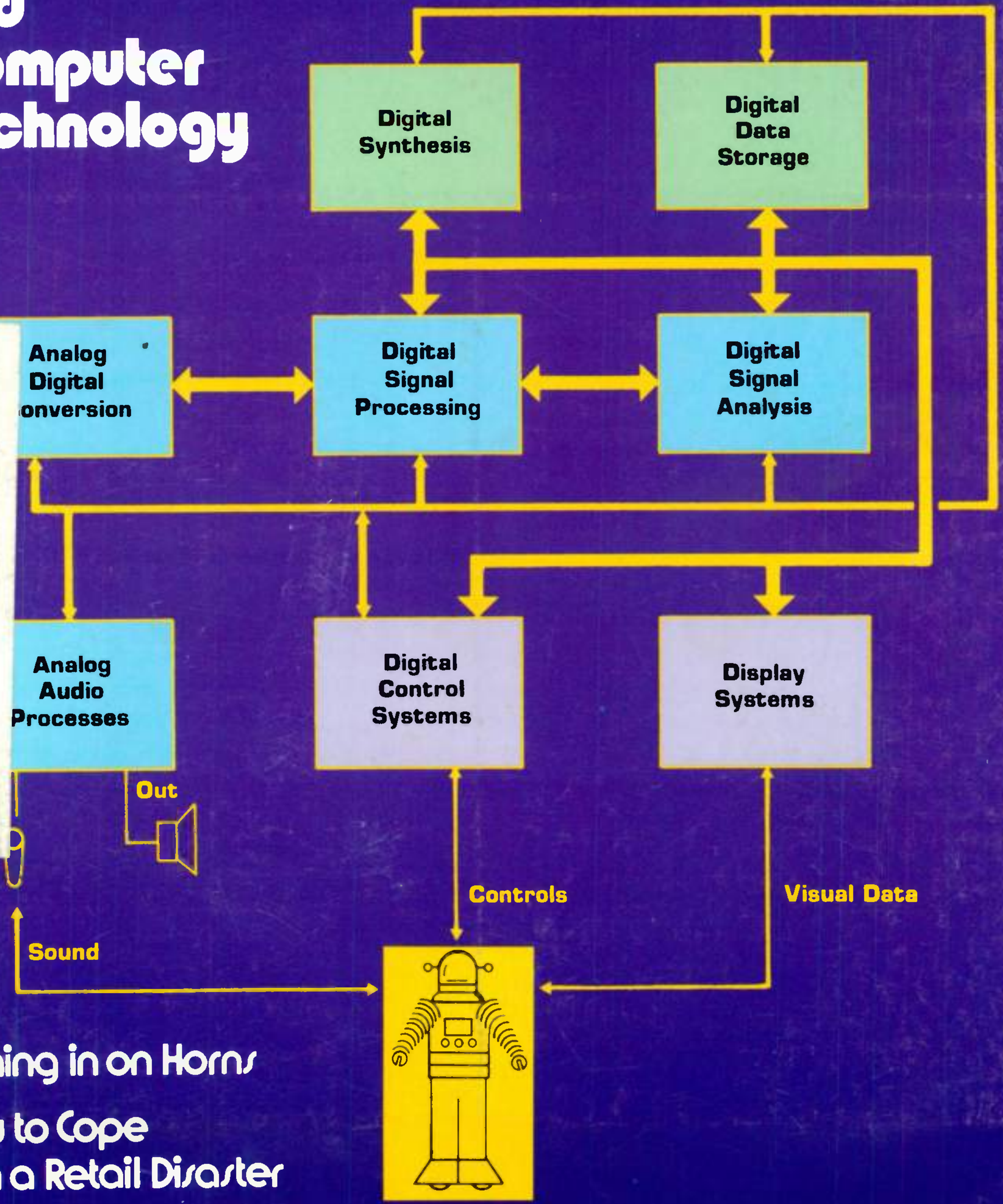
SERVING THE CREATIVE AUDIO AND MUSIC ELECTRONICS INDUSTRY

SOUND ARTS

MERCHANDISING JOURNAL

VOL.2 NO.3
APRIL 1979

Music and Computer Technology



Homing in on Horns

How to Cope with a Retail Disaster

THE LONG AND THE SHORT OF SOUND REINFORCEMENT.

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

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

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

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

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VOL. 2 NO. 3

SERVING THE CREATIVE AUDIO AND MUSIC ELECTRONICS INDUSTRY

SOUND ARTS

MERCHANDISING JOURNAL

APRIL 1979

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Cover art by Barry Simon

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SERVING THE CREATIVE AUDIO AND MUSIC ELECTRONICS INDUSTRY

SOUND ARTS

MERCHANDISING JOURNAL

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A LETTER FROM THE EDITOR

A sociological event took place in my bedroom a few weeks back. Don't jump to conclusions. My bedroom is the site of my Trinitron and the event was prompted by ABC's showing of "The Heroes of Rock and Roll," raptly viewed in my house by members of three contiguous but different generations.

A teenager, an almost-senior-citizen and an in-the-middle-grew-up-with-rock-and-roll enthusiast reacted to that TV show in ways peculiar to their respective generations. Yet they also attested to the universal importance and acceptance of what was once a very specialized kind of music.

