fit' into the final format, which in this case is dynamically non-critical. So when comparing impedances, levels and the like in choosing gear combinations, one must throw in some experience to make the final determination of compatibility. The point that I am trying to make is, that one must know what the numbers represent when applying them. They may seem to work mathematically and under sine wave testing, but the real world of recording is full of *music*, which is far more complex than numbers would lead you to believe.

All of the terms defined regarding levels assume that music is "steady state" and it is not! Music has dynamics. *Dynamic range* is the difference between the loudest and lowest parts of the music. This dynamic range must be considered at all times during the recording process. The *dynamic range* of music is far beyond what the best electronic gear can handle. That is why compressors, companders, noise reduction systems, manual gain riding and many other techniques and devices exist. The recording/reproduction chain has not yet reached the point where full *dynamic range* can be realized, and so the recording process still remains a compromise.

This will be Michael Tapes' last Terms column. Larry Blakely from DBX will be writing this Recording column beginning with the April 1978 issue.

ELECTRONIC MUSICAL INSTRUMENTS & ACCESSORIES

tain crystals and special ceramic materials; the original types were crystal while the more recent devices are primarily ceramic types.

Transducer: This is the general term which applies to any device which changes a signal from one form to another. The most basic examples of transducers are the microphone, which changes sound waves in air into electrical impulses, and the loudspeaker, which changes electrical power into sound waves. In common usage, the term "transducer" is used to refer to contact pickups in general or piezoelectric contact pickups in specific.

Electret Pickup: A type of contact pickup which is actually more like a contact microphone as it picks up airborne sound vibrations as well as the physically transmitted vibrations of an instrument's body. These devices basically consist of an electret (permanently polarized) condensor microphone element which attaches to the instrument.

Phase Shifter: An effects device which generates a moving, swirling type of effect by electronic means. Basically a phase shifter mixes a phase delayed signal with the input signal to produce a pattern of cancellations and reinforcements across the audio frequency spectrum, and these cancellations can then be swept up and down in frequency to produce the swirling effect by varying the amount of phase delay periodically. Phase shifters, or phasers, use a series of phase shift stages to generate the phase delay; each pair of phase shift stages will produce a single cancellation point.

Parametric Wah-Wah Pedal: Generally refers to a type of wah-wah pedal in which other characteristics besides the frequency can be controlled. Most often there will be a choice of filter types plus a bandwidth control.

SOUND REINFORCEMENT

Echo: A wave which has been reflected or otherwise returned with sufficient magnitude and delay to be perceived in some manner as a wave distinct from that directly transmitted.

Equalizer: A device designed to compensate for an undesired amplitude-frequency or phase-frequency characteristic, or both, of a system or transducer.

Fidelity: A measure of the degree of exactness with which sound is duplicated or reproduced.

Filter, Wave: A transducer for separating waves on the basis of their frequency.

Flat Response: A related but more definitive term than simply *Frequency Response.* Implies a degree of excellence, but in order to be meaningful, must be further defined in terms of dB variation from absolute flatness over a specified frequency range (± 3B from 30 to 16,000 Hz). An indication of a sound system's ability to reproduce all audible frequencies supplied to it. Maintaining the original balance between low, middle and high frequencies.

Frequency Range: The range between the highest and lowest pitched sounds which a tape recorder or other sound system component can reproduce at a usable output or volume level.

Ground Loop: The path in a ground circuit between two or more points intended to be at the same voltage (zero), but which are not because of ground resistance. It can result in hum or other undesirable disturbances in circuits where signal strength is low.

Insertion Loss: The ratio of the power delivered to that part of a transmission system which will follow a transducer, to the power delivered to that same part of the system after the insertion of the transducer.

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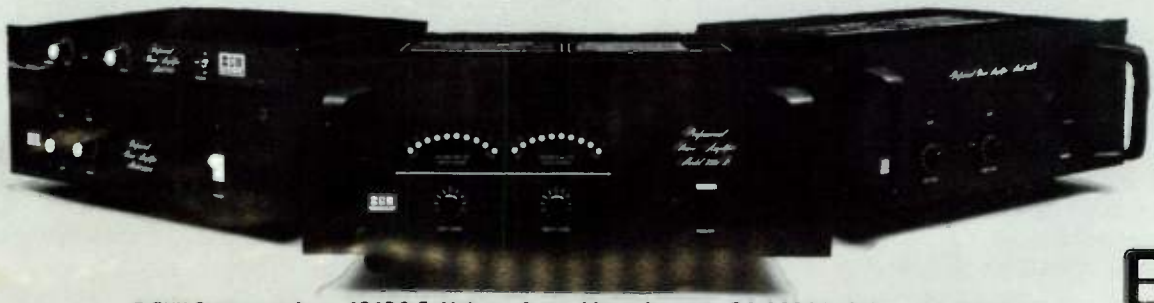

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What is "Sound Pressure Level?"

Sound pressure level is an expression of a ratio given in decibels to a reference pressure. For example, in linear measure, 36 inches is 36 times greater than one inch, the base reference. The reference level for sound pressure level is 20 micronewtons per square meter. This is called zero dB SPL (sound pressure level). Because of the magnitude of the ratios encountered in sound pressure levels, these ratios are converted to logarithms with multipliers. To give you an idea of the magnitudes involved; one newton per square meter (or 50,000 times greater than the base level) will give you 94 dB SPL. At the same time, 101,300 newtons per square meter will give you 194 dB SPL. This last number represents the complete modulation of atmospheric pressure and would be the largest possible sinusoid you could get because you can't get less than a total vacuum. Sound pressure levels are typically measured wwith a precision grade, recently calibrated, sound level meter. To be meaningful to the musician, sound pressure should include the power applied to the speaker, typically one watt on a stated impedance (a nominal impedance is usually given), the type of the signal put on the transducer; either sine wave, for signal frequency measurements; or pinknose for broad band measurements; and lastly a given distance (these days that distance is one meter). One meter readings will always be 1.7 dB SPL greater than four foot readings (if you want to convert old measurements to new measurements). Here is a note of caution, sine wave measurements will always be higher than pinknose measurements. An example of this is the EIA standard of 1 watt at 4 feet at 1 kHz, is of little value when evaluating loudspeaker efficiency. Because pure tone ratings do not give a full picture of the total frequency response of the device. Accurate and complete sound pressure level references are a value to the musician to determine efficiency of playback and instrument speakers. By using power multipliers a musician can determine maximum sound pressure level output of speaker systems and when to use ear protection.

Tom Walter
Community Light and Sound
Philadelphia, PA

What is the difference between phasing and flanging?

Aphaser is an effects component designed to produce a phase shift response created by a comb-filter effect with frequency related notches, which is mixed back with the original signal through a series of all-pass filters exhibiting a phased response that is variable in frequency. When this signal is mixed back in with the input signal, cancellations and reinforcements of the original signal occur. The resulting comb-filter effect spreads over a wide range of frequencies as these notches are moved up and down the audio range—the phasing effect is obtained. The subjective effect of phasing is best at mid and low frequency ranges.

A flanger is an effects component designed to provide repeatable reel flanging effects, caused by mixing a dry and time delayed signal to create a comb-filter response with harmonically related components. The effect of flanging is created by mixing a variable time delay signal back with the original dry signal. The resulting comb-filter response is characterized by the precise mathematical relationships created by the time delay. The subjective effect is that of conventional reel flanging without the necessity of manpower and multiple tape decks. The comb-filter response of flanging causes random program material (i.e., drums, cymbals, and other percussion) to take on musical tonality. Unlike phasing, flanging is subjectively more noticeable at mid and high frequencies due to the time delayed response.

Ron Wilkerson
MXR Innovations
Rochester, NY

What is the ratio of impedance to power in regard to the output of an amplifier?

As an example, we've checked out some Yamaha power amps on our bench that were rated at 225 watts at 8 ohms. If you loaded that same amp down to 4 ohms, which is twice the load on the amp, that amplifier will put out 350 watts. That's almost ⅔'s more. We've also checked out some Cerwin-Vega amps at a boost of 100%, and BGW's jumping up another ⅔'s, when they were loaded down to twice their rated impedance loads. So it varies from model to model. In fact, some amplifiers will putout more than others of the same line. The disadvantages of running your amplification in this manner is that it will work harder and run hotter. The farther you load an amp down, the more load you put on your output devices, which brings up the distinct possibility of the gear failing on you at a gig. Some amplifiers you can load down to 2 ohms, which is like doubling the load off of the 4 ohm reading. Cerwin-Vega used to make a power amp that was rated at 280 watts at 8 ohms. By the time it got down to 2 ohms, which you *were* able to run it at, it was 1000 watts.

Jim Mathews
Leo's Music
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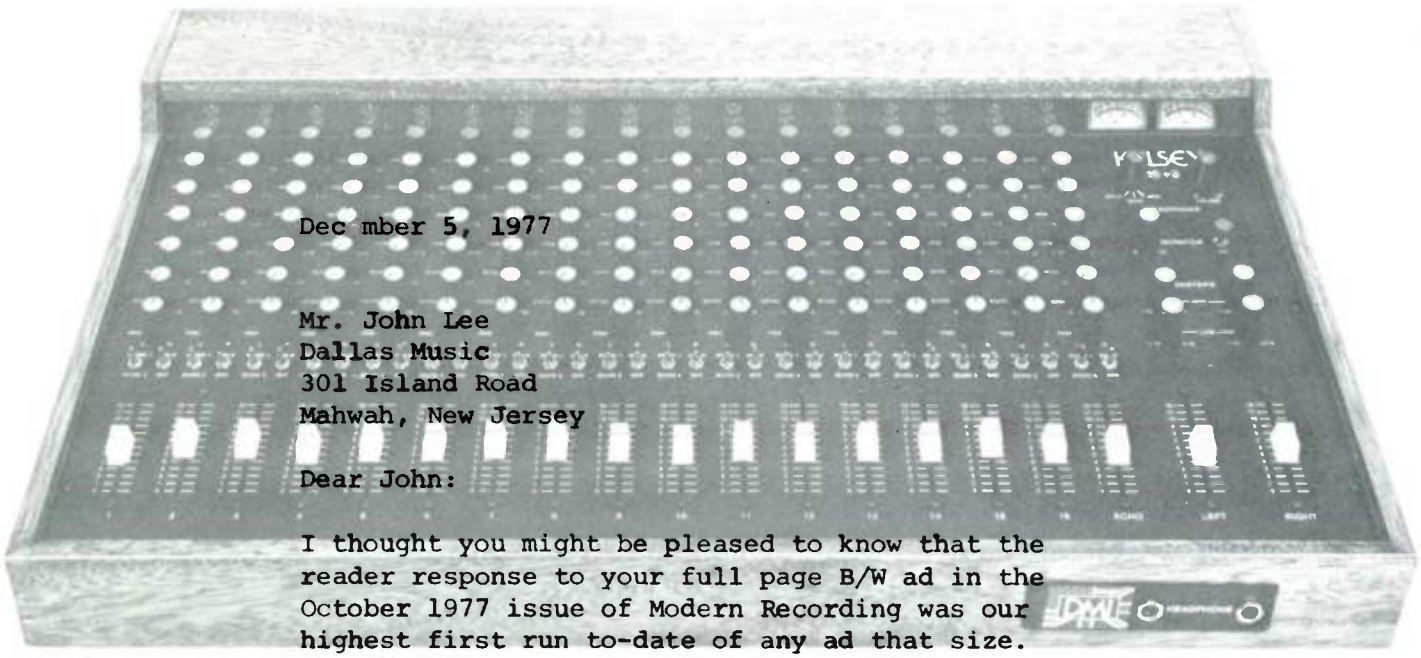

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1

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* * *

2

By Chris Foreman/Altec Sound Products

S3FM MULTI-ADAPTER:

For those who haven't yet discovered this highly useful adapter, the S3FM (Switchcraft part number) is a metal tube with a male XLR-type, three pin connector at one end and a female XLR-type, three pin connector at the other end.

3

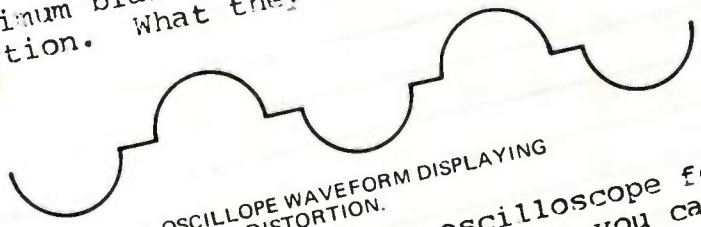
Pads, phase inverters, ground-lift adapters and any number of in-line devices can be constructed inside the S3FM.

NOTCH DISTORTION ADJUSTMENT BY EAR:

Many power amplifiers have an output stage bias adjustment which also controls the amount of notch distortion in the output waveform. The instructions for this adjustment may read something like

this; "Inject a 1 kHz tone (at a low level) at the input to the power amplifier, monitoring the output of the power amplifier with an oscilloscope. Adjust the bias control (screwdriver pot) until visible notch distortion appears on the waveform. Then adjust the bias control in the opposite direction until the notch distortion just disappears." These directions will normally set an

optimum bias current and low notch distortion. What they don't tell you is



OSCILLOPE WAVEFORM DISPLAYING NOTCH DISTORTION.

that you don't need an oscilloscope for this adjustment in most cases; you can do it by ear. The human ear is very sen-

sitive to notch distortion, especially on low-level waveforms (low volume). Thus, a good way to adjust this type of control is to connect the 1 kHz waveform to the input as before, but connect a pair of fairly high quality headphones to the output. Obviously, the level must stay low to avoid overdriving the headphones, but low levels are needed for this

test anyway. Then adjust the bias control until you can hear obvious distortion in the waveform. This distortion, which adds harmonics to the waveform, sounds like a "buzz" on top of the tone, or like the effect a "fuzz tone" device would have. Then adjust the bias control back until the distortion just disappears (until you can't hear it anymore). Do not adjust the control any farther back than this point. If you are in doubt about over biasing the output stages, leave a very small amount of distortion on the waveform; it will probably not be noticable at higher levels since this type of notch distortion shows up primarily at low levels. This adjustment method is especially valuable in the field where an oscilloscope may not be available, and in most cases, your ears will hear the notch distortion at least as well as it could be displayed on an oscilloscope. Only an accurate distortion analyzer can do a better job!

By Bob Heil/Heil Sound

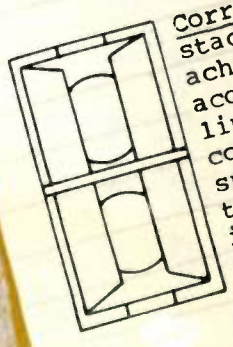
STACKING HORNS FOR HIGHER EFFICENCY:

This is more of a tip, but you can pass it along to your customers who are having "system efficency" problems. The efficency of horn arrays can be greatly increased by proper stacking or placing so that you can take advantage of the physical size

of the enclosure adding to other enclosures producing horn lengths actually larger than the one individual enclosure. We are not saying that by merely stacking one enclosure on top of another, you will "couple" them together. When we refer to "coupling" we refer to an acoustical happening, not just a physical stacking. A gain of 3dB can be achieved by proper-

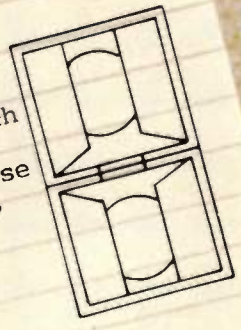
ly coupling two horns together. It would take a doubling of your power amps to get a 3 dB increase, so you can see why proper stacking is so important.

These illustrations show the front loaded horns. Each type of bass horn must be physically placed so their horns are coupled together, not merely stacking them on top of one another.



Correct physical stacking to achieve proper acoustical coupling. With this coupling, the speakers "think" they are actually in a cabinet twice as big as they really are.

Incorrect physical stacking which defeats the entire purpose of stacking.



CONSUMER PROFILE

The Mixer

What it is and Who it is

By L.A. Krause

Mixing consoles, or "desks" as they are called by our English friends, come in all shapes, sizes and variety of functions. As I sit here writing this article, I cannot help but reflect back on all of the different consoles I have used when I was doing recording work for a living. Some of the desks were as simple as a six-input basic mixer such as the Ampex AM-10 (which is still my favorite general purpose portable mixer) to a 40 x 40 Automated Processes console which had more controls than a NASA mission control room. Needless to say, there is a very large difference in price between the two mixers. In my small listening room, I have several hundred tapes that I have recorded or friends have given me over the past ten or so years which have been mixed on every type of console imaginable. Some of the material is excellent, most is acceptable, and the remainder is simply awful. The average audiophile would (and often does) look at the track assignment sheet or log and see what type of equipment was used and without even listening to the tape, try to tell me how good the material will be. This brings me back to the purpose of writing this article, which is an attempt to define what constitutes a *workable* mixing console that can be used in recording and sound reinforcement applications for today's sophisticated musician/consumer.

There seems to be three groups which cause a great deal of confusion about audio products in general. Group number one is the audiophile who has just enough knowledge to really be a pain in the hindquarter. These people have no concept of what a recording engineer actually does or how the equipment is used in a professional environment. As an example, instead of being concerned about the mixer's operational flexibility and reliability, they are more worried about a frequency response of ±0.5 dB from 20.0 Hz to 20.0 kHz, or if the console has a distortion factor of less than .001%. My point is, why worry about something you cannot pick up with microphones, couple through transformers, lay down on tape, reproduce over monitor speakers and cannot hear. To say that a product is bad because it is 2.0 dB down at 20.0 Hz and 20.0 kHz is simply ridiculous.