The older generation, whose own youth coincided with swing and croon, reacted predictably to early r and r. ("The lowest common element.") And yet, early Beatles and early Motown came in for raves and sing-along.

The sixteen-year-old viewer, heard occasionally to pick on guitar and keyboards, found the TV show a revelatory experience. Fats Domino, Muddy Waters, even Buddy Holly were new voices lending a sense of musical heritage. It was back to Roots.

The in-the-middle viewer, of a generation that is of a piece with that of rock and roll, had a happy feeling of having been born at just the right time.

I started talking about all this, not because I wanted to talk about my bedroom, but because that TV show heightened my feeling that popular music exerts a pervasive and often unnoticed influence on everyone—cognoscente or not—and rock and roll music in particular is part of the cultural and emotional heritage that crosses ages. That's sometimes easy to forget in the hustle of moving product. But it's awful nice to remember that that inventory services "the purest of the arts."

I was thinking of all this as I was reading the copy sent in by Allen Hester for this month's issue. When Strings and Things in Memphis was destroyed last year, all parties rushed to Save the Guitars—the collectors pieces that were in the store. There were some economic considerations, no doubt, but no doubt also a little bit of the heart and heritage of everyone was placed in that collection.

Allen's article is not, however, about guitar collections—it's about coping with unexpected damage to your place of business and how to, if not plan for it (that's too heavy), at least anticipate the possibility.

A business is after all a business, music notwithstanding. But in order for it to be successful, art and management have to balance. Business savvy and artistic sensibility. In a successful musical business you can't have one without the other.

Regards,

Judy Lipton

Judith Morrison Lipton

Our microphones are more often heard than seen.

We really don't have to broadcast the virtues of our equipment.

Especially if you've ever broadcast *on* our equipment.

Infact, go into almost any professional facility, and it'll be easy to spot Sony. With one exception:

Our miniature omni-directional electret condenser mike. The ECM-50PS is so small, you'd never expect such big performance. Yet this tie-tac microphone offers a wide frequency response, with full coverage from any direction.

On your visit you'll also come across the Sony C-37P. This is a professional condenser mike that's at home on stage or in studios. This versatility is enhanced by a selector switch that lets you go from omni to uni-directional. And thanks to FET circuitry, the 37P boasts a remarkably wide dynamic range, allowing sound pressures of up to 154 dB.

With the ECM-56F, Sony moves in the direction of a uni-directional condenser microphone. Offering Sony's exclusive Back Electret design, this unit combines a wide frequency response, with uncanny smoothness.

The Back Electret also sets the ECM-53FP ahead. The microphone: a flexible Cardioid for desk or podium.

The Sony C-74 microphone (not pictured), is a gun-type. You'll often see it at news conferences, where loaded questions are asked. This uni-directional condenser microphone is acknowledged as the standard in its category.

It's no stranger to theatres, sound stages, large halls and television studios. When you can't get proximity, make sure you're not at a distance from Sony.

Sony's line of microphones is as complete as you'll find anywhere.

But it also has Sony's disciplined quality and on-going perfectionism. Which you won't find anywhere.

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FORUM

I would like to personally thank you for such an informative magazine. Our staff here at Summit Sound looks forward to reading your excellent articles every month. As we use these magazines in our reference library, it was requested that I ask you for back issues from Vol. 1 No. 1 to Vol. 1 No. 8. If this is possible, it would be appreciated by all of us for sure.

Keep up the good work and have a beer on us!

Best wishes,
Dean Ohira
Director of Operations
Summit Sound
Los Angeles, CA

Is it possible to obtain some missing monthly volumes? We are scheduled to receive SOUND ARTS every month, but occasionally we miss an issue or it goes home with an over-zealous client! The issues we need include December, April, March and February (all 1978). We will pay for these back issues if need be. No problem. Also, we sincerely wish to continue receiving the magazine as the monthly articles are an outstanding educational aid for all of us.

Thank you kindly,
David Ries
Marquette Melody Shoppe
Marquette, Miss.

As we are just about the only disco dealers in this area, we greatly appreciate the attention you pay to the increasing interest in disco lighting and sound. It might also be of interest to your readers to know that many of our disco sales are to mobile disco outfits.

Thanks again for a fine magazine—it is an able assist to us all!

Jim Steils
Manager
Sound Advice
Columbia, S.C.

I have to take exception with the retailer "composite comment" in your February 1979 issue which reads: "The reps don't know anything. My salesmen know more. I'd rather see a factory person." It is surprising this attitude was so pervasive because it has not been so evident in my dealings with 150 retailers during the past six years. Maybe I am naive but most of the reps I know are damn good salesmen. There are some, of course, who are not as knowledgeable as they could be but why should a few "bad apples" spoil it for the whole bunch?

Sales reps who are serious about and enjoy their work recognize the need for more self-education but this is a difficult task. Not only do we have to be familiar with our own products, but with those of the competition. Also, due to limited resources, many companies cannot provide adequate sales training.

One of the most frustrating tasks of a rep is to spend valuable but necessary time educating a retailer's salesperson, only to see that individual leave the business and be replaced by a novice. It is similar to the grief musicians experience when they lose a band member and have to break in a new one.