The second group is the professional recording engineer who actually has to put up with all of the idiosyncrasies of the artists, producers and well-meaning friends. You really have to be part engineer, part psychiatrist, and part crazy (also being six foot, five and having a black belt in karate helps). It is a very difficult job to locate up to forty microphones, make the required noise gate and limiter adjustments, adjust the foldback levels and cue assignments, if required, and make sure that the tape transport and noise reduction is properly aligned and at the same time, try to keep everyone in a good frame of mind. When everything goes reasonably well, the engineer goes home at 4:30 or 5:00 in the morning very tired and practically deaf. After all this work, the end product (discs) will have a frequency response of may ± dB from 30.0 Hz to 12.5 kHz. If only

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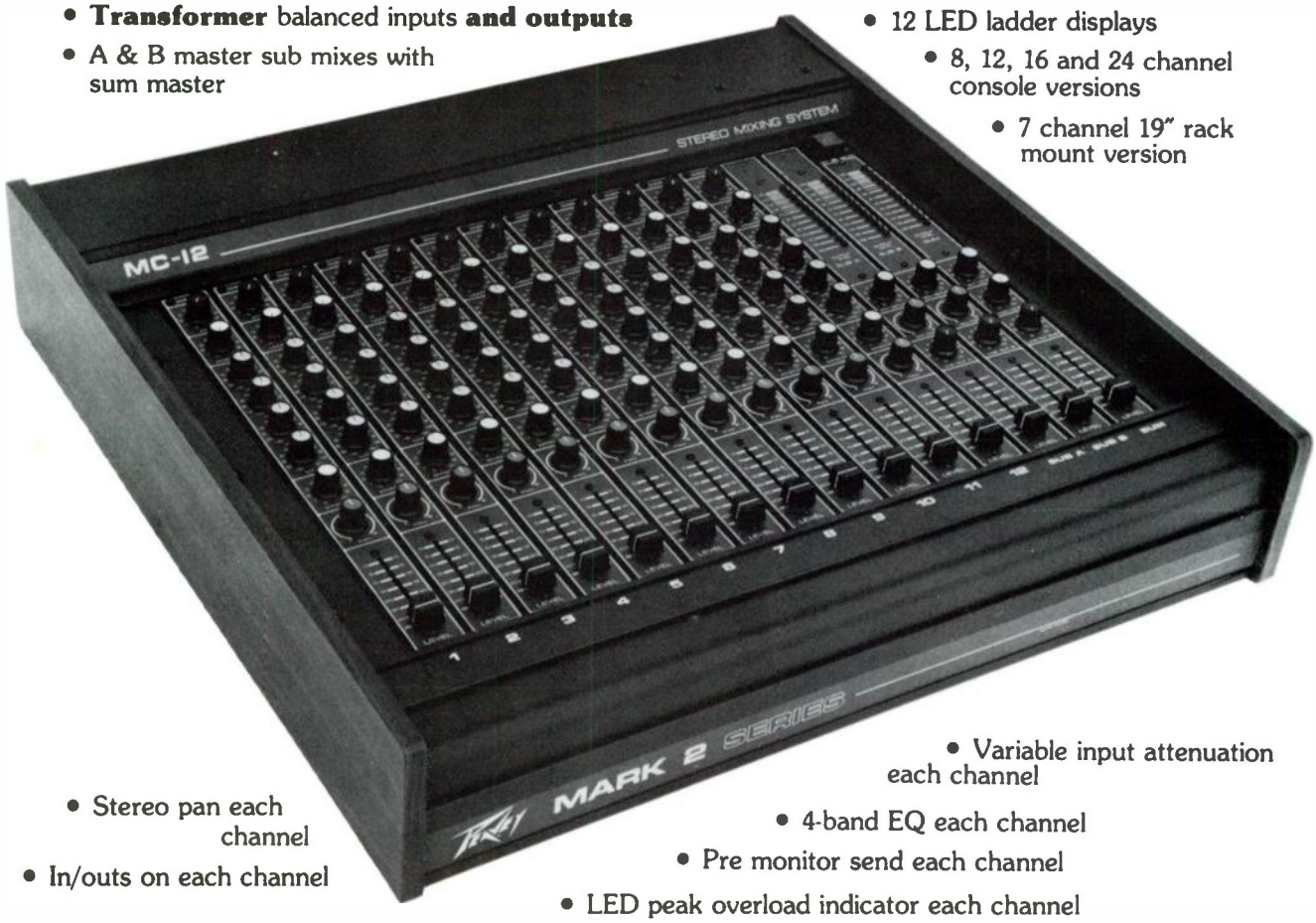

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group one would realize this!

Group three consists of the semi-profssional musician who, lacking the funds to go into a large studio, purchases a mixing console, a few good microphones, and a four-track (more recently, eight-track) semi-professional tape recorder and records his own material. These people will in the future be the most knowledgeable of all three groups. After learning through the trial and error process and after a certain length of time, they make enough errors so that most of the things they do turn out to be quite acceptable. I have heard tapes made by local groups that would compare in quality to high budget studio efforts. This article is directed to the requirements of this group who, if supplied with equipment designed for the intended purpose, will be turning out the next generation of creative, high quality recordings.

It is a fairly well established fact that most groups would like to use their mixing consoles for their sound reinforcement system and then take it home to the basement for use as a recording board. The intention is economically and technically good. It is a shame, however, that most manufacturers do not take these requirements into consideration when designing their products. Don't get me wrong, there are a couple of manufacturers who do offer consoles that can serve dual roles. Even in these units, there are several trade offs in the realm of control flexibility and total system interface. The following represent my opinions as to what should be included on an affordable dual purpose mixing console.

The first stage consists of a transformer coupled microphone input which also supplies the required voltage for phantom powering of condensor microphones. Because there are more and more improvements in microphone technology, there is an ever-increasing output level from the microphone. It is therefore important to have an adequate amount of input padding available. Steps of 10-20-30 dB, along with a gain trim control, allows to optimum dynamic range and signal-to-noise ratio adjustment. A very important feature of any input section is a phase reverse switch. It is absolutely amazing how often cables and direct boxes are wired wrong (all European cables have pins 2 and 3 of an XLR-3 cable reversed). It is also very useful when recording piano and drums. Regardless of how well placed the microphones are, there always seems to be some kind of phase cancellation with these instruments. If you are still skeptical, mix the two channels to mono and reverse the phase of one microphone, *surprise*! A microphone/line or tape input selector switch is also required and will be discussed in a subsequent paragraph.

A channel send and receive patch point must be included between the input preamplifier and the equalization circuitry to facilitate connection of noise gates and/or compressor limiters. Once you have used these devices, you will wonder how you ever did without them. It takes a lot of worry off your mind to know that an over-enthused bass player or drummer is not completely *creaming* their respective tracks.

Equalization for a dual purpose console can be as simple as a bass and treble control or as complicated as a full-blown parametric equalizer. For most applications, a five-band equalization network will provide adequate control for 99% of the requirements. For a five-band equalizer, a shelving response at 100.0 Hz and 10.0 kHz and peak and dip equalization at the other EQ points seems particularly satisfactory. The band widths of the peak and dip functions should be chosen for a smooth, slightly overlapping response. If at all possible, you should leave the critical mid frequency range of 3.0 to 5.0 kHz alone or use as little EQ as possible. If additional EQ is needed, it can be patched in through the preamplifier send and return jacks. High pass and low pass filters are very important and unfortunately, often omitted from a console design. My favorite frequencies are 50.0 Hz and 125.0 Hz for the low end rolloff and 7.5 kHz and 15.0 kHz for the top. Cutoff rates of less than 12.0 dB/octave are almost useless. Low pass and high pass filters will often make the difference between a recording which is dull because the bass guitar is leaking into every microphone in the room and the overhead cymble is getting into the bass drum microphone. A simple insertion of the 125.0 Hz filter into the overhead microphone will often work miracles.

The channel fader should be located immediately following the equalization circuitry. Other channel functions should include a mute switch for removal of the channel from the mixing buss and the channel pan pot. A solo switch is essential on both the recording and sound reinforcement mixer. The solo function allows the engineer to locate potential troubles before and during the performance or recording session. The solo switch simply removes the normal monitor signal from the monitor buss and places the channel output from the "soloed" channel on the monitor buss. This allows you to listen to one input at a time without disturbing your mix.

Three very important functions of each input channel are the echo sends, foldback sends, and monitor sends. For a mixer that is designed for the dual role of recording and sound reinforcement, the echo sends (at least two sends with individual level controls) should be selectable for pre-equalization or post-equalization and channel fader. This will allow you to remove the echo from a mix simultaneously with the channel output when the switch is in the "post" position.

The monitor sends (left, right and center) should also be selectable with a pre-equalization, post fader switch. This will allow you to do a rough monitor mix without disturbing your channel pan pot. When referring to monitor sends, these are the program signals which are applied to the control room monitor system in a recording studio or back to the stage when the desk is being used for sound reinforcement.

The foldback sends (minimum of two) should be selectable pre or post EQ. The channel fader should have no effect on what goes to the foldback busses. An example of foldback is when you are you laying down a vocal track when listening to a pre-recorded harmony. The harmony track would not be assigned to a record track but instead be folded back (sometimes called "cued") to the vocalist headphone monitor system. This allows the vocalist to lay down his track in sync with the harmony. The two tracks are then combined during the final mixdown.

At this point, I am going to make a departure from the somewhat conventional view of panning, assignment and tape overdub functions. Because this article is about the musician who wants to record his own material, I have a few suggestions that will help in offering the correct ocnsole for both recording and sound reinforcement. Let's look at the actual application before going any further. With a studio desk the whole idea is to record

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a certain number of tracks, mix these down so you have a spare track or two and then add in any additional harmony, strings or vocals. With the advent of affordable four-track recorders, it would seem that a mixer with eight, twelve or sixteen input channels and four assignable outputs would be sufficient. This is true to a point and the point is that you are limited to the use of a four-track recorder. There are now available, eight-track recorders within the budget limitations of the average musician. What are you going to do now? My design criteria is to have the capability of using a four-track eight-track or sixteen-track recorder with a standard console. Because the average consumer/musician has been conditioned to think that the console must have the same number of main outputs as the recorder track inputs, this requires a little explanation. Consider the following: How many outputs go to the final master recorder? There are two outputs (*forget* quad, *most* people have) regardless of the size or cost of the console. If you can record and mixdown any number of tracks to a two-track format, the mission will have been accomplished.

It is possible to design a mixer which is affordable and has the operational flexibility required for multi-track recording and mixdown and still maintain its usefullness for sound reinforcement purposes. The key to this flexibility is to provide a tape send for pre-equalization whenever completely "dry" recording is required or post-equalization and channel fader. In the post-equalization and fader mode, the channel fader becomes the record level control. It should be noted that when in the pre position, the high and low pass filters are available if their use is necessitated. With a tape send available on each channel, it is possible to use any recorder/console combination. To further enhance the flexibility of the mixing system, an *overdub normal* switch is provided so that the record head output (sync mode) may be monitored through the foldback system. In the *overdub* position, the signal from the tape input is routed directly to the foldback level controls. In the *normal* position, the foldback signal is derived after the equalization circuitry but before the channel fader. This configuration allows the engineer to change the mix level without disturbing the foldback levels. Interlocked with the *overdub normal* switch is a set of controls for noise reduction switching. You must be able to put the noise reduction system into the bypass mode when monitoring off the record head.

The master section of a mixer should contain submaster and grand master control functions. Assuming that there are four main busses, there will be four submasters which control the mixing level applied to the main outputs through the grand master fader. Although only two busses are required for stereo mixdown, the availability of four submasters allow for a very flexible grouping when used for sound reinforcement. As an example, submaster one may be assigned to percussion; sub two, vocals; sub three, instruments; and sub four, keyboards. Because all of the subs will combine for a mono mix, it becomes very easy to derive a good balance. While on the subject of stereo or mono sound reinforcement, I am of the opinion that trying to do a stereo sound reinforcement mix is unfair to the artists and the people who pay the bills—the audience. I have, through experience and the attempts of others, learned that the only place you have a reasonable stereo image is at the mixing console. It is for this reason that on a mixing console, regardless of the number of output busses, there should always be a sum mix of the busses in a monaural output.

The master section also contains the level controls for the foldback busses, control room monitors, solo levels and effects returns. The master section can be extremely complicated or relatively simple and straightforward. I have found that the more complicated something is, the greater the likelihood of messing something up. There have been many times that an engineer has had to spend an hour or more trying to find out what's wrong with his console when someone simply rotated a knob. You are afforded the luxury of time in a studio; when doing live performances, it is an entirely different situation.

Metering of the output levels can be accomplished by several methods. The VU meter is certainly one of the most widely used and accepted indicators. It is a shame that they have three basic problems: They are expensive, they are fragile, and they can only provide approximately 23.0 dB of indication range. On the plus side, they are visible in bright sunlight. The L.E.D. (light emitting diode) display is beginning to be used in more equipment as time goes on. The L.E.D. array offers several advantage is that dynamic ranges of 40.0 dB or greater can be covered accurately. The L.E.D. array is much more accurate because there are no moving parts so that even the fastest of transients are displayed. You can see a peak that goes to +8 that would not move the needle of a VU meter. L.E.D. displays are very rugged; they can withstand a shock which would ruin the best of VU meters. In addition, because L.E.D. displays are completely solid-state, there are no lamps to burn out.

All microphone inputs and the line outputs from a console must be transformer coupled. There is an electronic approximation of a transformer in that an operational amplifier can be configured to provide a balanced input at audio frequencies. The key phrase is "at audio frequencies". An operational amplifier can be used to approximate a transformer and at audio frequencies, it works fairly well. It is unfortunate that some of the most common forms of noise, such as C.B. radios, light dimmers, and radio stations are not in the audio range. Because the operational amplifier does not have gain at the higher frequencies, the transformer action ceases to exist and result will be interference. A police radio overriding a vocal part at a concert is quite disturbing, but the running of a perfect vocal track in the studio can be disastrous. Operational amps are cheap and light in weight, but they cannot perform to professional standards as replacement for even a medium-quality transformer.

I have endeavored to convey some of the knowledge I have learned from trial and error, on-the-road experience, being in the studio, and from a university education. It has been said before that you learn from mistakes and if you make enough mistakes, you will eventually become quite knowledgeable. There is a lot of truth in that statement. I have two thoughts that I relate to the students in the classes I teach. The first of which is that the stupidest question in the world is the one that is not asked; and the second is that if you give a man a fish, he will eat for a day, if you teach a man to fish, he will never to hungry. I hope that I have taught you how to fish for a console.

It's The Little Things That Count

See us at the Frankfurt International Fair, Frankfurt, West Germany, Feb. 26-Mar. 3, Hall 5A Stand 52613.

MXR Innovations Inc., 247 N. Goodman St., Rochester, New York 14607, (716) 442-5320.

MXR Professional Products Group

CIRCLE 90 ON READER SERVICE CARD

Western NAMM Sound Shoppe Product Roundup

Eight years ago the first NAMM Western Market was held with some 64 exhibitors displaying their wares. Last January's NAMM Western Market covered 46,840 square feet at the Disneyland Convention Center in Anaheim, California, and featured 222 exhibitors and thousands of visiting participants and dealers from all over the world. The expansion and growth of this industry is staggering and it was clearly evidenced by the numerous new products introduced at the Southern California expo.

Fender MA 6 Sound System

CBS/Fender proudly displayed their new consumer PA, the MA 6 Sound System featuring pre-fader monitoring. This unit's mixing console allows the sound man to monitor all six channels together or seperately. The MA 6 offers 6 channel monitoring ahead of the channel volume controls, plus 5-band monitor EQ. Distortion is minimized by gain overload controls, each channel has an indicator light in case of input overload. The desk also includes an effects re-patch system as well as a built-in spring reverb for the main mix, the monitor mix, or both. The speaker enclosures that come with the Fender MA 6 Sound System are thiele-aligned vented enclosures—each loaded with two 12" speakers and a mid/high frequency horn along with a built-in 18 dB/octave crossover at 1.6 kHz. The retail cost of the system is suggested by the manufacturer at $1,200.00.

CIRCLE 1 ON READER SERVICE CARD

Meteor Light and Sound Company, a division of Hammond Industries Inc., debuted their Clubman One-One-M Mixer, for the consumer/club musician, and the Powermaster 75 Stereo Amplifier for professional or high-end high fidelity home use. This 20 pound power package pushes 75 watts per channel, both channels driven, at any frequency from 20 Hz to 20 kHz, at any power level from 250 mW to 75 W with less than 0.1% total harmonic distortion. The suggested retail price is $449.00.

CIRCLE 4 ON READER SERVICE CARD

New on the Roland 'shelf', in addition to their GR-500 Guitar Synthesizer's reintroduction, were several items including the Revo Rotating Sound System, featuring no moving parts; and a new low-noise, high performance effect unit called the Boss Bi-flanger. Especially unique was Roland's Microcomposer MC-8, an eight-voice music processor as well as a digital sequencer that professional

Clubman One-One-M Mixer

musicians can use as a helpful composing and recording tool. The value of the MicroComposer, according to Roland, lies in its ability to completely control a synthesizer in the production of electronic music in the studio. The unit translates musical notes into digital information, which can be loaded through the MC-8 or directly from a synthesizer keyboard. Using a four-channel open reel tape deck, the MC-8 can provide up to 32 different voice tracks. Such factors as pitch, tempo, dynamics, and timbre can be assigned numerical values. Multi-channel output provides for controlling as many as eight independent voice lines at the same time. Memory capacity is sufficient to enable all voice lines in an average popular song to be stored.

MicroComposer MC-8

"The MC-8's Sync Function allows precise multi-track recording, and its multi-channel output will significantly multiply the voice capacity of any recording system," Roland U.S. President Tom Beckmen explained. "For example, with an eight-channel recording system, more than 60 independent voice lines are possible without resorting to second-generation recording overdubs."

Timing for animation or commercial work can be done within 0.1 second. Patching to Roland's System 700 or System 100 synthesizers is simple. The unit also is compatible with other makes of synthesizers as well. The MC-8 carries a suggested retail price of $4,495.

CIRCLE 5 ON READER SERVICE CARD

TAPCO announced the addition of dual-channel power amplifiers, featuring PowerLock™, to their line of mixers and equalizers for pro sound use. Both basic amplifier models may be operated as two independent channels, or as a single-channel unit bridged configuration. Model CP120 is rated at 61 watts per channel, is capable of 510 watts output in the bridge mode. Model CP500M is rated the same as the CP500, but has a readout package that includes true peak-reading meters, a blown fuse indicator, and a thermal protection indicator. Both basic amps are designed for 4-Ohm operation for greater multiple-speaker hookup capability.

PowerLock™, an integral part of the TAPCO design, senses any large input signals that could cause prolonged clipping. PowerLock™ then controls the output level during the period of time that would otherwise be perceived as clipping distortion. Normal music transients, less than 1 ms in duration, are allowed to pass unaltered. The result according to the manufacturer, is complete absence of distortion and greater average power output, with input signals up to 30 dB above the amp's normal +4 sensitivity.

The amps feature double stacked ¼" jack inputs, all-steel monocoque construction for professional, on-the-road use, and high reliability components throughout. Transformer-coupled balanced XLR inputs are available as an option at additional cost. The CP120 has a stereo headphone jack on the front panel for monitoring purposes. Pro net prices for the units are: C 120 $339.00, CP 500 $649.00, and the CP 500M $779.00.

CIRCLE 6 ON READER SERVICE CARD

Teac and TASCAM unveiled their new FinanceAmerica plan (see related article this issue), while dbx, Shure, MXR, Quilter, JBL, Gollehon, Cetec, and AKG all had their sound reinforcement and recording related gear outfront for all to see and, in some cases, hear.

CIRCLE 3 ON READER SERVICE CARD

The guitar synthesizers, introduced at last year's Atlanta NAMM show, were out West in strength. ARP had their Avatar, Roland their GR-500, Ampeg their Patch 2000, and 360 Systems demonstrated their particular unit. In fact, synthesized sounds are spreading and percussion synthesizers were also 'upfront' from companies like Star Instruments and Pollard Industries.

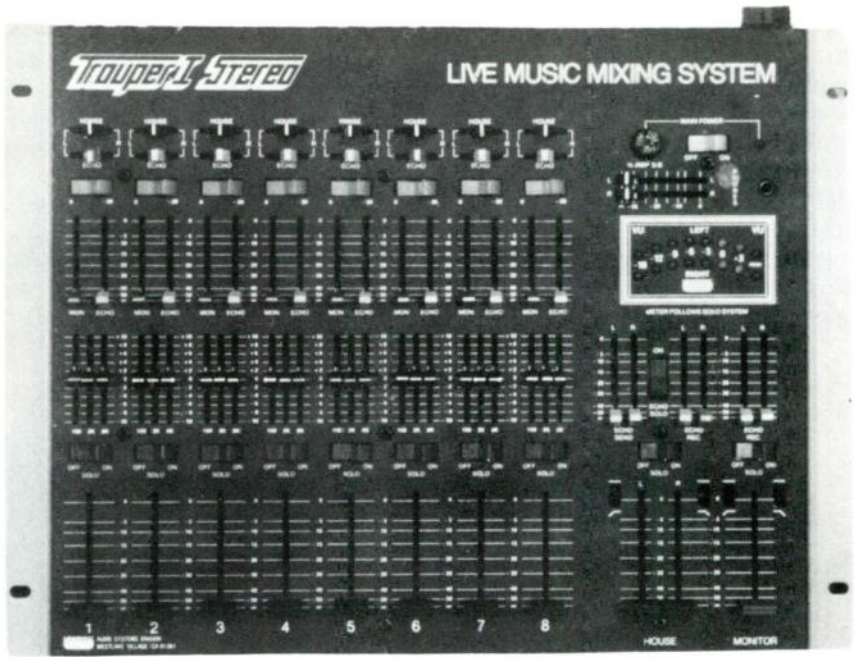
DiMarzio, Mighty-Mite, and Lawrence Sound Research were all well represented in the replacement pickup field. Especially stunning was DiMarzio's new four-color catalog that featured their new acoustic pickups and contact transducer as well as their Key Mix System. The Key Mix System is a unit designed for three uses: first, as a low cost yet accurate piano pickup; second, as a mixing system for up to three acoustic instruments; and third, as a deluxe pickup mixing system for the acoustic guitarist who wishes to utilize amplification to its fullest capabilities. The system is

Tapco CP120 Amplifier

equipped with three Acoustic Model DiMarzio Contact Pickups, which are fed into a pre-amp mixer (pre-amp mixer available seperately also). The mixer has individual volume controls for each input, as well as a master volume, bass and treble controls.

CIRCLE 2 ON READER SERVICE CARD

Hohner unveiled a new line of electric guitars, named the 700 Series. In addition to Hohner's *instrument* contribution was a first-showing for their selection of seven sound modifiers. There's a volume pedal, a phase shifter, two distortion effect units and three tonal effect units. The Vol-Kicker volume pedal provides a smooth and quiet volume control by depressing and releasing the foot-pedal. The Vari-Phaser, a battery powered effect pedal, controls and colors the phase shifting process by adjusting the three controls "Harmonic Emphasis", "Phasing Speed", and "Phasing Intensity". The two distortion effect units, Dirty Booster and Tri-Dirty Booster, offer a variety of functions and are battery operated. The Dirty Booster provides *Straight Distortion* and *Variable Volume* for distortion flavorings. The Tri-Dirty Booster provides three functions, *Straight Distortion, Sustain,* and *Tone Controlled Distortion.* The Wah-Wah'er is designed with a foot pedal and produces a Wah-Wah effect as well as volume control. The Dirty Wah-Wah'er pedal gives the consumer/musician five different tonal effects; *Volume, Wah-Wah, Distortion, Distortion/Wah-Wah,* and *Volume/ Distortion.* The Multi-Exciter, also a pedal, features five different functions; *Volume, Wah-Wah,* a *Surf* effect, *Tornado* and *Siren* effects. All three units



Trouper I Stereo Mixer

are battery powered. All of the Hohner Sound Modifier pedals are cast aluminum with durable black crinkle finish with a non-skid rubber top.

CIRCLE 7 ON READER SERVICE CARD

The folks at Uni-Sync were at NAMM in force and their newer items included a stereo version of the Trouper I Live Music Mixing System as well as the Model 1003 Microphone Splitter. Particularly new for this BSR company is their new line of Professional Power Amplifiers. Immediate plans call for production of 50 and 100 watt amps with a 250 watt version currently being designed. The Model 50 and Model 100 have the following features: Individual level controls for each channel, power/peak/protect indicator, XLR and ¼" phone connectors, stereo/mono bridging slide switch, 5-way banana binding posts, independent power supply for each channel, new modular construction for serviceability, on/off transient speaker protection for DC offset, full complimentary output stage and limiting circuitry for SOA. The Model 50 will have a suggested retail price of $329.00 and the Model 100 $549.00.

CIRCLE 11 ON READER SERVICE CARD

News from Acoustic included the model 408 Bass Enclosure, the Model 813 compact PA Enclosure, and a couple of guitar oriented amp heads. The Model 320 head is designed for bass use while the model 330 is made for the guitarist. Each unit includes a 5-band graphic equalizer and a 300 watt RMS power rating. Two small amps are new for Acoustic; the model 114 is a 50 watt/two 10" speaker guitar unit while the Model 116 is designed for the bassist who needs a compact power plant.

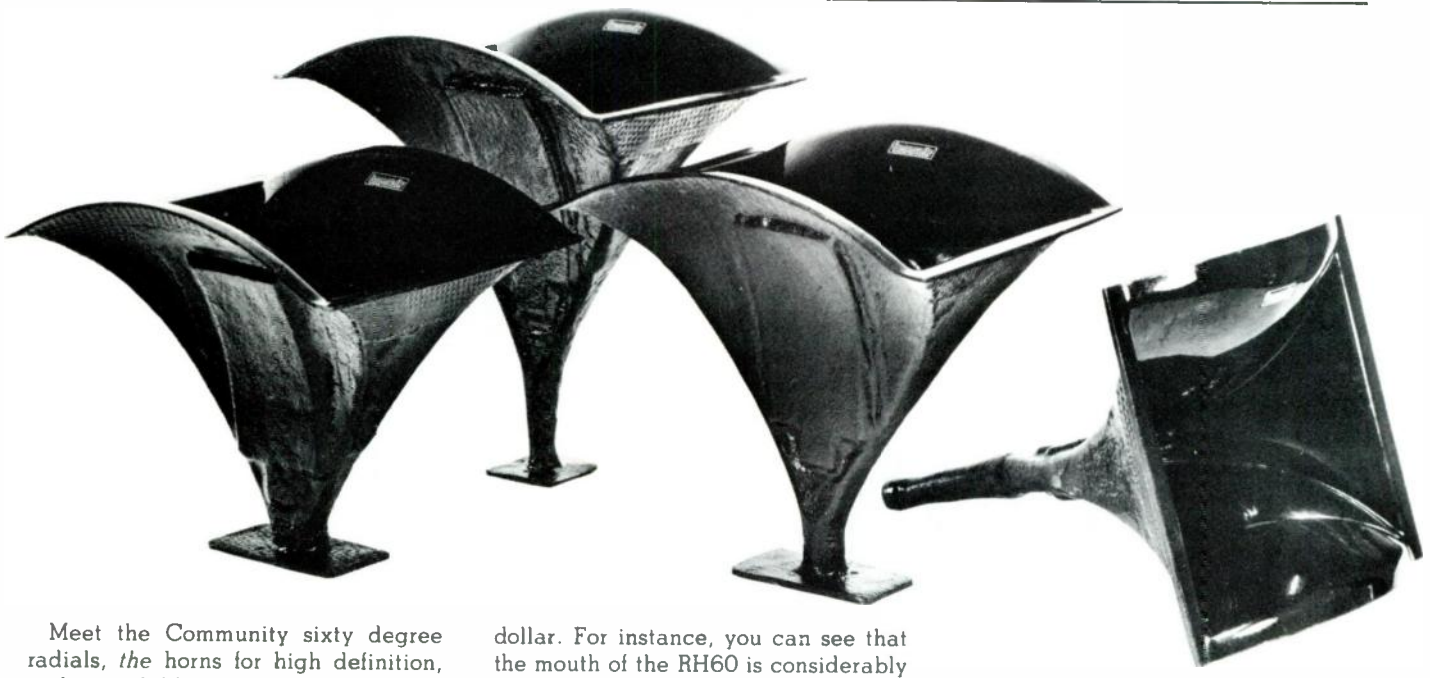
The 116 is structured with ¾" solid core plywood for strength and it houses a single 15" bass speaker. According to Acoustic, the 116's pre-amp responds quietly and the 75 watt RMS wattage rating gives the bass player a uniquely powerful yet compact amplification unit.

CIRCLE 8 ON READER SERVICE CARD

The items displayed at the NAMM Western Market were many and varied, we couldn't possibly include everything that was introduced. But wait until June 24, 25, 26, and 27 in Chicago! You ain't seen nothin' yet!

Hohner Sound Modifications

RH60 RADIAL HORNS



Meet the Community sixty degree radials, *the* horns for high definition, understandable sound. The horn pictured is our RH60, the midrange mainstay of the large system We've recently added two new sixty degree horns to our line for HF and VHF projection, the SRH60 and the SQ60.

Some people still think that our horns and cabinets look a little strange, perhaps not realizing that at Community shape and construction are determined by the laws of physics, not marketing, packaging or the almighty dollar. For instance, you can see that the mouth of the RH60 is considerably taller than that of comparable sixty degree horns. Why? Well, if a horn is to act as a wave guide at its lower operative frequencies (which it is) it must have a tall mouth to support the larger wave forms generated near crossover. The idea of a thin, wide-mouthed radial may be pleasing in terms of packaging and handling, but it is a convenience that does not pay off in operation. Some conveniences that do pay off in operation are one-piece construction, low resonance, high strength-to-weight ratios and the meticulously executed design that characterizes a Community horn.

Would you like more information? We recently published a catalog which details the performance of all Community products. Already it has been called a must for anyone wishing to design a sound system on a professional level. Please write or call to order.

RESPONSE AND SPL

RH60-A DRIVER: **GAUSS HF4000**

BANDWIDTH PINKNOISE: **250Hz-16KHz**

1 Watt @ 1 Meter **107.24** dB-SPL

1 Watt @ 4 Feet **107.28** dB-SPL

									−12	−7	−6	−4	−3	−1	0	+1	−1	−2	−1	−3	−5	−5	−9	−9	−9	−13	−17	
Hz	40	50	63	80	100	125	160	200	250	315	400	500	630	800	1K	1.25	1.6	2.0	2.5	3.15	4.0	5.0	6.3	8.0	10.0	12.5	16.0	KHz

SRH60-B DRIVER: **ALTEC 288.16G**

BANDWIDTH PINKNOISE: **350HZ-16Hz**

1 Watt @ 1 Meter **108.99** db-SPL

1 Watt @ 4 Feet **105.52** dB-SPL

											−10	−6	−4	−2	0	0	0	0	0	−2	−2	−4	−8	−8	−6	−10	−12	
Hz	40	50	63	80	100	125	160	200	250	315	400	500	630	800	1K	1.25	1.6	2.0	2.5	3.15	4.0	5.0	6.3	8.0	10.0	12.5	16.0	KHz

SQ60-C DRIVER: **EMILAR EA. 175.16**

BANDWIDTH PINKNOISE: **800Hz-16KHz**

1 Watt @ 1 Meter **103.85** dB-SPL

1 Watt @ 4 Feet **102.14** dB-SPL

														−6	−2	−2	0	0	+2	0	0	0	−2	−2	−5	−12	−16	
Hz	40	50	63	80	100	125	160	200	250	315	400	500	630	800	1K	1.25	1.6	2.0	2.5	3.15	4.0	5.0	6.3	8.0	10.0	12.5	16.0	KHz

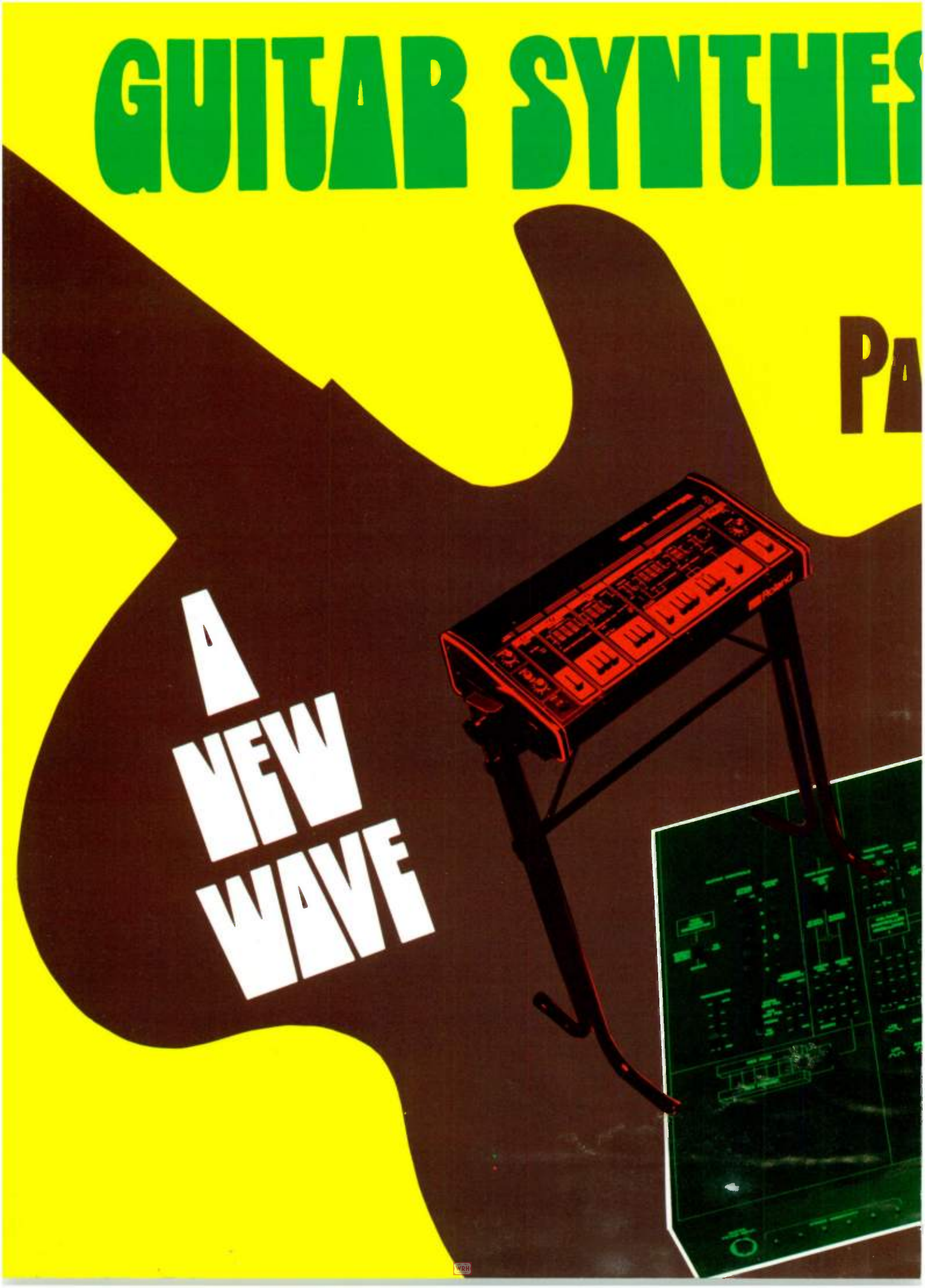
Community

Community Light & Sound, Inc. 5701 Grays Avenue Philadelphia, PA 19143 (215) 727-0900

CIRCLE 89 ON READER SERVICE CARD

GUITAR SYNTHES

PA



IZERS

S ONE:

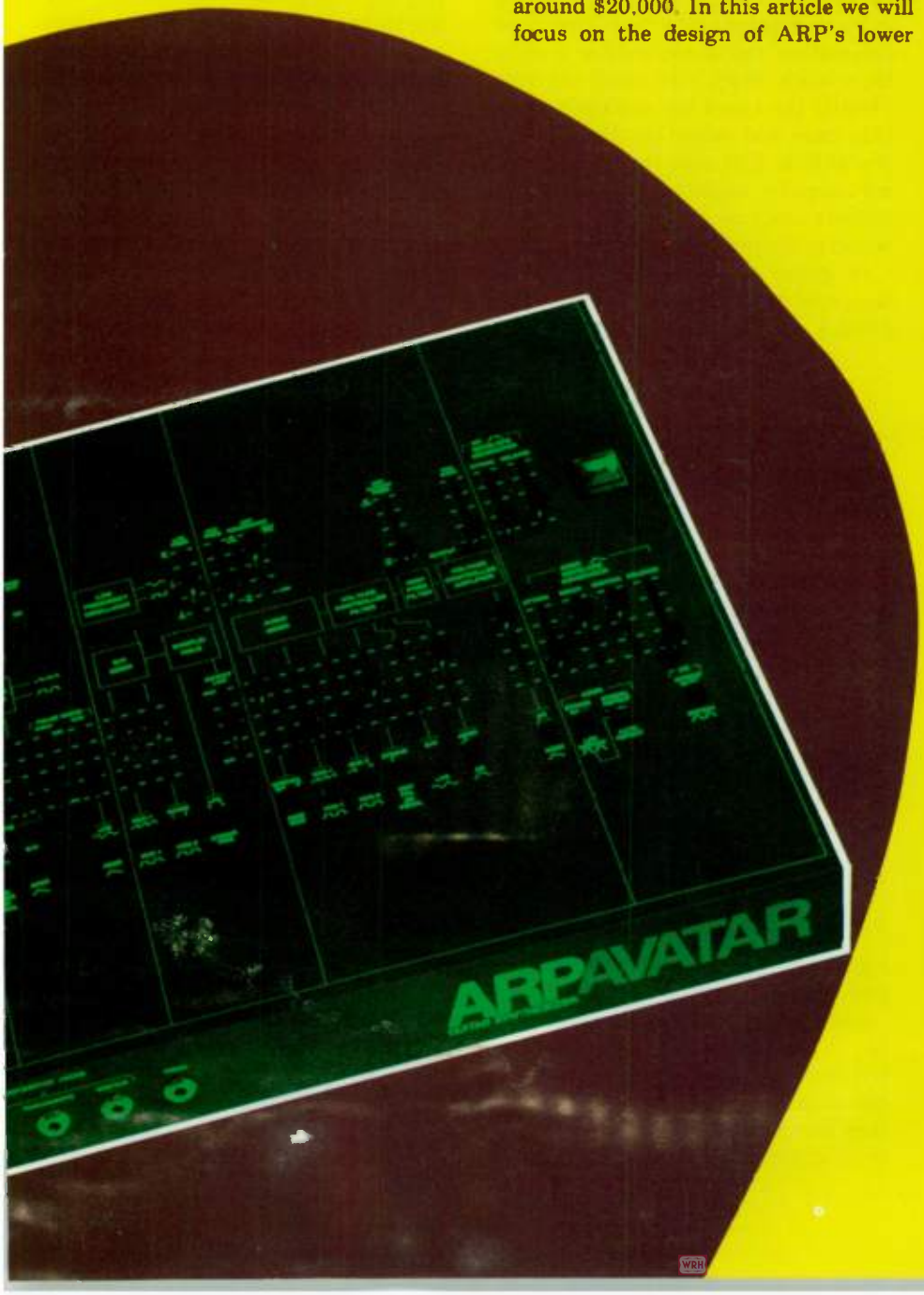
ARP AVATAR

By David Friend*

Definitions

In any rapidly expanding market, product definition can become clouded by manufacturers attempting to ride an upsurge in product demand. A number of confusing labels have already been applied to electronic guitar devices, so it seems worthwhile to begin with a review of definitions and to understand what makes a true "guitar synthesizer."

Several manufacturers have recently introduced true guitar synthesizers. There are several units in the two thousand dollar price range, and ARP makes a grand-daddy limited edition unit called the Centaur that sells for around $20,000. In this article we will focus on the design of ARP's lower priced model, the Avatar. There are also a number of other products which are dubbed "guitar synthesizers" by their manufacturers but which, in fact, are merely modifiers.



Modifiers, such as fuzz boxes, wah-wah pedals, or phasers, are not synthesizers. Although some modifiers claim to produce "synthesizer effects," these devices always rely on taking the actual audio signal from the guitar and processing it through various electronic circuits to create modifications or distortions of the original signal. Such devices produce no sound of their own. When the guitar strings stop vibrating, then the sound stops.

A synthesizer, on the other hand, creates its own sound from scratch. The sounds that emerge are generated by oscillators, not the guitar strings. Consequently the sound that comes out of a guitar synthesizer may have nothing in common with the sound of the actual guitar. If you'll think for a moment about keyboard synthesizers, you probably know that the keyboard itself does not generate sound. The keyboard simply tells the synthesizer circuitry what notes to generate. The guitar synthesizer works on the same principle. The main function of the actual guitar sounds is to tell the synthesizer circuitry what note you are playing on the guitar so that the synthesizer circuitry can generate a corresponding note. The audio signal from the guitar does not pass all the way through the system as with modifiers.

In general, if a unit does not have tone generating oscillators which can produce tones of infinite duration, then it is not a true guitar synthesizer. If the only sound being generated is coming from the guitar strings themselves, then you are really dealing with a modifying device rather than a synthesizer.

In order for the guitar synthesizer to follow the pitch you are playing, a special circuit usually called a "pitch-to-voltage converter" or "pitch extractor" is needed. The job of the pitch-to-voltage converter is to analyze the guitar signal and determine what note is being played. Once this pitch has been determined, it then produces a voltage which tells the oscillators in the synthesizer what note to play.

While it is easy for a trained human ear to determine pitch readily, it is a difficult task for an electronic circuit because the waveform from a guitar is

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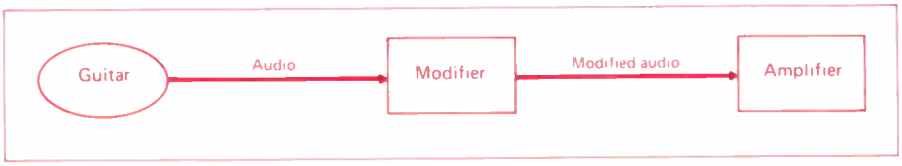
Presented at the 58th Convention of the Audio Engineering Society. Reprinted with permission.

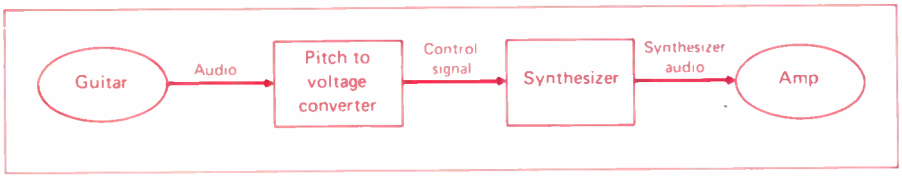
rich in harmonics and is very unpredictable. The development of effective pitch-to-voltage converters has been the last hurdle in the production of the guitar synthesizer. The operating of a guitar synthesizer will depend entirely on the accuracy, speed and adaptability of the pitch-to-voltage converter.

The Hex Pickup

One of the requirements for successfully extracting the pitch of a guitar string is to have the pitch-to-voltage converter look at the output of only one string at a time. The waveform that is produced by a single guitar string alone is quite complicated for an electronic circuit to analyze. If the matter is further complicated by having a mixture of signals from several different strings, the job becomes impossible. Consequently guitar synthesizers require a means of listening to one string at a time or each string individually.



Standard guitar pickups produce one output for all six strings. Clearly such pickups will have limited effectiveness with pitch-to-voltage converters since they would always be producing signals from several strings at once.



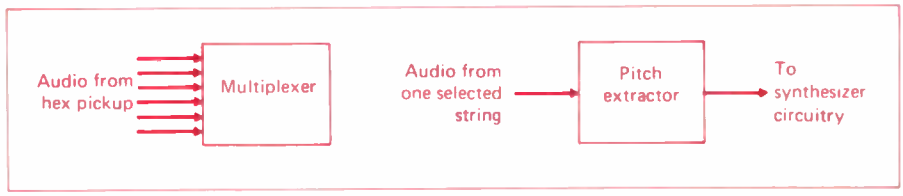
To solve this problem, we decided to obtain a separate electrical signal for each string. The ARP uses a relatively small pickup that can be attached to most existing guitars and produces a separate output for each string. The six outputs appear on the pins of a snap-action connector which is mounted on the user's guitar. The ARP Hex Pickup uses what we call "focussed magnetic fields" to provide nearly total isolation between strings.

Each pickup consists of six variable reluctance transducers. Because of the small size of the transducers, extremely fine wire had to be used in the coil assemblies. In addition, the pickup employs a fairly sophisticated magnetic structure which maximizes the output of each transducer while minimizing interaction between coils.

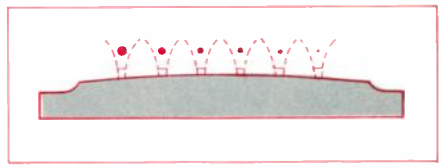
The Pitch-To-Voltage Converter

Let's take a closer look at the problem of pitch extraction and some of the theoretical limitations. Pitch-to-voltage converters are basically pattern recognition computers and are, of course, far inferior to the human ear at pitch recognition. The human ear can pick out and identify a pitch in a signal that is 99% noise. If you were standing in Times Square in the din of rush hour commotion and someone blew a whistle a block away, you could not only identify the sound but you could probably hum and remember the pitch of the whistle. But even the most powerful computer would require a long and tedious analysis to find that whistle sound in the general background noise.

In guitar synthesizer pitch extraction, speed is of the essence. When you pluck a string, you don't want to have to wait around for the circuits to analyze the signal and produce an output. So it stands to reason that a cleaner and more clearly defined original signal can be analyzed faster and more accurately than a complex signal that includes a lot of audible garbage.



Let's talk about accuracy and speed for a moment. Because music is ordered and organized, the human ear can sometimes "hear" pitches before they're even played. We have the ability to anticipate sounds based on our understanding of harmony, melody and rhythm. When a chord is being resolved, you *know* what the next note will be. And even if that next note is out of tune, you know what it should be and can compensate in your listening somewhat.



A pitch-to-voltage converter is relatively stupid. Garbage in, garbage out. It has no musical training. If your intonation is off, the pitch extractor will follow right along—out of tune. If you scrape your strings or don't pick cleanly, it will go wild trying to figure out what all the noise is about. It doesn't have the ability to disregard playing errors and sloppiness the way the human ear does. And what sounds acceptable on a guitar sound may sound awful when you are synthesizing a chorus of strings.

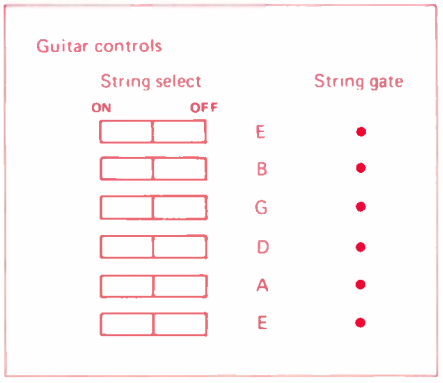
However, it is possible (and highly desirable) to design a pitch-to-voltage converter that will ignore many types of spurious signals. As a simple example, the pitch-to-voltage converter we designed for the Avatar knows the normal pitch range for each string, and will ignore signals that are outside this range including most scraping sounds, thumps, pick noise, and the like. It is also smart enough to tell the difference between an honest pluck using a pick and an accidental brushing of the strings with the fingers or noise from fretting. It can also "read" an implied fundamental from a harmonic series even if the fundamental itself is totally absent.

Speed is probably the most important and most difficult aspect of pitch extraction. The human sense of rhythm is extremely precise and it is easy to perceive even the slightest delays in response. Experiments we did a few years ago at ARP using keyboard synthesizers showed that most musicians were annoyed by delays of 40 milliseconds and most could easily perceive delays of 20 milliseconds.

The frequency of the low E string on a guitar is 82.4 Hz. That's about 12 milliseconds for one complete cycle. In order to avoid speed problems, then, the pitch-to-voltage converter would have to complete its job in just two cycles of the input waveform. Now, two cycles is not very much information. Nevertheless, we decided that we would set as a design objective the completion of the pitch extraction process within two cycles of the input waveform.

Front End Multiplexer

Because of the relatively high cost of the pitch-to-voltage converter circuitry and because of the musical considerations of lead-line playing, it was decided to use one pitch-to-voltage converter in the Avatar. The pitch-to-voltage converter is preceded by a six-to-one analog multiplexer that passes the signal from the last string plucked. In the event that more than one string is plucked at a given instant, the stronger signal will be selected.



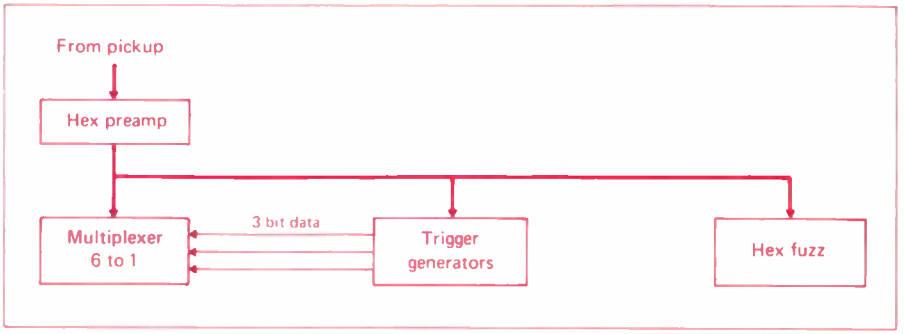
The six "string gate" lights on the panel are connected to the multiplexer and tell you which string is being selected by the multiplexer at any time. Beside each switch is a letter, corresponding to the open strings of the guitar, and a red LED light. These switches enable the player to assign the synthesizer section to any string on the guitar and red LED will be lit as long as the signal is usable.

Front End Detail

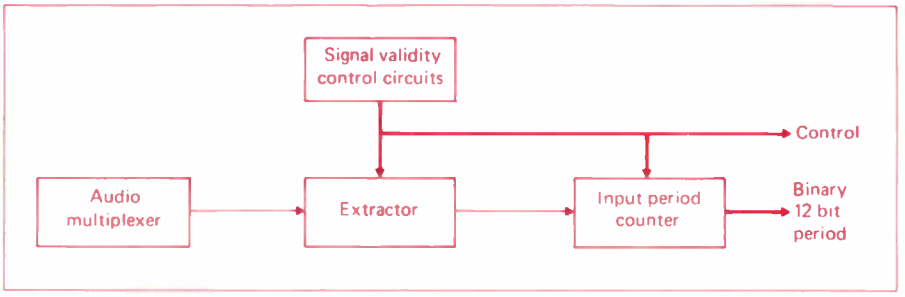
Each of the signals from the guitar strings is independently amplified in the 'Hex Preamplifier Circuit' on Board E. The six outputs of the 'Preamplifier Circuit' are sent three places: 'The Fuzz Circuit,' 'The Trigger Generator Circuit,' and the 'Audio Multiplexer Circuit.'

'The Fuzz Circuit' produces (if selected) clipped guitar signals to the guitar output. Since the fuzz is created per string and then mixed, the effect is a "clean" distortion with low intermodulation distortion. The Hex Guitar outputs (fuzz and straight) are byproducts of the extraction process and have no function in the conversion of guitar signals to a control voltage.

'The Trigger Generator Circuit' produces a trigger for each plucked string. The six trigger outputs are routed through the 'String Select Switches' (which enable or disable each string) to the trigger select circuit. From the triggers, the Trigger Select circuit produces a three bit strong code for the last note played. A trigger pulse is also generated for each pluck to reset the extraction circuitry on new notes.



'The Audio Multiplex Circuit' selects one of the six guitar signals and routes it to the Extractor module (E' Board) according to the three bit string code (last note played). The code is also sent to the extractor circuit to set up the approximate pitch range of the string signal it is to receive.



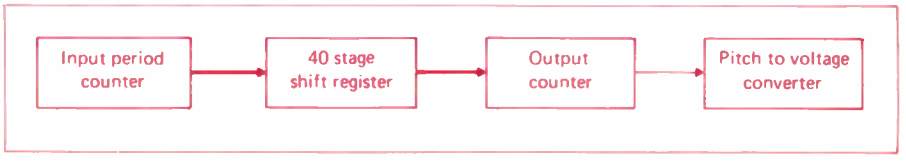
'The Extractor' transforms the guitar signal into a clean pulse wave which is only gated out when the signal is considered valid (stable and of usable amplitude).

Control circuitry insures that the extraction begins only on a valid string pluck, insures the signal is valid and gates the signal out only while the signal is clean.

'The Input Period Counter Circuit' digitally measures the period of the pulse wave from the 'Extractor.' 'The Input Period Counter' is a 12 bit counter clocked at 250 kHz which resets to zero for each new pulse wave period. The 12 bit period number is then latched and held until the next period count.

This number is loaded into the '40 Stage Shift Register Circuit.'

On an initial string pluck, the shift register is clocked fast (250 kHz) to process the first few period counts quickly. The shift register is then clocked at a slower rate of 1 kHz so that the 12 bit pitch number emerging from the tail end of the shift register will be 40 milliseconds old. This delay in transmitting the pitch information allows the system to latch onto correct pitch information when a string is damped by releasing a fret. Although it is not readily apparent from listening to a straight guitar signal, when a note is damped out by raising the finger off a fret, the pitch drops as the amplitude dies away. Without the 40 millisecond pitch delay, the output



counter would latch onto a flat pitch at the end of each note released in this manner.

The Output Counter converts the 12 bit period code into a high-frequency pulse wave which is a constant multiple of the frequency of the guitar signal. This pulse wave is fed into a high-speed exponential pitch-to-voltage converter which produces a standard 1 volt/octave control voltage output. This output signal is connected to the synthesizer circuitry in much the same way that a keyboard control voltage would be used in a conventional keyboard synthesizer.

ROLAND GR-500

By Jeff Baxter

Last year at the NAMM exhibit in Atlanta, Roland Corp. USA introduced its all-new GR-500 Guitar Synthesizer. Since then interest in the new instrument has been remarkable. Pros such as Jimmy Page, Steve Miller and Jeff Baxter himself, have started recording and performing with the Roland GR-500 system. Ed.

This is to be an article about guitar synthesizers in general . . . and the Roland Guitar Synthesizer in particular. Because I play the guitar, I hope to convey something of the excitement I feel about the development of this new musical tool. I have had a Roland Guitar Synthesizer traveling with me on the road for eight weeks, and I have also had time to experiment with it in the studio. I think I have learned enough about it to discuss its advantages and offer a practical critique.