In the case of the factory expert, this person may appear to know more about the product due to technical expertise but is this person as able as a salesman to turn facts into benefits which improve the chance for a sale?

Perhaps your magazine can feature articles which present a clear picture of a sales rep's true role. It is only through better understanding that we can all work together to make this industry successful and pleasant for all who are part of it.

Thank you for your attention and fine efforts with an excellent much-needed publication.

Very truly yours
Tony Vespoli
Sales Representative
Ypsilanti, Michigan

Mr. Vespoli's comments are appreciated. We are planning some articles on the rep's role in the merchandising chain, and how to interact with him to optimum advantage. —*Editor*

TERMS:

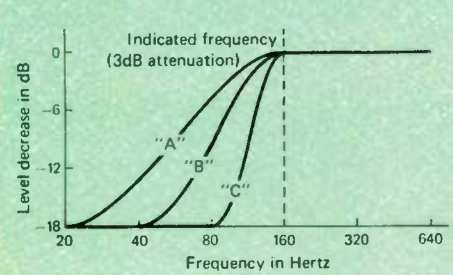
A CONTINUING INDUSTRY GLOSSARY

RECORDING

By Larry Blakely

As can be seen in the diagram below, curve "A" affects frequencies approximately three octaves below the indicated frequency and therefore increases at a rate of 6 dB per octave. Curve "B" affects frequencies approximately two octaves below the indicated frequency and increases at a rate of 9 dB per octave. Curve "C" is much sharper and only affects frequencies approximately one octave below the indicated frequency which increases at a rate of 18 dB per octave.

It is important to note that if the peaking type or shelving type equalizers were in an 18 dB of cut (attenuate) that the rate of decrease would be the same. Just imagine that the drawing is inverse or upside down. Filters are also affected by "Q" or the slope at which the level decreases versus frequency.



HIGH PASS FILTER

It can be seen that curve "A" decreases in level at a rate of 6 dB per octave and takes three octaves to reach an attenuation of 18 dB. Curve "B" decreases in level at a rate of 9 dB per octave and requires two octaves to reach an attenuation of 18 dB. Curve "C" decreases in level at a rate of 18 dB per octave and only takes one octave to reach an attenuation of 18 dB.

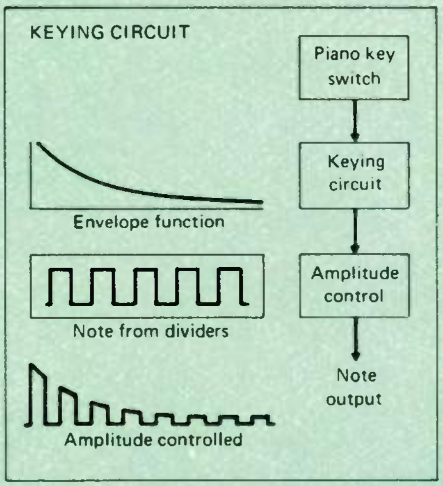
The same will apply to a low pass filter, which can be seen in the drawing on the back.

It can be seen that curve "A" attenuates at a rate of 6 dB per octave. Curve "B" attenuates at a rate of 9 dB per octave, while curve "C" attenuates at a rate of 18 dB per octave.

ELECTRONIC MUSICAL INSTRUMENTS & ACCESSORIES

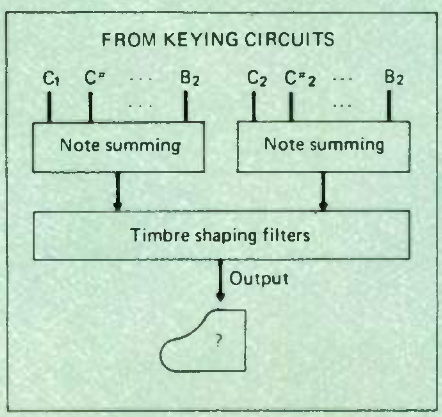
By Mike Beigel

As we promised at the end of our discussion last month, the drawing below shows a synthesizer keying circuit.



Note-Summing Circuit: A simple analog adder that sums the outputs from all the keying circuits. Sometimes all the notes from the piano are summed into one circuit, but in better designs each octave goes to a separate summing circuit.

Timbre Shaping Filters: The pulse-wave outputs of the piano keying circuits are not musically desirable, so audio filter circuits shape a more pleasing sound spectrum. As with the summing circuits, the filter sections can operate on all the notes for economy, or on each octave separately for better musical sound. The



SOUND REINFORCEMENT

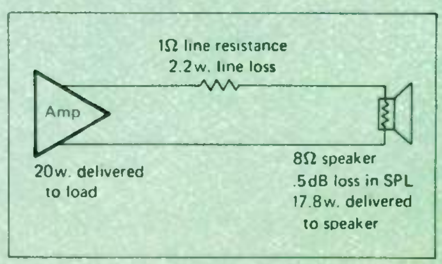
By Glen E. Meyer

High Impedance Voltages: The most common voltages used in high impedance/constant-voltage systems are 25 and 70.7 volts. When compared to 25 volt systems, the 70.7 volt system requires a smaller wire diameter (remember, current is equal to power divided by voltage) to carry the same wattage. 25 volt systems are sometimes used in fixed installations because some states require wire to be run in conduit when the voltages encountered are greater than 25 volts. And conduit is generally more expensive to put in than the larger wire size required by the 25 volt system.

I do not wish to imply that every time one has a long speaker wire run, a high voltage system should be used. The cost and the performance capabilities of the transformers need to be considered. Most inexpensive line transformers do not have adequate low-frequency low-distortion capabilities for higher-powered low-frequency systems.

Power Loss in Long Lines: For long wire runs, the power losses in the wire become significant, especially for smaller wire sizes. For a 0.1 dB loss in sound pressure level, the total wire impedance, out to the speaker and back, must be limited to 1% of the speaker impedance. The somewhat larger losses of 0.5 dB and 1 dB (in most cases acceptable) are encountered when the percentages increase to 6% and 12% respectively.

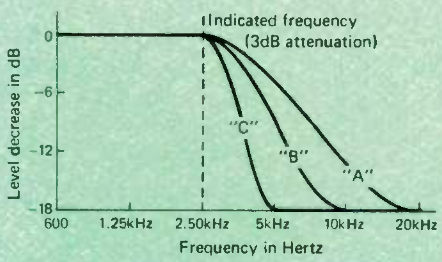
The power supplied from the amplifier terminals to the load is divided between the resistance of the wire and the impedance of the speaker. See diagram below for an example of this.



TERMS (CONTINUED)

A CONTINUING INDUSTRY GLOSSARY

RECORDING



LOW PASS FILTER

Filters are typically built with attenuation rates in multiples of 6 dB per octave. Filters are normally specified as 6, 12, 18 or 24 dB per octave slope.

High-Pass Filter: A filter that will pass all frequencies *above* the frequency indicated on the front panel. All frequencies below the indicated frequency will be attenuated. The amount of attenuation of frequencies below the indicated frequency depend upon the "Q" (rate or slope) of the filter.

Low-Pass Filter: A filter that will pass all frequencies *below* the frequency indicated on the front panel. All frequencies above the indicated frequency will be attenuated. The amount of attenuation of frequencies above the indicated frequency will depend upon the "Q" (rate or slope) of the filter.

Band-Pass Filter: A filter made up of a high-pass filter and a low-pass filter. Such a band-pass filter would have two indicated frequencies. This filter will only pass frequencies between the indicated frequencies. For example: if the indicated frequencies were 10 kHz and 1 kHz, all frequencies above 10 kHz will be attenuated, all frequencies below 1 kHz will be attenuated, and all frequencies between 10 kHz and 1 kHz will pass.

Switched-Type Filters: Can be of the high-pass, low-pass, or band-pass types. Such switched-type filters will have a number of predetermined frequencies on the front panel that can be selected (switched) by the operator.

ELECTRONIC MUSICAL INSTRUMENTS & ACCESSORIES

"voicing" switches on electronic pianos usually select different filter characteristics or combinations.

Gizmotron: An electro-mechanical accessory that simulates the "bowed string" sound on an electric guitar. A rotating "bowing wheel" is brought into contact with the guitar string and vibrates the string as long as contact is maintained (see Figure 1). Six bowing wheels and actuator buttons

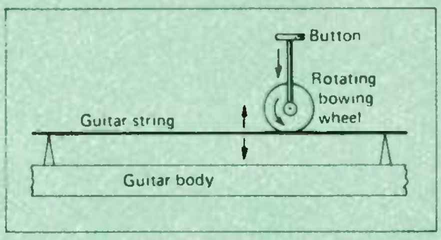


Figure 1

allow the guitar to be "bowed" polyphonically, simulating a string section. The wheels are all driven by a rotating staff coupled to an electric motor (see Figure 2). The electro-mechanical approach has the advantages of a fast onset time for the sound, and variable

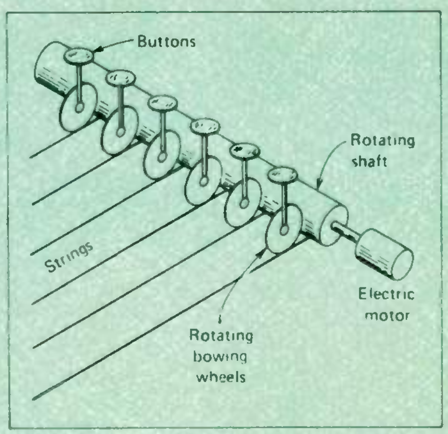


Figure 2

loudness dynamics. The device doesn't interfere with normal guitar playing: bowed sounds and plucked sounds can be played simultaneously on different strings when using the device.

SOUND REINFORCEMENT

If we assume that the amplifier is delivering 20 watts to its load, then only 17.8 watts is being delivered to the speaker while 2.2 watts is lost in line resistance.

A basic discussion of microphones and their use in sound reinforcement systems follows.

Microphone: A transducer which converts sound into an electrical output or voltage.

Microphone Attenuator: An electrical device (generally passive) inserted between microphone and preamplifier which, through a resistive network, eliminates input overload by lowering microphone output.

Feedback: A "howl" or "squeal" produced in a sound system that occurs when the output of the loudspeaker enters the microphone or guitar pickup and is reamplified. Proper microphone type and use can help reduce feedback. More on this later.