But first let me say that the concept of the guitar synthesizer is definitely one whose time has come. Finally the guitar player has access to the fundamental building blocks of music itself; wave form, frequency, envelope and amplitude. What was once an exclusive part of the keyboard player's sacred world is now available on the fretboard. The adapting of the physics of the guitar to the synthesizer format makes it possible to apply not only the *music* of the guitar but also the *dynamics*. Synthesized music can now be created by the guitarist making use of all the wonderful little subtleties one finds only in the guitar format. For instance, the ability to bend a note. Not just the electronic signal through the use of a manual or pre-set control, but to actually have the *source* at your fingertips is a beautiful thing. Producing vibrato directly from the string into the synthesizer gives the vibrato so much character that one begins to look at *vibrato* more as a noun than an adjective. It allows synthesized music to be played from a guitar player's point of view while also broadening the horizons of synthesized music. Before, keyboard players only had each other to use as sounding boards for ideas. Now keyboard players and guitar players can work together, each contributing something that relates in a unique way to their instrument. It also interjects an additional measure of discipline into the expanding of these horizons by the emergence of basic rules, as these different approaches converge. When I first started using my guitar synthesizer there was a keyboard synthesizer being used in the group. Immediately we found that we had to be more particular about *what* we played, especially when we played *together*. We had to be more aware of such points as where the other player was in the frequency spectrum, and what was being used to synthesize a particular sound. Instead of restricting what we could play, this level of cooperation resulted in our making better music *together*.

Roland has approached the guitar synthesizer concept with the belief that the instrument itself must be an integral part of the system. There has been some criticism that this would force the guitarist to give up his favorite axe in return for the benefits of producing synthesized music. But may I remind the critics that there are very few keyboard synthesizer systems incorporating a Steinway. The guitar portion of the guitar synthesizer is virtually another instrument. Consider, for example, the six-string lap steel guitar, and the multi-neck pedal steel guitar with its pedals and knee levers. They both have certain basic things in common. However, it is easy, to see which is the more "open-ended" system. In short, instead of adapting the synthesizer to the guitar, Roland has incorporated the dynamics of the guitar into the synthesizer.

The Roland Guitar Synthesizer is divided into two main parts; the *electric* guitar section, and the *synthesized* guitar section. The electric guitar section consists of a low-impedance, hum-canceling guitar pickup, with voltage controlled volume and tone controls. Additional equalization is provided by a three-position switch mounted on the guitar, and a fader-type control mounted on the synthesizer module.

The guitar synthesizer section consists of four main parts: the "polyensemble" section, the "bass" section, the "solo melody" section, and the "external synthesizer" section. Each of these sections has its own gain control and on/off switch mounted on the guitar itself. In addition, each section has its own output jack, with switchable output levels, allowing complete separation of each module for multi-channel situations. The outputs of all modules may, of course, be combined into one output jack.

The first module on the guitar synthesizer is the "polyensemble" section. This section consists of variable filters and an envelope generator that work in conjunction with all six strings to produce polyphonic sound.

Next is the "bass" module consisting of percussion controls, a touch sensitivity switch, a switch controlling string decay, and a switch to select the number of strings to be involved in the "bass" section, i.e., strings 5 and 6; 4, 5 and 6; or 1 through 6. The "bass" section has its own envelope generator.

The next module is the "solo melody" section. For all intents and purposes this is a complete synthesizer with the guitar string acting as a VCO. The "solo melody" section contains a mixer to enable you to combine wave forms connected to an LFO and an envelope generator. This section also contains a VCF and a VCA with variable touch sensitivity. In addition, there is an LFO plus an envelope control that can be activated by a remote switch mounted directly on the guitar. The VCA has its own envelope control. A control in the mixer section allows the "polyensemble" section to be processed through the "solo melody" section.

The final module is the "external synthesizer" section. This connects the guitar synthesizer to any external synthesizer, allowing even more versatility in making synthesized music. The "external synthesizer" module has a touch sensitivity switch. A "transposer" transposes any note to a third, fifth or fundamental within a four-octave range, thus allowing the guitarist to alter the overall pitch without changing any pitch settings on the VCO's of the external synthesizer. This section also has a porta-

mento effect control with an on/off switch mounted on the guitar.

There are several other points you should know about. The guitar has a toggle switch for selecting the guitar function only, the synthesizer section only (including the external synthesizer), or both. LED's mounted on each module of the guitar synthesizer and connected to the corresponding on/off switches on the guitar show the player at a glance which modules are in the circuit. And last but not least, the master volume control governs the signal output for the whole system.

Thus far I have discussed the advantages of the guitar synthesizer, its components and how they work together. Now I would like to get into its practical consumer/musician applications by describing some of the ways I have found to utilize this relatively new musical tool. I have had the opportunity of taking this system on the road with the Doobie Brothers for a grueling eight-week tour on the United States, as well as employing it in the studio on numerous occasions. The Roland adapts itself with equal efficiency to either situation. Let me give you an example of its application in a concert situation in which my guitar part consisted of quite a few separate things.

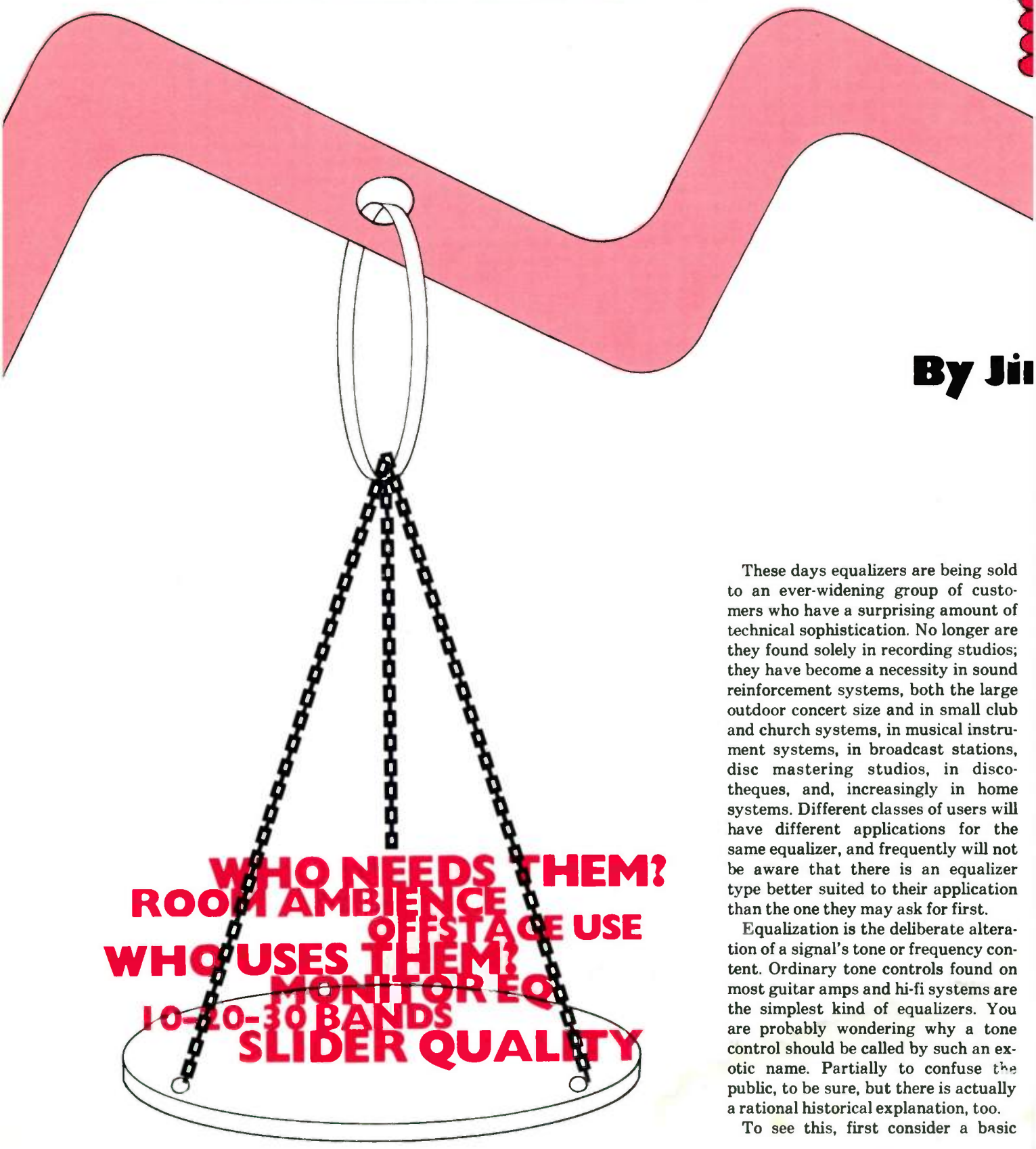
One of the parts I was to execute consisted of playing the top three strings at regular intervals using the *electric guitar* function, while playing a 'baritone sax' part on the bottom two strings. I also had to be prepared to perform a 'cello' line and a guitar solo, to be followed immediately by a 'horn' part consisting of a sustained, three-note chord. I placed the top three strings of the guitar in the "guitar" mode. Then I placed the bottom two strings in the "bass" mode, programmed to produce the attack, sustain, timber and decay of a baritone sax. In this way, I was able to play the guitar and the baritone sax lines simultaneously. With the "solo melody" section preset for 'cello', and the "polyensemble" section preset for 'trumpet' and 'sax,' the Roland GR-500 Guitar Synthesizer made it possible for one person to play all these different parts on a single instrument . . . a feat which could be accomplished on no other single instrument.

In a live situation, this instrument is so versatile it allowed me to play my own parts in addition to covering for a broken keyboard synthesizer. Because the Roland synthesizer is a *guitar* synthesizer, it is capable of producing, in the studio, a sound subtle enough to make a recording engineer mistake it for a guitar in conjunction with a human voice.

My Roland Guitar Synthesizer has been drenched with numerous beverages, carted around the country, inundated nightly with artificial fog, knocked off the stage, frozen for two weeks in a truck in Montana and subjected to the indiscriminate knob twisting of many curious hands. It has survived all of this with minimal maintenance. The guitar itself is a well made, well designed instrument which plays and feels as a guitar should.

I know that I speak for many guitarists as well as myself when I say that it is about time for a good guitar synthesizer. I think that the Roland Corporation has put together one of the finest guitar synthesizers I know, and I believe they will continue to place the latest in synthesizer technology at the guitarist's fingertips. The fun has just begun!

EQUALIZATION



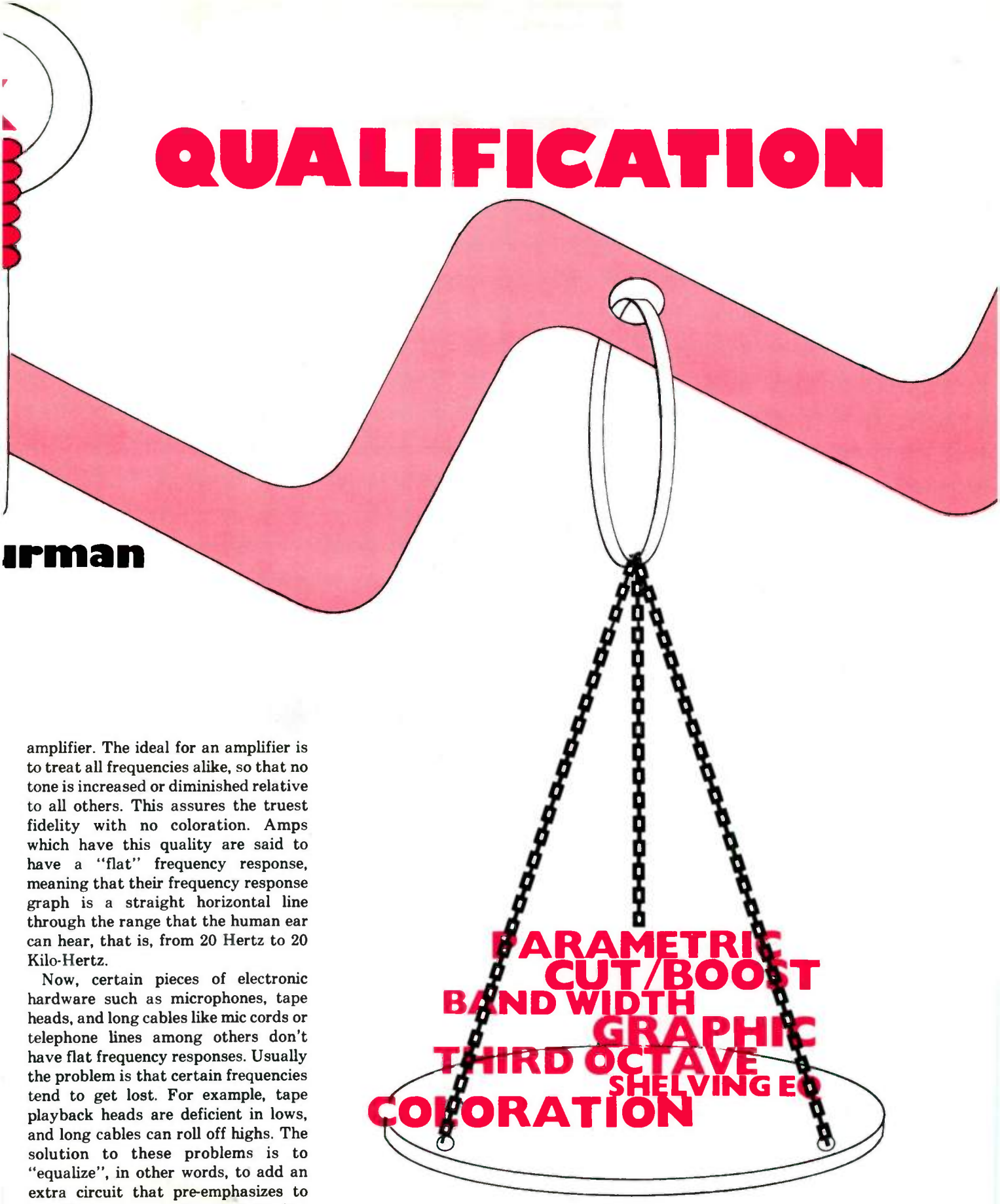
By Ji

These days equalizers are being sold to an ever-widening group of customers who have a surprising amount of technical sophistication. No longer are they found solely in recording studios; they have become a necessity in sound reinforcement systems, both the large outdoor concert size and in small club and church systems, in musical instrument systems, in broadcast stations, disc mastering studios, in discotheques, and, increasingly in home systems. Different classes of users will have different applications for the same equalizer, and frequently will not be aware that there is an equalizer type better suited to their application than the one they may ask for first.

Equalization is the deliberate alteration of a signal's tone or frequency content. Ordinary tone controls found on most guitar amps and hi-fi systems are the simplest kind of equalizers. You are probably wondering why a tone control should be called by such an exotic name. Partially to confuse the public, to be sure, but there is actually a rational historical explanation, too.

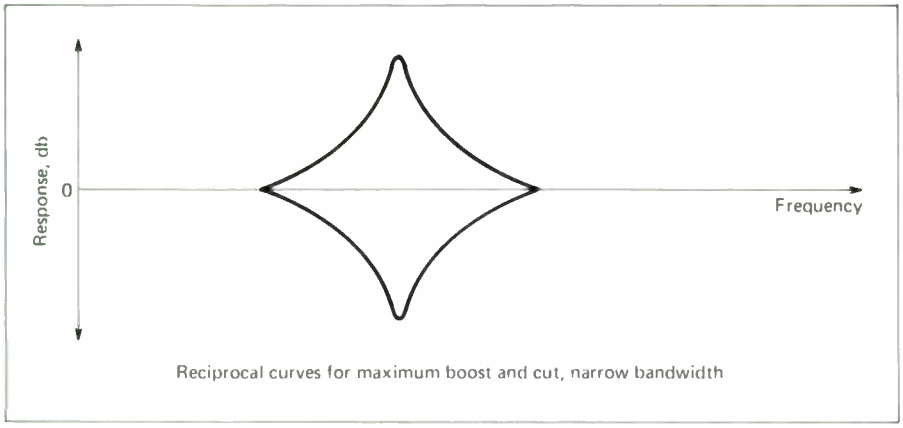
To see this, first consider a basic

QUALIFICATION



amplifier. The ideal for an amplifier is to treat all frequencies alike, so that no tone is increased or diminished relative to all others. This assures the truest fidelity with no coloration. Amps which have this quality are said to have a "flat" frequency response, meaning that their frequency response graph is a straight horizontal line through the range that the human ear can hear, that is, from 20 Hertz to 20 Kilo-Hertz.

Now, certain pieces of electronic hardware such as microphones, tape heads, and long cables like mic cords or telephone lines among others don't have flat frequency responses. Usually the problem is that certain frequencies tend to get lost. For example, tape playback heads are deficient in lows, and long cables can roll off highs. The solution to these problems is to "equalize", in other words, to add an extra circuit that pre-emphasizes to exactly compensate for whatever has been lost. To this day fixed equalization circuits form a fundamental, although seldom appreciated part of every phonograph, tape recorder, and



Reciprocal curves for maximum boost and cut, narrow bandwidth

telephone circuit.

But today equalization has taken on a broader meaning. The word is used to describe any tone changing circuit regardless of whether the purpose is to compensate for some deficiency or simply to create a new sound, one that may not be quite natural but which sounds better to the ears of a musician, engineer, or producer. It is for this latter purpose that equalizers are built into recording and sound reinforcement boards. In addition, there are a number of "black box" type equalizers that are suited for use on stage or which can be used simply as auxiliary gear in a studio, to be patched in where needed. These equalizers are available in several formats, depending on the exact nature of their effects.

The usual bass, middle, and treble controls found on guitar amps are generally inadequate. They affect very large segments of the audio spectrum, so that subtle changes in emphasis are impossible to get. Also, many tone controls of this type don't have any setting which is flat, meaning that every setting has some coloration and you never know what you started with.