Frequency Response: The manner in which the output of a device responds to and varies with changes in input frequency. A "flat response" microphone, for example, indicates near-equal response over the entire range of frequencies. In obtaining optimum performance, microphone response is often "tailored" by introducing low-frequency "rolloff" and controlled high frequency "boost." These "rolloffs" and "boosts" must be accomplished smoothly. Abrupt variations in the frequency response curve indicate poor microphone quality.

Microphone Mixer: An electronic device which permits combination of several signal inputs into a single output. Usually independent preamplifiers, tone, and volume controls are provided for balancing input signals.

Windscreen. An acoustically transparent shield, usually placed around the entire microphone, which sharply reduces the audible effect of wind or rapid movement of the microphone through the air.

This door represents expanded realms of creativity.

MXR opens this door.

MXR has made a commitment to itself, its customers and the future of the music industry—that is to continue as the leader in the field of electronic signal processing.

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TROUBLESHOOTERS' BULLETIN

1

LOW FREQUENCY INTERFERENCE

Interference can be a major problem at the lower frequencies when recording with unidirectional microphones. Unfortunately, directional mics have a tendency to become omnidirectional at low frequencies. Unwanted sounds can be easily picked up without proper precautions. The situation is remedied with low-cut filters (built into many mic

2

models); proper equalization; and with care in the placement of the mics themselves.

NICK MORRIS
SONY PROFESSIONAL AUDIO

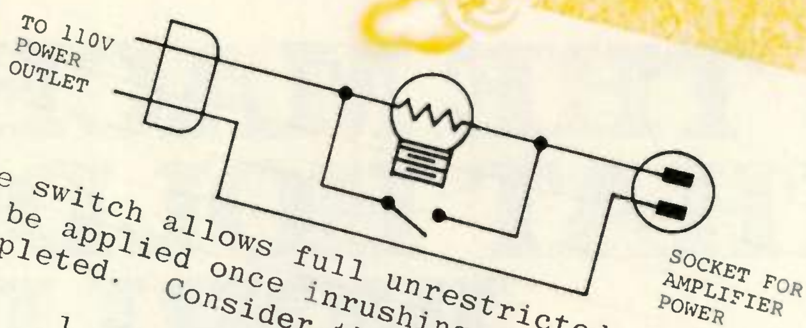
CURRENT LIMITING TIPS

Most of us have become aware that the most precarious point in an amplifier's existence is at the time of turn on. Inrushing currents and sudden temperature changes can

3

be quite fatal. If you have just serviced the unit, throwing that power switch is akin to crossing the Rubicon, traditionally there has been no turning back. To avoid this moment of truth, you might want to adopt a simple current limiting scheme that insures "soft" turn on, allows bias to be set, and will protect the amplifier while you work on it with the power "live."

The technique involves simply putting a light bulb in series with the line cord. Use a 4-Square Box with a bulb socket, switch and outlet socket wired thus:



4

The switch allows full unrestricted power to be applied once inrushing or service is completed. Consider the following:

1. Initial turn-on will cause the lamp to go to dull brightness and then taper off to a dull glow. This is the result of the inrushing current to the filter capacitors.

5

2. Servicing an amplifier when operating in the restricted current mode facilitates repairing the unit with the power on. Insufficient current is available to damage the output devices and the bulb acts as an indicator of the shorted condition and protection against a short created during the service action.

3. Shorts and high current situations such as high bias, a shorted semiconductor or wiring will cause the bulb to illuminate and limit the power available to the amplifier. This condition facilitates repairs with the power on. Once the shorted condition is relieved the lamp level will drop to normal. As the bias is increased the lamp intensity will increase proportionally indicating current draw.

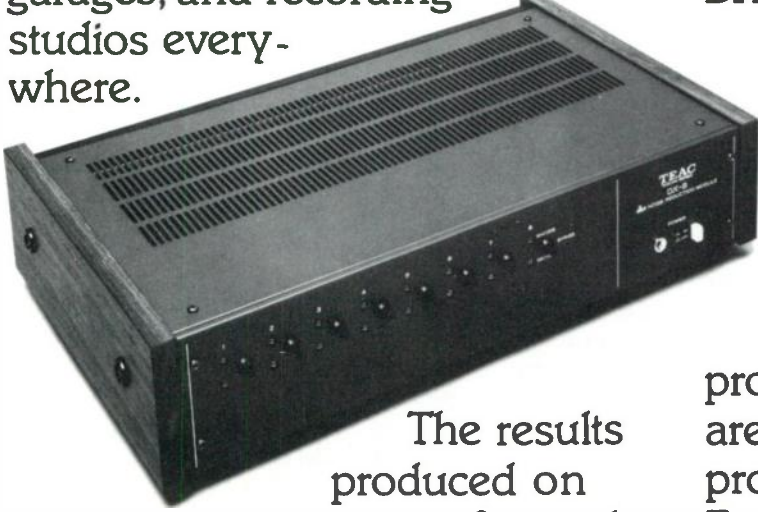
6

4. Bulb size should be equal to the power output of one channel of the unit under test.

BARRY THORNTON
CHIEF ENGINEER
SAE

THE MACHINE THAT HOLDS THE WORLD TRACK RECORD.

The Tascam Series 80-8 has become the most popular 8-track multichannel recorder in the world. Its reliability has been proven in basements, garages, and recording studios everywhere.



The results produced on the 80-8 are a matter of record. Sometimes gold.

The 80-8 proved a new standard was needed. Eight tracks on half-inch tape. 15 ips only. This new format allowed us to create a combined record/reproduce head, with full frequency response in the sync mode.

The 80-8 proved multichannel recorders could be relatively easy to operate. Our Function Select buttons determine the record, monitoring and dbx* status. One button for each track.

The 80-8 proved that performance and versatility could be affordable. Signal-to-noise is better than 95 dB (weighted) with our integral dbx unit (Model DX-8). Once installed, it's totally automatic. And our new Variable Speed Control** lets you adjust 15 ips ±20% to solve tough cueing and timing problems or add creative effects.

The 80-8 is proving that in professional recording, results are all that count. Because to us, pro means results. On demand. For payment.

If you agree, see your Tascam Series dealer for the machine that can prove it. Because it makes sense to do business with the people who have the track record.

*Registered trade mark of dbx, Inc.
**Installation required; a new DC servo-controlled motor is included.

TASCAM SERIES
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Common Consumer Questions

What is tape flux?

Tape flux is somewhat difficult to explain because it's invisible. As you know, every magnet produces a "field." Magnetic (or inductive) *flux* is a measure of that field's strength. And just as there are units of measure for various types of electrical output, such as Volts or Watts, there is also a unit of measure for magnetic "output." The established units of magnetic flux are *Webers* or *Maxwells.* (The two are not interchangeable: 1 Weber = 10^8 Maxwells.)

Since recording tape is nothing more than a plastic backing material coated with a large number of microscopic magnets, these units of magnetic flux can serve as a measure of the tape's "output." Thus the term *tape flux.* It means the same thing whether you are talking about cassette or open reel. When a piece of recorded tape is passed over a magnetic head, the head detects changes in the tape flux level and converts them to electrical current, which can then be amplified and ultimately "heard" over a pair of loudspeakers or headphones.

A term often used in tape-talk is *fluxivity.* This is simply the amount of flux for a given unit-length of tape, and it is used to define the level of the recorded signal. Maxwells per inch, for example, would be one way of quantifying tape flux or signal level. But the most popular units of tape fluxivity by far are *nanoWebers per meter* (nWb/m), *picoWebers per millimeter* (pWb/mm) and *milliMaxwells per millimeter* (mM/mm), because these yield the most convenient numbers when dealing with current tape technology. For example, the "reference level" specified by Dolby Laboratories for their B-type noise reduction system as applied to cassette decks is 200 nWb/m at 400 Hz. The NAB "0 VU" operating level for open reel decks is defined as a tape fluxivity of 185 nWb/m at 700 Hz.

There are several different known methods for measuring tape fluxivity. The most widely used methods usually involve running the tape in question over a head with known characteristics. As you can probably tell, tape flux in the strictest sense is a matter of concern primarily to engineers and researchers. For the professional recordist, advanced amateur or service technician, the industry offers numerous alignment tapes which contain test tones recorded at pre-determined flux values. These test tapes, of course, are an essential part of maintaining international compatibility standards for both open reel and cassette decks.

All the average tape consumer needs to know about tape flux is that it's there and working to produce the desired results. If a customer insists on "seeing" tape flux in action, point him or her in the direction of the tape deck's level meters.

Harron K. Appleman
Technical Director
Nakamichi Research (U.S.A.)

What methods may be used to produce echo, reverb, and delay effects?

The natural method puts the musicians in a large room or studio with walls, floor and ceiling that will reflect the sound. Materials that absorb sound such as wall paper and carpets are not used. Sounds reflected off the walls, floor and ceiling create the echo. This is often called an echo chamber.

A similar method is to place a speaker and a microphone at opposite ends of a similar echo chamber. The musicians in another studio are piped over the speaker and the microphone picks up the sound which travels directly to it from the speaker. Then the sounds which reflect off the walls, floor and ceiling reach the microphone a short time later, creating the desired echo effect.

Both of these methods tend to be difficult to control, due to the factors which determine the actual sound. These factors include the actual size and shape of the room, and the materials used in the building of the walls, ceiling and floor.

An old trick sometimes used by broadcast engineers to create an echo was to create a feedback loop between two channels on the board. Usually this is done through an outside circuit, such as the amplifier in a tape deck. The signal is actually inputed through one channel and fed to the output or program bus. The signal is picked up from this bus and fed through this loop and then to the other channel. The time it takes the signal to travel through this loop creates the echo effect, much the same as the extra time it takes signals in an echo chamber to bounce off the walls and hit the microphone after the original sound. After going through this loop, this delayed signal is fed into the output bus, through the second channel's fader. A problem with this method is lack of control—the delay time is whatever it takes the signal to travel through this loop. Another problem is that this loop can easily develop enough feedback to start feedback oscillation.

Tape echo records the sound on magnetic tape which then passes by one or more playback heads. The time from when the signal is first created until it is reproduced by the playback heads simulates the time it takes the sound to be reflected back in the echo chamber. The delay time in the tape unit is determined by the distance between the record and playback heads and the time it takes the signal on the tape to move from the record head to the playback head. Here we have control over the delay time by changing the distance between heads or the tape speed or both.