A big improvement is an equalizer that offers a greater degree of frequency selection. The kind built into studio boards are often of this type. Typically there will be two knobs for bass, two for midrange, two for treble. One knob of each pair selects a choice of two or three frequencies, while the other knob selects the amount of equalization in decibels. Turn this knob left, and you get a cut; right and you get a boost; center position represents flat response.

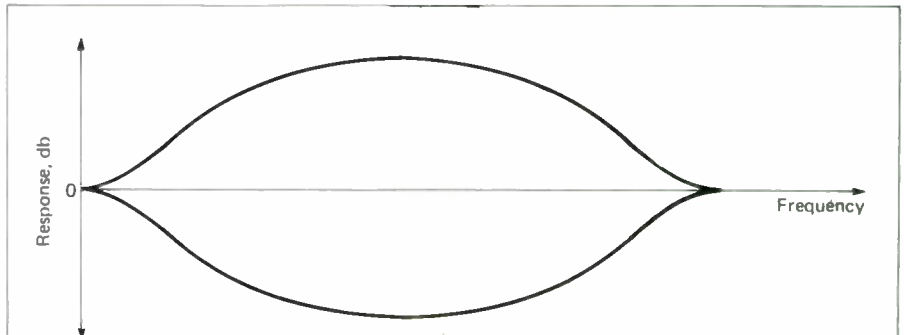
Still more flexible is the "graphic" equalizer. This is the type that has the row of verticle slider controls (although a few have rotary controls). Each slider corresponds to a different frequency, and controls the boost or cut there. Up is boost, center is flat, and down is cut. The sliders on the left-hand side always correspond to bass and ascend through midrange to treble as you move across to the right. Thus, one can get a fair idea of the frequency response curve the graphic is adjusted to with a glance at the positions of the knobs on the sliders—hence the name *graphic.*

The graphic equalizer offers the big advantage of allowing simultaneous control of many frequencies. The exact number varies directly with price. Five slider units are often seen built into guitar amps, small P.A. boards, and even home stereo receivers. Units with ten sliders offer significantly more control. These are often called "octave equalizers" because the audio spectrum comprises just about 10 octaves —one control for each. Units with 20 or 30 sliders (half- and third-octave equalizers respectively) offer progressively finer frequency resolution at correspondingly higher prices.

An important point to consider in buying or selling a graphic equalizer is the sliders themselves. Slide controls are available in a tremendous range of quality levels—from pure junk to carefully designed and built units that will give accurate and trouble-free performance over a long lifetime. One important factor is length of the slide action. Longer throws will allow easier and more precise setting and reading. Note that the slot cut in the panel is frequently longer than the actual distance the lever can move. Some sliders travel as little as 1¼". Also, check the feel of the slide. It should move smoothly with just enough friction so it will stay in place reliably when set. If you find any gravelly or rough spots in the travel, try another brand. The sliders should have some protective covering over the slot, too, to keep dirt out as much as possible. If possible, find out if the sliders are cleanable if they start sounding scratchy. Some types can be ruined if sprayed with the usual aerosol cleaner sprays.

Graphic equalizers find their greatest usefulness in applications where a complex response curve must be created. This need most often arises in room equalization, that is, compensating for the complicated pattern of response peaks and dips caused by the geometry and pattern of absorbing and reflecting surfaces found in living rooms, concert halls, and studio control rooms. Then too, any user who enjoys the unique response-curve-at-a-glance feature will value the graphic for any equalization use.

Perhaps the most flexible equalizer of all is still another kind called a parametric, because all the "parameters" of equalization—frequency, amount of boost or cut and a third factor called bandwidth—are directly adjustable. With a parametric equalizer, you are not limited to a choice of certain specific frequencies. The frequency controls are *continuously* variable, meaning that you can equalize the in-between frequencies, too. For example, one can zero in on exactly those frequencies which provide the greatest improvement in the sound. This "zeroing-in" feature is also important



Reciprocal curves for maximum boost and cut, wide bandwidth

If Akai, Aiwa, Centrex, JVC, Kenwood, Meriton, Nakamichi, Optonica, Pioneer, Royal Sound, Sansui, Sharp, TEAC, Toshiba, Uher and Yamaha, in addition to Bang & Olufsen, Dual, Fisher, Harman-Kardon, Lafayette, Sankyo and Tandberg all recommend our SA for their machines...

...shouldn't you?

All of the tape deck manufacturers above recommend our SA cassettes for use in their machines in the "High" or "CrO_2" bias position. In addition, all those in the first part of the list clearly indicate they prefer SA, since they bias their decks specifically for it.

So it only makes sense, when you sell one of these machines, to make sure you sell a case of TDK SA cassettes to go along with it. Especially, since tape sales are so profitable, and take up so little sales floor and warehouse space.

Our barrage of consumer advertising in over two dozen national publications, on syndicated radio shows on 225 FM stations, plus co-op local radio and print, helps you sell by telling millions of potential customers that TDK SA is "The Machine for Your Machine." All this, plus the extra help we give you with a full complement of in-store P-O-[illegible] and [illegible]motional aids.

Selling a tape deck without selling tape to use in it, is like selling a camera without film. Put extra profits into your component sales by selling the tape that's recommended for so many leading tape decks: TDK SA. It's only one product in the trend-setting TDK full line of cassettes, open reel and eight track tapes.

And if you have any doubts, all you have to do is check our references.

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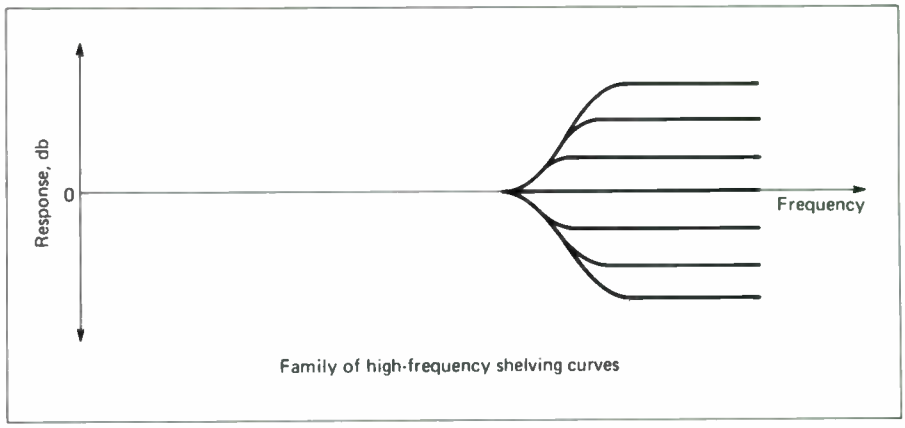
for eliminating unwanted sounds, like hum or feedback. It also makes certain special effects like wah-wah or phasing possible, by "sweeping" or rotating the frequency control back and forth in boost or cut settings respectively. These effects would be impossible to produce with any equalizer type other than a parametric.

Most parametrics are either three, four, or five band units. Each band allows one frequency to be controlled, so a three-band unit can control three frequencies simultaneously. If there are three bands, there are three frequency, three bandwidth and three equalization controls. The ranges covered by the three frequency controls are not the same, but are staggered so that the first frequency control covers the bass portion of the audio spectrum, the second covers the mid-range, and the third covers the treble. There should be a good deal of overlap on these ranges, so that most frequencies can be controlled with either of two possible bands. Whichever band is more convenient is used.

Frequency and amount of equalization alone, however, are not enough to completely determine the response that an equalizer will have. For boost and cut effects do not apply at precisely one frequency only, but rather apply to a range or *band* of frequencies, with the specified frequency at the center. It is the function of the bandwidth controls to set the range.

A useful range of bandwidth settings might allow variation from about ⅓ octave up to several octaves (with the extremes of the band being defined as the points 3db below maximum boost or above maximum cut.) Some parametrics offer symmetrical, or reciprocal boost-cut curves. In others, for a given bandwidth setting, the cut curve is narrower than the corresponding boost curve. (In this type of equalizer, technically speaking, the bandwidth for boost and cut can be considered the same if a different definition is used for the cut curve: The extremes are the points 3 db below the zero line instead of 3db above the maximum cut.) The reciprocal curves are most suitable in applications where a precise boost is added in a certain stage of work and later on an exactly matching cut is taken to negate the effect. This requirement comes up occasionally in disc mastering but rarely in other applications. For general musical use, the non-reciprocal curves are more useful because of their ability to attain very narrow cut curves ("notches") which are ideal for removing undesirable sounds which are primarily limited to one frequency, such as hum and feedback, in the process leaving the music with the minimum possible coloration. A correspondingly narrow boost effect is musically pretty useless.

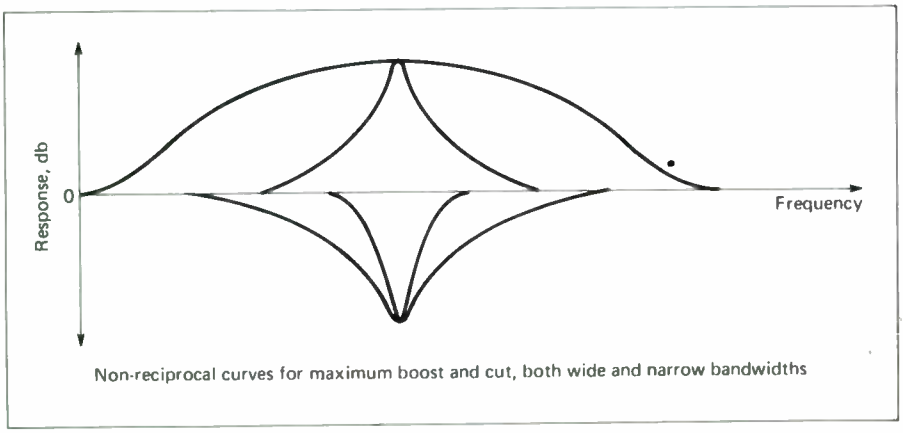


Family of high-frequency shelving curves

Graphic equalizers have bandwidths too. Ideally, the bandwidth should exactly correspond to the frequency separation between one slider and the next, so that there is absolutely no interaction. Unfortunately, this ideal is impossible to reach in a practical equalizer, meaning that a boost at one frequency will also boost to some extent at the adjacent sliders' frequencies. So, when shopping for a graphic, check the response curves to make sure there is neither too much nor too little overlap. The bandwidth for each slider on a graphic equalizer will typically be fixed at some in-between compromise between the narrowest and broadest settings available from a parametric.

A few equalizers offer shelving equalization. Usually this is an extra feature on a unit that is basically graphic or parametric. Since shelving equalizers apply equal boost or cut at all points, but only one side of the selected frequency, the concept of bandwidth is inapplicable here. This feature can be very handy for large-scale cutting, but should be used with great caution for boosting because of its potential for increasing undesirable sounds like hiss, rumble, footsteps, etc. which lie near to or beyond the 20 Hz and 20 KHz extremes of the audio spectrum.

One final point to consider in evaluating any equalizer is the amount of boost or cut available, in decibels (db). Extremes of ±12 db are most common; ±15 db is better; and ±20 db is best because it allows that much more flexibility. Some manufacturers avoid providing this much boost-cut range because it is less foolproof, in many, but certainly not all circumstances, a 20 db boost will cause clipping. Whether it does or doesn't depends on the signal level and frequency content, and the available headroom. So much boost would rarely be used in recording situations, and might be asking for trouble in a home stereo system, but it is *very* handy in musical instrument systems. Musicians are well-known for turning their tone controls up to maximum in the search for an individualistic sound, and they should have just as much as



Non-reciprocal curves for maximum boost and cut, both wide and narrow bandwidths

Our point of view: You can't know too much about a good thing. Number 31 in a series of factual discussions.

audio-talk from audio-technica®

MICROPHONES, The First Basic Term: Dynamic.

With all the high-end tape recorders being sold, your customers will need good microphones if they are to get full value from their purchases. But how can you tell good microphones from lower quality units? And how can you help your customer choose the best microphones for his needs?

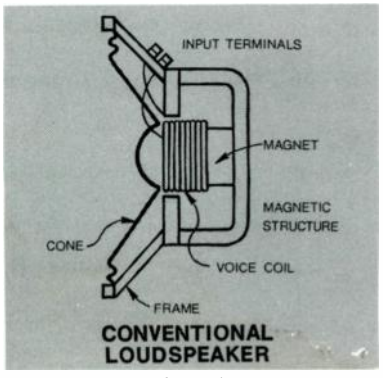
THE 8 BASIC TERMS

Well, first you should know the basics about these hi-fi components. Much of what you already know about hi-fi equipment can be applied to microphones. And if you understand just *8 basic terms* you are well on your way to becoming a microphone expert. These key words are: 1) Dynamic, 2) Condenser, 3) Omnidirectional, 4) Unidirectional (or cardioid), 5) Proximity Effect, 6) Feedback, 7) Impedance, and 8) Sensitivity. Most should already be familiar to you, so let's see how they apply to microphones.

We'll start with how microphones work. Just like loudspeakers, headphones, and phono cartridges . . . microphones are transducers. Microphones convert acoustic energy (sound) into electrical energy. While there are many ways to do this, two are most used for high fidelity recording: *Dynamic* and *Condenser*, and we'll discuss these in most detail.

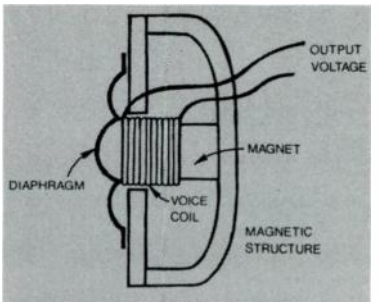
THE DYNAMIC MICROPHONE

If you compare a drawing of a typical dynamic microphone with a loudspeaker you'll see many similarities. With both there's a diaphragm (or cone) with a coil of wire attached. The coil is in a magnetic field and the diaphragm and coil can move. But with the loudspeaker, the cone moves when current from the amplifier flows through the coil. It's a result of the interaction between the magnetic field of the speaker's magnet and the magnetic force created when current flows through the coil.



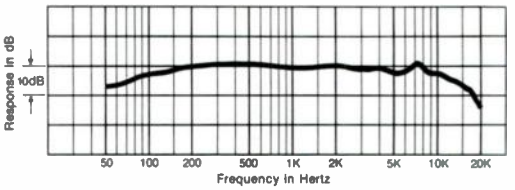
CONVENTIONAL LOUDSPEAKER

With a microphone, both the diaphragm and voice coil are small and light. Thus, they are easily moved as sound pressure changes. Current is generated in the coil as it moves through magnetic lines of flux, creating a small output voltage. You might consider a loudspeaker as a motor and the microphone as a generator. Both are very similiar devices, but used for opposite purposes. In fact, some intercoms use small loudspeakers as *both* microphone and speaker, simply switching the same transducer from one end of the amplifier to the other for talking or listening. The very finest dynamic microphones are much more than "little speakers" however. Their sophistication of design and construction makes them capable of outstanding performance in critical applications.



DYNAMIC MICROPHONE

Because of the inherent simplicity of the element, dynamic microphones are renowned for ruggedness and reliability. They need no external power sources. Output levels are high and work directly into almost all microphone inputs to provide excellent signal-to-noise levels. They need little or no regular maintenance, and with reasonable care will provide unchanging performance for years. Frequency response can be controlled in design to provide extremely wide, smooth output over the entire audible range, or "tailored" in response for special applications. While dynamic microphones are specially suited to beginners in recording. they are also the first choice of professional engineers for many recording applications.



FREQUENCY RESPONSE, MODEL AT-802

Proof of the smooth, wide range capabilites of good dynamic microphones is the response shown for the Model AT802 moving coil dynamic. With performance like this, you can safely recommend a dynamic microphone for just about any kind of recording. And because of the robust construction, dynamics are ideal for popular music recording and sound reinforcement as well as for general P.A. applications.

In our next issue, we'll concentrate on a relatively new form of microphone construction, the *Electret Condenser*, our next basic term.

Jon R. Kelly
Vice President & General Manager

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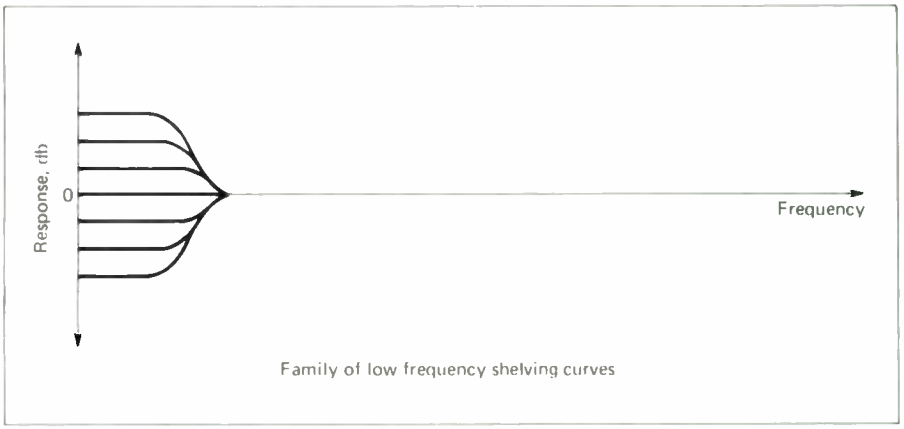
they want. In any case, in the hands of a professional sound person the greater range can only be an advantage. If extreme boosts cause distortion, the signal level driving the equalizer can always be reduced elsewhere to allow the necessary headroom.



Family of low frequency shelving curves

Some parametrics provide even more cut than 20 db. Of course, cutting is inherently more foolproof than boosting because it never leads to distortion. Infinitely deep notches (meaning complete silence at the selected frequency) are available on some units and are a feature worth looking for.

Equalizer applications fall into two major categories, general purpose and compensation. General purpose use means use wherever you would use tone controls to make sounds more pleasing to the ear. Most recording studio equalizers use falls into this category. A track may need more presence or brilliance in order to sound good to the producer. Two similar-sounding guitar tracks may be EQ'ed differently to heighten the contrast between them. An extra-loud electric piano note may need to be suppressed. A mix aimed at the disco market may need larger-than-life bass, and so on. If an equalizer is to be patched in where and when it is needed, a parametric is probably the best choice in this application because of the greater range of effects it can provide and especially because of its "zeroing in" feature. If separate equalizers are to be provided for each track in a studio board, simpler types may suffice for routine use but at least one parametric should be available when the simpler units can't get the necessary effects.