Spring reverb units use one or more metal springs between a transducer and a pickup. The transducer vibrates the spring the same way the voice coil of a speaker vibrates a speaker cone to produce the sound. These vibrations travel across the spring to the pickup which acts like a microphone. The time it takes the sound vibrations to travel the length of the spring from transducer to pickup creates the delay. The same sound vibration may travel the spring more than once, resulting in several separate echoes, each softer than the one before.

Plate echo units are similar to spring units except that they use a vibrating metal plate or foil instead of springs. Pickups may be placed at the end of the plate or some place before the end of the plate.

Spring and plate units can be very sensitive to outside vibration and movement and must usually be completely enclosed and shock-mounted to prevent problems.

Modern computer electronics technology has resulted in several types of electronic delay devices which have found their way into audio equipment during the past few years. These delay devices typically consist of a single integrated circuit which has the advantages of requiring only a small amount of space and having no physically moving parts, enabling them to be unaffected by vibration and movement. Being so small, these devices easily fit into the electronics of a control board.

Shift registers and bucket brigades comprise one group of these devices. Both operate in similar manner. When the signal enters the device, it is held at one point, for a short time, and then moves to another point, where it is held for a short time and then moved to the next point and so on. As this occurs the next portion of the signal follows it from one point to another and so on. In these devices the signal moves through a given number of points at a certain speed determined by how often the signal is moved to the next point and the total number of points between the input and output. Sometimes an electronic clock circuit will control the movements of signals from point to point in these devices.

Another of these devices is the random access memory, often called a RAM. The RAM also stores signals at individual points within its circuit. However, we can choose which point or points we want to use. Since each point takes a known amount of time for the signal to pass through it, we simply use the number of points that will give us the amount of delay we want.

Neil Lewbel
Consultant, Technical Writer
Kew Gardens, N.Y.

So... u want to know about

Synthesizer Basics Part II

By Craig Anderton

Last month we discussed synthesizers in very general terms—how a synthesizer is a collection of modules, some of which generate, and some of which modify, sound. Additional modules act as control devices that make the generators and modifiers perform in specific ways.

This month, let's take a closer look at sound generators, specifically the kinds included with today's synthesizers. A good place to start is the *oscillator* module, the part of the synthesizer that physically generates tones we can modify further on down the signal path.

I'd like to begin by answering a deceptively simple question that people often ask: "But how does a bunch of wires make sound?" Actually, there's nothing that mysterious about the process; sound waves correspond to variations in air pressure (or at least that's the accepted explanation!), speakers can create variations in air pressure by moving their paper cones back and forth, and all the oscillator does is generate a signal that moves the speaker cone back and forth . . . thus creating changes in air pressure that we hear as sound.

However, there are *lots* of different ways to move a speaker back and forth . . . anything from AM radio, to bass guitars, to signals from space. What's interesting about a synthesizer is that you have a tremendous degree of control over the exact sound you want to produce. Chances are if you can picture a sound in your head, given enough synthesis equipment you could exactly duplicate that sound in reality.

OSCILLATOR WAVEFORMS

Here we'll run into some words that often send the non-technically oriented running for cover . . . but really, it will all make sense by the end.

Some sounds are soft and mellow, like flutes; and some are bright and brassy, like trumpets and fuzz guitars. So it seems logical that if we want to be able to synthesize any existing or imaginary sound, we should be able to control the timbre (or tone quality) of our sound generator as fully and completely as we can. Remember how we've said that sound is a bunch of waves? Well, the timbre of that sound depends on the form of the wave . . . so we end up with the concept of a *waveform*, which identifies the shape, or character, of the sound wave we're dealing with. Luckily, electronic circuits can generate quite a wide variety of waveforms.

By now you may be wondering what accounts for one waveform being different from another waveform. The answer lies in the *complexity* of the waveform. For example, when you hit an "A" 440 on a piano you're getting a *lot* more than just a single "A" 440 tone; you're also getting a bunch of higher overtones (called harmonics) of the signal. This may seem puzzling, since how can several different sounds be in the same place at the same time? For an analogy, think of white light. It appears to be a single type of light, and yet, when put through a prism you can see that it is really made up of a number of different colors. But when these colors are merged together, you associate that combination of colors as "white." Similarly, by adding different sounds together in different combinations you obtain different waveforms.

At this point, the conventional thing to do would be to trot out information on harmonic content and show some oscilloscope drawings of the different waveforms—but let's break with convention for a few sentences and just talk about these things in terms of sound quality.

The simplest waveform to the ear is the *sine* wave. It is a very mellow, very basic waveform that is flute-like in character. One step further in complexity is the *triangle* wave. This sounds similar to a sine wave, but has a bit more buzz to it. . .almost as if you ran a sine wave through an amp with a little bit of distortion. The triangle wave resembles the sound of a clarinet.

The reason the sine wave sounds so plain is because it has *no* harmonics, whereas the triangle wave has a little bit of harmonic content. For more complex synthesized sounds, however, we need to use waveforms with significantly more complex harmonic structures.