Equalizer use for compensation encompasses room equalization in control rooms, discotheques, and homes, as well as concert situations in which live microphones are present. If a room can be adequately analyzed with suitable test equipment (this usually requires a pink noise source, calibrated mic, and real-time spectrum analyzer), a third-octave graphic can generally do the best job of matching the complex of response peaks and valleys, yielding the closest approximation to the idea of flat response. This is most practical in fixed installations. On the other hand, on-the-road concert users may find parametrics more suitable for compensation where feedback in the monitor system is the greatest problem. (A number of small PA boards include a 3 to 5 knob graphic-type "anti-feedback" equalizer. These are virtually useless, though they may be better than none at all.) The parametric's big advantage here is its narrow notch, which can be zeroed to an exact match with the feedback frequency. Once this is done, the monitor volume can be advanced until feedback again occurs, this time at a different frequency. Then, another band can be used to suppress that frequency as well, and so on. Ultimately the monitor volume can be made dramatically louder before howling. This method also offers the advantage of not requiring any particular test equipment.

Perhaps the most demanding equalizer application is in disc mastering, because of the need for resettability. There the most expensive and sophisticated equalizers are found. The "continuously variable" aspect of the parametric may be a disadvantage here, and stepped and detented frequency and equalization controls may be necessary.

That's what Kerry Livgren of Kansas asked as he sat behind the keyboard of the new Korg polyphonic synthesizer. It didn't take long to transform Kerry from questioner to convert. Because after a few minutes of playing he found a lot more than the portamento.

Kerry discovered features he had never seen in any synthesizer. Like a totally polyphonic keyboard with each key triggering its own dynamic filter, envelope generator, and VCA. Polyphonic voltage-controlled sample and hold. Six different simultaneous modulations. A programmable wheel, joy stick and transient controllers. And a patch panel that performs musical miracles.

In fact, Kerry liked Korg so much he used the synthesizer on the Kansas album Point of Know Return. Here's what he had to say after the recording session. Take it Kerry.

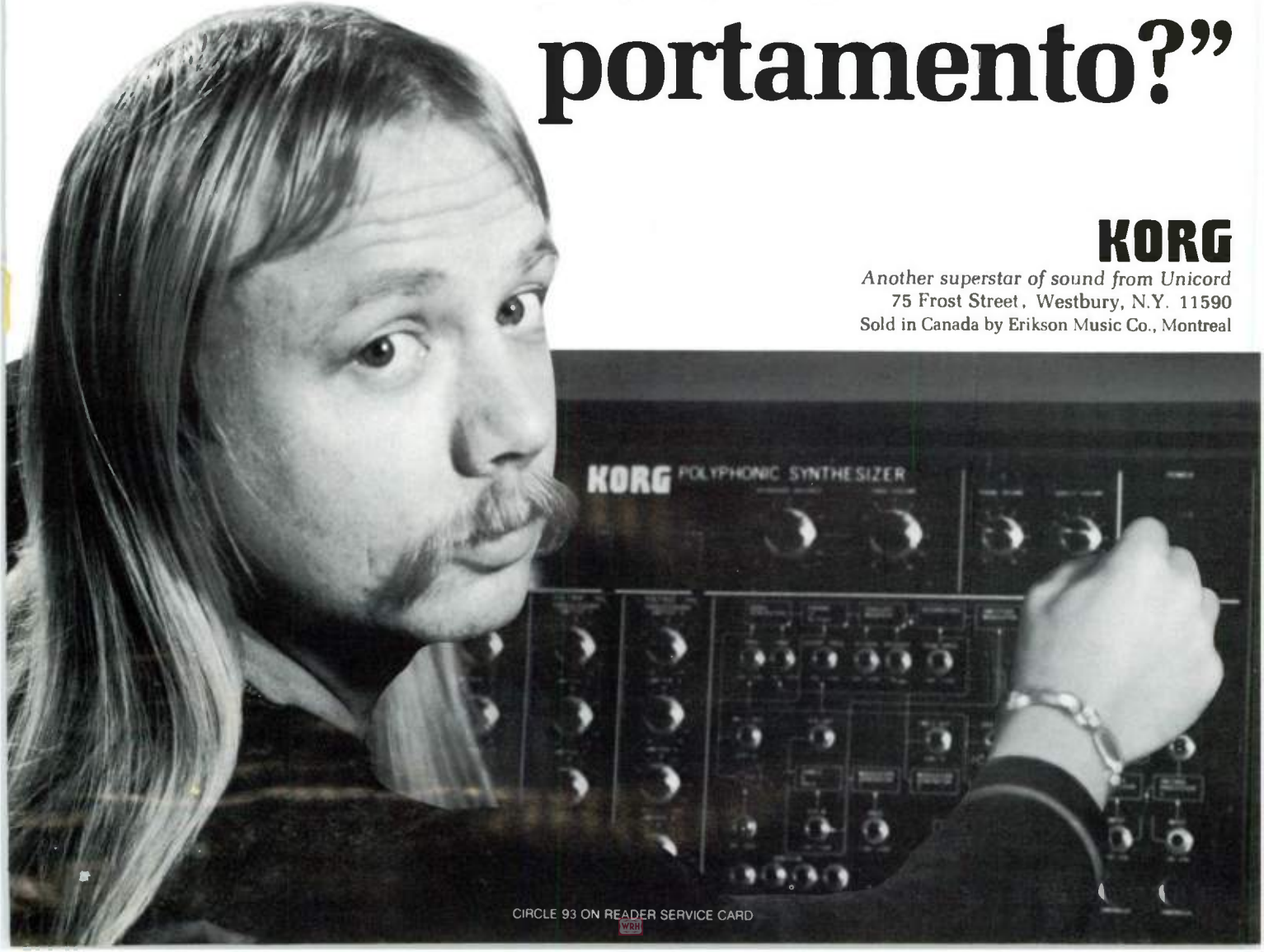
"A New Ax Every Ten Seconds."

"The Korg Professional Laboratory Systems synthesizer is in a class by itself. It sounds totally different from any synthesizer I've ever played. It's like having a new ax every ten seconds."

There are 10 Korg models of exceptional value, ranging from the Professional Laboratory Systems group to the new Micro Preset, which puts 315 instant voice combinations at your fingertips.

Whether you're looking for a preset, a polyphonic, or a dynamic portamento, you'll find a Korg synthesizer that meets your exacting requirements.

"Where's the portamento?"



KORG

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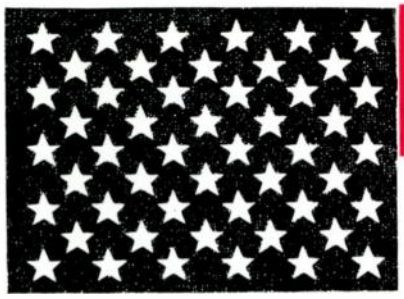

FINANCE AMERICA

A National Lending Program for Musicians

By Ken Sacks

Ken Sacks is National Sales Manager for the TEAC Tascam Series.

The recently announced finance plan, hammered out over a period of many months by Tascam personnel and the FinanceAmerica lending arm of Bank of America, points up a nagging problem that has plagued musicians and aspiring professional recordists for a very long time. The plan, by which musically oriented consumers can purchase Tascam series equipment on approved credit through regional offices of FinanceAmerica, is the first of its kind devised for the recording/music/pro audio industry.

Until now, a young musician struggling to create his music, attempting to record his musical offerings, or simply wanting to "get it on" as a serious recordist had to sign his house away, or his first-born son, in order to qualify for a loan. I've talked to many dealers who sadly related lost sales caused by a lack of specialized credit reference for their customers.

At one time—and I'm being only partially facetious—musicians, writers and preachers were automatically shunned by lending institutions because their collective ability to pay off a loan was less than what was considered a "good risk" venture. A young musician can work a 9 to 5 job and play on weekends with other young musicians. Sometimes referred to as the "Holiday Inn Circuit," these young artists simply didn't have the money to be able to buy even moderately expensive recording gear. And the cost of using existing studios was completely out of the question.

Even though TEAC Tascam Series has made it a fundamental premise of its marketing program to provide this rather sophisticated gear—elaborate mixers, recorder/reproducers and signal processing devices—at an extremely alluring, affordable price, that price was nonetheless beyond the reach of many young and not-so-young musicians. We prided ourselves on having established a product line that dramatically reduced the cost of using this kind of equipment. Still, dealers related stories of lost sales, tales of young people who would drool over the gear and then be forced to turn away empty-handed because of the lack of financial stature.

So this program has a two-fold advantage. First, it creates an attitude of pre-acceptance on the part of one of the nation's largest lending institutions to *helpfully* assist the new professional consumer. Secondly, it provides the dealer with a startling new means of sell-through. And, of course, we at TEAC Tascam Series hope it will enable us to sell a lot more Tascam gear.

Listen to Ken Evans, regional senior vice president of FinanceAmerica, as he elaborates to the managers of selected regional branches which will handle the Tascam Series' credit applications:

"In the early to mid-sixties, such equipment (as Tascam) was used primarily in highly professional and very expensive studios. The would-be musician couldn't afford to use such equipment, hence, he had to wait for a 'break' whereby he could then 'cut' a demonstrator record at a large studio.

"TEAC recognized the need to supply equipment for the semi-professional musician and even the professional who could work in his own home, create music any time of the day or night, and, hopefully, prepare finished masters for presentation to record companies. Then TEAC developed the original Tascam product line, whose concept of these mixers and recorder/ reproducers was to put in the hands of would-be musicians moderately-sophisticated and affordable equipment to do the job. For the first time, an aspiring musician could have a professional mixing console as necessary to his creative fulfillment."

"TEAC, then, deliberately sought out the 'music instrument dealer' to sell Tascam products because his customers were buying the tools of their trade from him. This new market is now beginning to spiral dramatically. In view of this, TEAC's management has asked FinanceAmerica to incorporate its creativity and resources with TEAC's to come up with a retail finance program which will be realistic to the needs of their dealers and professional consumers."

That's the concept in a nutshell. The key to the whole thing is the credit application, which, in addition to obtaining the standard credit data, asks the prospective buyer to list his agent or personal manager, union card number, and his bookings. This information allows the FinanceAmerica branches to verify the data to more accurately guage the real total income of the potential customer/musician. FinanceAmerica also cautions its branches to remember that booking data is essential because most customers are fully employed, holding low income jobs during the time they are not performing, and this income amounts to only ⅓ to ½ their gross earnings for the year.

A separate section of the application is devoted to the studio operator, who is not a professional musician necessarily, but a technician who obtains Tascam equipment for the purpose of recording for musicians for a fee. Be-

cause, in the words of Finance-America, "this individual most likely derives a greater portion of his gross income from this operation, he will have to supply either a W-2 or an audited financial statement."

Now the musician can buy the equipment at a rate commensurate with his income to create music, to record, and to use in playing his gigs.

The terms require a 10 percent add-on in all states where this is permitted by law and an add-on as close as possible to 10 percent in the others. Down payments are 10 percent up to $6,000, and 20 percent over $6,000. Terms are generous: 36 months up to $2,000; 48 months, from $2,001 to $6,000; 60 months, $6,001 and over.

The plan has been greeted warmly since its announcement. Dealers, in particular, are delighted that such a plan has finally been put together that will help them increase sales. Obviously, they feel that a *growing* market for this kind of product does, in fact, exist and they naturally want all the help they can get in providing the best possible climate for each sale. Already sales have been made, according to the reports we have received from the field, on the strength of this plan.

I am personally pleased with the dealer reaction because it relates directly to the relationship that exists between dealer and manufacturer. A manufacturer who cares about his dealers and actively participates in providing a good climate for an increase in sales will be successful.

We feel it is the beginning of a new era for music production in this country. At a time when nearly 80 percent of the music produced is electronic, as opposed to acoustic, it is only fair, it seems to me, that those endeavoring to practice their art, and eventually to make a living from it, have at their disposal a proper credit plan for which they can apply.

This particular marketing area has grown tremendously and is still expanding as more and more people get involved in music as a way of life and a serious avocation. Our research at TEAC Tascam indicates the growth will continue as more companies produce a wider range of newer sound equipment and a broader range of musician instruments.

The future looms bright . . . especially for the new semi-professional/professional consumer.

By Peter Starr

PHOTOS BY
NEIL ZLOZOWER/MIRAGE

DEALER DOSSIER

West L.A. Music
Los Angeles, California

Los Angeles, California is certainly one of the major contemporary music centers of this particular planet, and serving the L.A. professional and semi-professional musician for the last 11 years has been West L.A. Music, located at 11345 Santa Monica Blvd. Founded by budding business major/guitar teacher Don Griffin and drummer/teacher Ed Swansey, this outstanding musical and audio equipment outlet still stands in its original site—save a few rennovations and new space aquisitions. In 1970 Griffin aquired sole ownership of the business and he incorporated the dealership, thus West L.A. Music Inc. Today management of the West L.A. music center is the responsibility of Greg Hildebrand, who verbally painted the picture of one of the country's largest dealerships in professional and semi-professional musical instruments and audio equipment.

Describe the various aspects of West L.A. Music.

Hildebrand: Right now we are basically structured with one major retail operation but we are in the process of trying to expand rapidly into several other west coast locations. Basically, the key to expansion is in getting the right people. We are constantly looking for people who want to become managers of their own stores. We have one basic location which is the original store. That site became too small for our operations so we added a second location two doors down. That is our Pro-Sound Division which came about primarily because we were starting to deal with a lot of professional musicians whose needs were growing beyond just electric guitars and amplifiers. They expressed a concern of being able to get into some of the home recording equipment. We began looking for product lines that were more specialized than stereo stores, more professionally oriented. Consequently, we became the first TASCAM dealer in the country. This is before TASCAM was absorbed by the TEAC Corp. They had been promoting their product as a home recording studio product but they were very leary of opening a music store as a dealer because the whole premise of the line was that it had to be understood and they didn't feel that music stores in general could understand their products and sell them properly.

Was it that contact with professional musicians you had that TASCAM wanted to develop?

Hildebrand: That's right! Before that time they were selling their product strictly through pro-audio and studio suppliers. We became one of their largest dealers in the country right away because it was so natural joining the professional musician with that type of home recording capability. The TASCAM line has really flourished and we're still one of their largest dealers.

Do you only deal with TASCAM products in the pro-audio division?

Hildebrand: No, though that was pretty much the start of it. From there we branched out into all the other miscellaneous products that go along with the system. Our whole philosophy has been one of educating our sales force and then educating the consumer. We got into the JBL Professional Series PA products because our customers knew there were products out there that could achieve the type of performance they were looking for. But they didn't really have access to the product line, and they didn't feel comfortable with the items,so *that* group of consumer's was looking for *someone* who could take all of the available products and put them together so that the sound 'package' would be easy to operate and take on the road and provide maximum performance.

So you can put together systems that are specifically designed for the road?

Hildebrand: Absolutely. Right now we do every phase of the performing

musicians' tour packaging. We are equipped to custom design systems from the ground up for any particular group and we deal with most of the major rock groups, recording artists and record companies in Los Angeles.

Do you send your own crews out on the road to manage these systems?

Hildebrand: No, we don't. We've found that it is very hard to get the type of person in this business who is stable enough to go out on the road for long periods. We let the customer get their own people and what we will do is package the equipment so that it is easy to operate. Before the customer picks it up we set it up in our Pro-Sound Studio and go through it with their people and educate them about the important *dos* and *don'ts* involved with the package.

Are you involved on a strict sales basis or are there other options?

Hildebrand: We do large scale rental systems on occasion but that's not our main thrust of business. We've sold to most of the companies that are doing sound rentals, and again, the number of people needed to do something like that properly is pretty astronomical. We have supplied various TV shows with equipment on a rental basis. One of the reasons why we deal with record companies and do tour packaging is because we are one of the only retail outlets that does leasing for tours. A record company is normally the one in the long run who picks up the tab for large scale rock concert tours. They're more concerned with the show that's on the road, so they were going out and either buying or renting the equipment. These companies would rather rent the equipment than buy it for tax purposes, so we got into leasing equipment for tours and one of the big advantages is that the group has *brand new* equipment when they go out.

What happens to the equipment once it comes back from a tour?

Hildebrand: It depends on how the lease is set up. A lot of leases are set up so that the group can buy equipment at a greatly reduced price. If they don't want it, we'll take it back and if it is in usable condition we'll fix it up and warrant it ourselves and sell it as used equipment.

West L.A. Music has a large list of manufacturers that they represent. How do you determine which companies you will represent in your store?

Hildebrand: What we're looking for in companies that we deal with is that they have stability in the music business. In the music business there are an incredible number of people who go in and out of business very quickly. A company can start manufacturing a guitar amplifier and a year later they can be bankrupt. Our whole premise is that we have to be able to service what we sell. For this reason we opened our own service center largely because we were constantly running into a problem. We couldn't find existing service centers in the Los Angeles area who were willing to deal with warranty repairs. The equipment that we sell has very good manufacturer's warranties but they aren't worth anything if you can't find someone who is willing to repair equipment under a warranty. So we opened our own service center specifically as a service to our customers, to do warranty work. We had quite a bit of success with it but the problem was with the paper work. We would eventually be reimbursed by the manufacturers but the time involved with paper work was getting to be considerable. So we found a gentleman who had purchased his own service center in West Covina (a Los Angeles suburb) and he purchased our service center. The man's name is Dale Fryburger and his store is called Musicians Service Center. They do a large portion of the warranty repair work for the entire Los Angeles area. The service center is now located about a block from our main store and the back half of the center is our warehouse facility.

Let's talk about the musical instruments section of the store.

Hildebrand: The actual music store is located in the original West L.A. Music facility. We carry every major brand of guitar, amplifier, and piano. We deal primarily in portable, professional equipment. We normally stock very few Japanese imitations, the reason being that our market is more professional in nature. At Christmas time we do carry a limited number of Japanese products for the seasonal market. Right now about 50% of our business comes from the MI (musical instrument) division and *that* is increasing.

From your view point, what kinds of equipment are the professional and semi-professional musician buying? Are there any new trends in instrument and equipment purchases?

Hildebrand: Well, from the professional level there are several categories. We do quite a big business with major performing and recording artists but it is also limited to how many of them are out there on the road. The large scale rock concert business is dying rapidly and has been for quite awhile because of the expense of putting on these extravaganzas.

Are you noticing that reflection in terms of your own business?

Hildebrand: Absolutely. The type of equipment that was popular three or four years ago from a guitar standpoint were the large stacks of amplifiers and that type of thing; Marshalls, Sunns, etc..I think that within the last three years the live thing has been turning around a bit, centering more towards the club-working musician. The cost of putting on a large scale concert is becoming prohibitive. We've noticed that there has been a large

amount of trading in of large guitar amp stacks, even by the very big professionals. What they're buying is the smaller, self-contained powerful guitar amps that are compact.

What other areas of professional service does the Pro-Audio division get involved with?

Hildebrand: We have also gotten strongly into the studio business. Not so much in selling studio boards, but mainly synthesizers. We are a very large dealer on the west coast in synthesizer equipment. Very often our salesmen will go out and set up synthesizers for studio gigs.

Let's talk about some of the home studios your Pro-Audio division sells.

Hildebrand: Instead of going into recording studios and spending up to maybe $100 an hour for that studio time, a musician now has the capability of making good quality demo tapes for a basic complete price of five to ten thousand dollars. The most popular system is the TASCAM series Studio 8000, which is a home eight track recorder. This is combined with a TASCAM Model 5 mixer and then there is the 25-2 mixdown recorder. With that the consumer/musician can make extremely good home demos. It also makes it very inexpensive to record practice sessions.

Can your people also help these customers properly design studios in their home once they've purchased a home recording system?

Hildebrand: Absolutely. We are not trying to design complete studio mastering systems, but we will consult with the customer and educate him on how to best use their newly purchased systems.

Since you offer so many special services to the professional and semi-professional musician how do you go about promoting the store and all it's functions?

Hildebrand: Well surprisingly, when we first started out we used every sort of advertising possibility available. In 1972 and 1973 we took spots out on radio, television and in all the local newspapers and a lot of magazines. We were the first music store to offer these products to the musician. Our major form of advertising for the pro division was by word of mouth more than it was of any particular media advertising. I don't think you can really do a media advertising campaign on professional audio equipment products on a retail level. The word just gets around. There is a need for a certain amount of hype involved with a music operation. But when you get down to brass tacks and you're talking big dollars and performance with a group, the hype ends and you really have to convey to them that you know what you're talking about. Another point to relate is that you will stand by the products and make sure they are adequately serviced whenever necessary. You have to let the musician know that you intend to stand by the product and your service.

How does your store decide upon which new products from manufacturers to carry?

Hildebrand: If a manufacturer comes to us and asks us to carry his product, the first thing that we will do is get a sample of it and evaluate it ourselves. We judge a product on reliability, features vs. performance, features vs. cost, etc.. There is a need sometimes for products that are more affordable but are simply a reasonable facsimile as far as performance goes, so we might carry a line even though it isn't the most expensive and the specs are not the absolute best. We take all of these things into consideration and one of the most important factors that we evaluate is whether there is a need for *this* product. Unfortunately, a lot of manufacturers are pretty isolated with what they are doing and there is an incredibly large number of products that come out that may be an engineer's dream, but are not that practical to use while performing.

What sort of test do you come up with to determine practicality?

Hildebrand: The biggest test that we usually do is to lend new products to a lot of our better customers. We ask them to take it out on the road and use it and tell us what they think. We have gained enough market strength so that most manufacturers are willing to work on this level with us because they would rather know of any shortcomings before they produce 1000 units of something.

What happens if the tests are positive?

Hildebrand: If the line has some definite merits we will place the order and get the product in. When the product comes in, the first thing we do, before we even put it on the floor, is conduct staff seminars. We have seminars on the average of three days a week for different products and different state-of-the-art techniques; all different phases of the product itself, how to use it, what it does and what it

doesn't do are discussed.

Does the entire staff get involved in these seminars?

Hildebrand: Yes, we have 12 people in the store, plus five in the corporation. Basically what happens is that the product specialist or the rep will come in and make a presentation to the staff. In addition, we continuously have the manufacturers come in and bring the staff up to date on the latest techniques and products from their company. This also lets newer members of our staff become familiar with all the different products. A big part of our business is in training the sales staff to be familiar with the product. I might add that the key that I look for in a salesman is that he or she is able to communicate with musicians and be able to understand the products. We are getting into a very technical end these days.

What happens once the sales force is trained?

Hildebrand: We take the product and put it on the sales floor. We have a computerized mailing list that has everyone who ever purchased anything from us on it, plus all the musicians in the musicians' union in Southern California. We divide the list into sections and then we mail out invitations, inviting them to come in and see one of our clinics and demonstrations of new products. These clinics have become so successful that we pretty much have to do it by invitation only and take reservations. We just don't have the room for the number of people that usually try to reserve space. Part of our reputation is based on doing large presentations of new products on a continuous basis. People keep expecting us to keep introducing new products.

So West L.A. Music offers services that most other companies might not have either the money or the time to get involved with.

Hildebrand: That's right. It's very expensive to introduce new products. It's probably the most expensive part of a business. Let's face it, it's very easy to sell a product that someone comes in and asks for. When you're introducing a new product, or new product concept, you don't have that track record to work with. Basically you are buying a product or product line (you're talking about expensive items) and then exposing it in order to create a demand and then be able to fill that demand. That can take several months to achieve. The professional

division is the largest part of our business and it is that division which separates West L.A. Music from the rest of the stores in the area.

How frequently do equipment 'fad' changes occur in your pro-audio division as opposed to your musical instruments division

Hildebrand: The pro-division is considerably more stable than the musical instruments division. The pro-division's market is more stable than that of a guitar amp market because the parameters for a PA system, for example, are much different than a guitar amplifier. A guitar amplifier, from a technical standpoint, doesn't make a whole lot of sense in that it is designed to distort, and it is designed to break up sound, and to sound very raunchy to give you that good type of

rock and roll tone. A PA system is designed to be very accurate in reproduction. The problem is that the amount of distortion and raunch in amplifiers is very subjective with musicians. It depends on the musicians' particular style and taste. Musicians need change and so the equipment has to change with the needs.

Let's talk about the current methods of advertising your musical instruments division.

Hildebrand: Our biggest form of advertising at the moment is the classified section of the newspapers. We find that most musicians are pretty acclimated to looking in that particular section of the paper. We do weekly advertisements in the classifieds of the major local newspapers, and we primarily advertise new products. Again, our philosophy is to expose the public, educate the public, and sell them a good-valued product. You really don't need high-pressured sales tactics from this point of view. At this moment we are relying heavily on our mailing list, which is probably the most important form of advertising we have.

Some stores encourage people to come in, browse and generally just hang out, and by doing so perhaps some of them will purchase products. How does West L.A. Music feel about this philosophy?

Hildebrand: We have an entirely different philosophy from stores who encourage that. I'm not saying that one attitude is necessarily right and the other wrong. What is right for us is not necessarily right for another store. From our standpoint, we try to make the customers feel comfortable and we don't want them to feel that they are being pressured. We do discourage the casual browser who just wants to plug in guitars and play, it can drive the dealer crazy. It gets to be very noisy if 14 people are sitting in a room playing guitars, you also end up with a lot of used and beat up equipment that way. We do discourage people coming in and *jamming*, but we do encourage people coming in and looking, perhaps getting together with a salesman to talk. You don't have to buy something to come in, but at the same time we're just not set up to hold jam sessions.

Can you recall any interesting stories about any of the successful professionals you've dealt with?

Hildebrand: Well, we deal with just about all the pros here. Stevie Wonder is one that comes into the store quite often. He's in a position where he can afford to buy just about anything that comes out so he pretty much will take one of everything that comes out and is new. He has a sales consultant in the store, who is also the store manager, and he deals with Stevie on a one to one basis. One of the last times Stevie was in was on Christmas Eve and he used our phones to call all of his friends and find out what they wanted for Christmas. He made up a shopping list from our store!

How does the store compete nationally?

Hildebrand: Well we do get a large amount of national business, though we have never advertised nationally and we don't have a mail order catalog. A lot of the big groups who use our services provide word of mouth to other musicians. When our L.A.-based groups are out on tour, we will ship them equipment anywhere in the country. A common example is that a group will be out on the road and have a problem with a piece of equipment that cannot be fixed immediately. What we are set up to do is either send out new replacement parts or direct them to a service center where they will get it fixed and we will absorb the cost. We've found that it's the service more than anything else in our business philosophy, the standing behind anything we sell, that strengthens our reputation.

I get the feeling that West L.A. Music's success is due mainly to its ability to provide extensive and far-reaching service to its customers.

Hildebrand: That's right. The fact that the customer knows that when he comes in he's not going to get hyped. He won't be sold something that is not right for him. If somebody comes in and has read an ad about a new product that he thinks might be "the answer" to all of his musical problems, we will sit down with him and ask him exactly that he is looking for and try to determine whether he actually needs the product, or whether he might need something less complex, or perhaps maybe nothing at all. In this way we gain the customer's trust and the majority of our transactions is repeat business. We try to protect the consumers against what they may or may not know. That's why our reputation is so strong.

INDUSTRY UPDATE

NAMM's Western Exhibit is behind us all now, but January 20, 21 & 22 saw record numbers of industry folks strolling through Anaheim's Disneyland Convention Center. Some of the most notable visitors included; guitarist **Howard Roberts**, bassist **Ray Brown**, the Grateful Dead's **Bob Weir**, guitarist **Herb Ellis**, the Doobie Brothers' **Jeff Baxter**, former 10cc members and inventors of "The Gizmo" **Lol Creme** and **Kevin Godley**, drummer **Billy Cobham**, **Jim Messina**, the Eagles' **Don Felder**, and "**Jason**" the piano playing/guitarist family member of TV's "Waltons". A number of new products were introduced)see *NAMM/Sound Shoppe* article this issue) while thousands of dealers "wheeled and dealed" day and night. In addition to the many artists present, so were many of *Sound Arts*' contributors from within the industry. Among our friends were; Peavey's **L.A. Krause**, Elger's **Jeff Hasselberger**, Uni-Sync's **Larry Jaffe**, Whirlwind's **Michael Laiacona**, dbx's **Emil Handke**, Furman Sound Director/-Founder **Jim Furman**, and Contemporary Keyboard's **Dominic Milano**.

* * * *

Jazz/rock guitarist Marc Silverman will be touring around the country this year for Roland demonstrating their GR-500 guitar synthesizer at various dealerships. Silverman, who has composed and arranged music for the Roland system that incorporates five synthesizer sections, will showcase a variety of musical excerpts to illustrate the system's capabilities.

* * * *

Uni-Sync has announced the appointment of **Wayne Freeman** as Sales Manager. Mr. Freeman's duties will encompass domestic, as well as international, sales direction and management. He comes to Uni-Sync from BGW and has several years of consumer and professional audio sales management experience.

* * * *

After three months as staff writer in the newly formed communications group at TAPCO, **John C. Allen** has assumed the responsibility for all forms of communications at TAPCO, including corporate communications, marketing communications (including sales and service), and advertising and sales promotion for all TAPCO products internationally. John brings to TAPCO more than 20 years of experience as an editor, published author, illustrator, and professional communicator. Also on the TAPCO front is the appointment of **Clifford Miller** as Professional Products Sales Manager. Mr. Miller held a similar position for BOSE in the Northeast before moving to TAPCO. Cliff assumes responsibility for dealer training and the attainment of dealer sales goals for professional sound products sold by more than 500 TAPCO dealers throughout the U.S.

* * * *

Acoustic Control Corporation of Van Nuys, California has announced a couple of personnel appointments. **Richard Aspen Pittman** has been named to the position of Director of Marketing. In addition to Aspen's marketing duties, he'll be responsible for advertising, promotion, trade shows, group relations, dealer clinics, and a factory school. Aspen's former position at Acoustic was as West Coast Sales Consultant for five years. **Roger Smith** has also been named by this Southern California firm to the position of Sales Manager. Roger has been with Acoustic for three years as International Sales Director.

* * * *

An interesting combination: **Bob Moog's** creative input into the second-generation Gibson active electronics guitar, the RD-77. The instrument is currently in use by The Who's **John Entwistle**, recording for the next Who LP.

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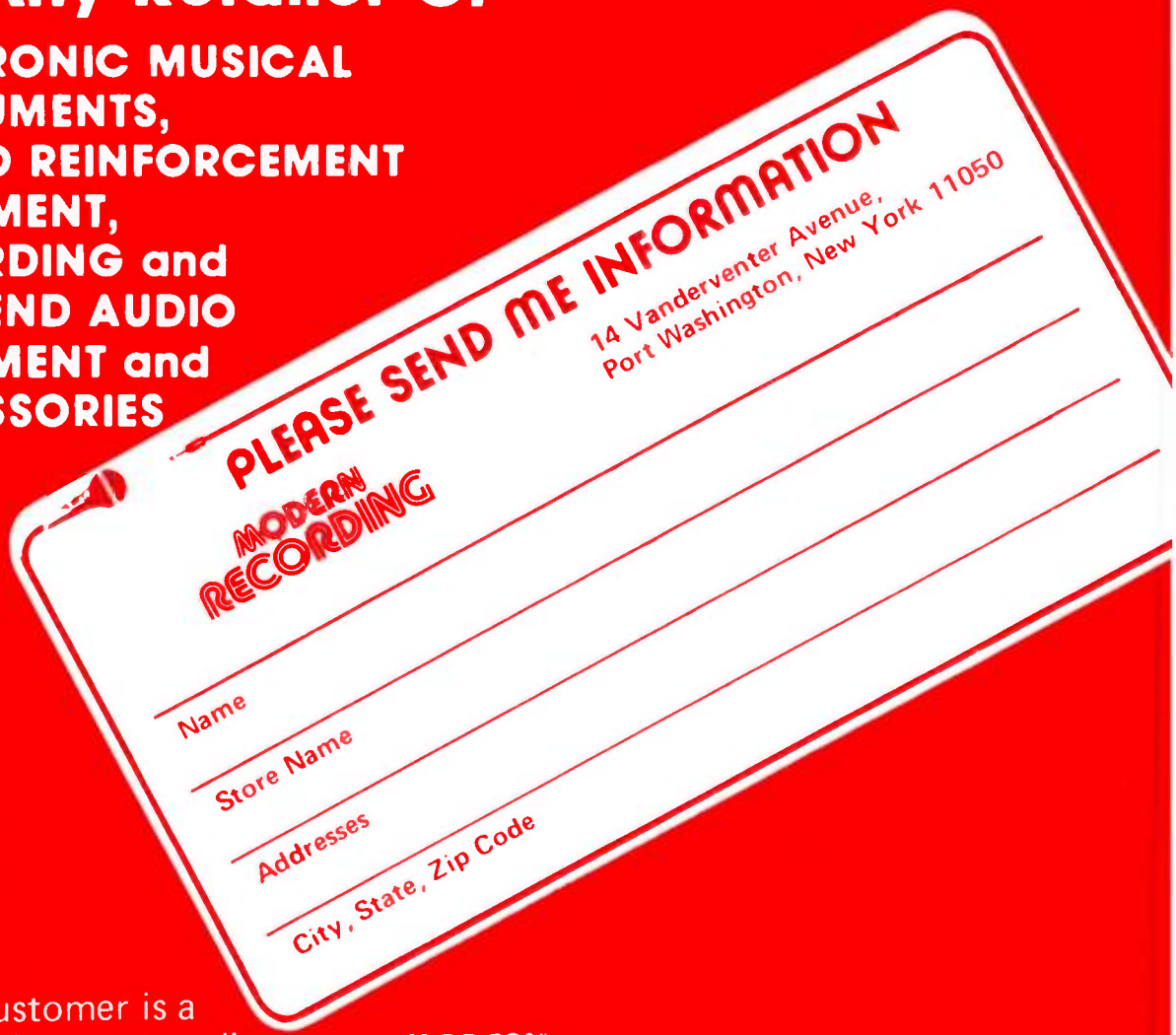

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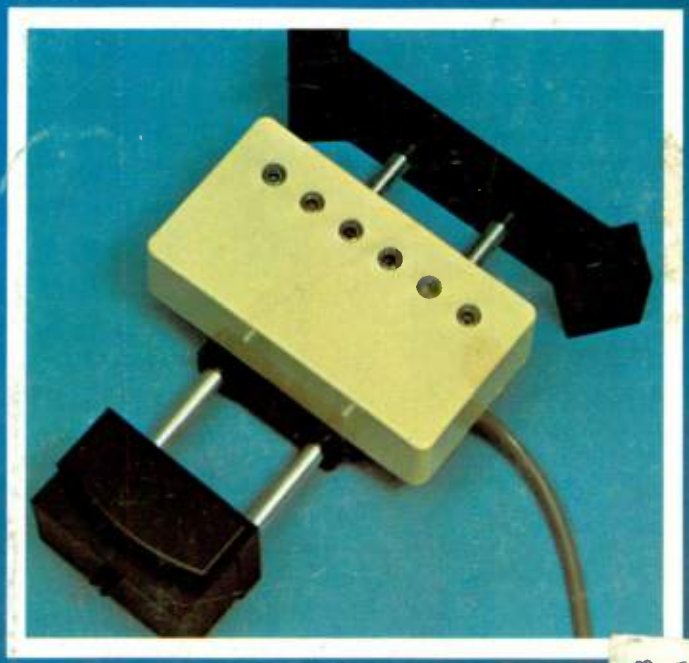

The DiMarzio Acoustic

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

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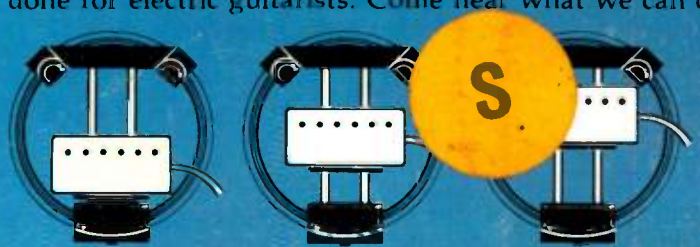

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