The *square* wave is brighter in sound than either the sine or triangle wave, but the *sawtooth* wave is the richest sounding of all, because it contains the greatest number of harmonics. This makes the sawtooth waveform an ideal candidate for synthesizing brass and other bright sounding instruments.

Another waveform often found on synthesizer oscillators is the *pulse* waveform. Electrically speaking it is a variation on the square wave, but can sound sharper and/or thinner, depending on the characteristics of the


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pulse. Some synthesizers have the option of a *variable* pulse waveform. By variable, we mean the ability to actually change the harmonic characteristics of this one waveform via some kind of knob or control.

A final common waveform is called *white noise*. This resembles the sound of rain, surf, or a noisy amplifier, and is good for synthesizing cymbal effects, adding additional sonic complexity to standard waveforms, etc.

The ideal synthesizer would have sine, triangle, sawtooth, variable pulse, and white noise waveforms available at the same time; many of the higher priced models do, in fact, have this capability. Lower priced synthesizers, on the other hand, will often only have one of two of these waveforms to choose from. At first, this might sound like a tremendous disadvantage—but in reality, there are ways around this apparent limitation. One of the ways is a module called a *filter*, which is capable of fine-tuning the harmonic structure of a given waveform. One popular type of filter is called a *low pass* filter. It gets its name because it will only pass sound below a certain frequency, and this frequency is specified by a knob or other controller. Anyway, by passing even as complex a waveform as a sawtooth wave through a low pass filter, it is possible to set the filter so that it cuts out all the high frequency overtones, thereby turning our once complex waveform into a simple sine wave.

Well, this certainly seems like a powerful combination—an oscillator capable of generating a variety of basic waveforms, plus a filter module that can further alter the basic waveform put out by the oscillator. However, there are some drawbacks to this basic system which have led to new developments in synthesis. While these developments generally haven't filtered down yet to the club musician on the street, knowing the rate at which electronics moves I'm sure they'll be showing up soon in mass-market instruments.

One alternative is called the *additive* synthesis method of constructing a waveform. The process we've discussed so far for generating sound is called subtractive synthesis, because we start off with a signal that has a fixed harmonic content and use the filter to *subtract* those harmonics we wish to remove. Additive synthesis instead builds a harmonically complex waveform out of a number of very simple waveforms (usually sine waves, but not always). However, an additive synthesis system generally requires more hardware than a subtractive type, which is why we haven't seen too many of the latter around.

Another alternative is to come up with a module that actually generates the complete waveform desired right from the start—no taking complex waveforms and subtracting, or taking simple waveforms and adding. RMI puts out an instrument called the Keyboard Computer that uses this principle (see SOUND ARTS, February 1979); it synthesizes the waveform of a signal by plotting the characteristics of that signal and mimicking those characteristics through *digital*, computer-like techniques.

Yet another alternative to conventional oscillators is already present on many synthesizers: an external input jack. There's no need to limit your synthesizer to modifying the sounds of its own oscillators; synthesizers may also modify guitars, microphones, electric pianos, and the like. The synthesizer will also generally provide some means of deriving as many control functions as possible from the external input. Using synthesizers with external input signals has not been extensively explored, but can produce some very interesting effects. For example, while producing a classical guitar recording I took the artist's acoustic guitar and processed it through a synthesizer system.

Oscillators have other terms associated with them, such as "exponential response," "volts-per-octave," and the like. However, these terms have no musical meaning; they refer to the electrical characteristics of the oscillator in question, and only become important when trying to interface different synthesizer systems together. All you really need to care about with oscillators is whether they are: stable, so they don't go out of tune with changes in temperature or mechanical vibration; offer a goodly number of waveforms, and if not, that there are some decent filters further downstream so you can modify the waveforms a little more; and they have correct, even-tempered intonation when connected to a keyboard . . . it's no fun playing a synthesizer with an intonation problem.

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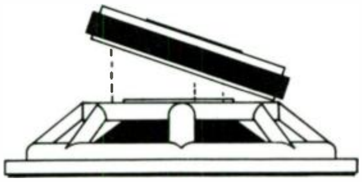
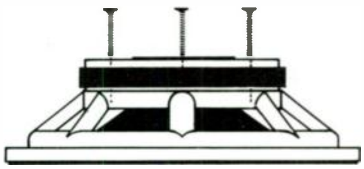
The combination of a rigid cast-aluminum frame and high-efficiency magnetic structure is a feature found in many professional quality loudspeakers. What places the Black Widow Series far ahead of its competition is its field replacable basket assembly.





This feature, usually found only in high quality compression drivers, allows the user to be "back in business" in a matter of minutes, rather than days or weeks.

The high efficiency and high power handling capabilities found in the Black Widow make each model the best choice for its sound reproduction application. Again, what separates the Black Widow from other high quality transducers is its unique integral coil form/dome structure. When a loudspeaker is subjected to very high power levels, the voice coil temperature rises very rapidly, causing the loudspeaker impedance to increase. The result of this increase is a loss of efficiency. The Black Widow Series provides a most effective method of minimizing any impedance increases due to heat by utilizing the one-piece coil form/dome as a heatsink. Just as high power amplifiers use aluminum heatsinks to dissipate heat, the Black Widow coil form/dome is produced with low mass, high rigidity aluminum.

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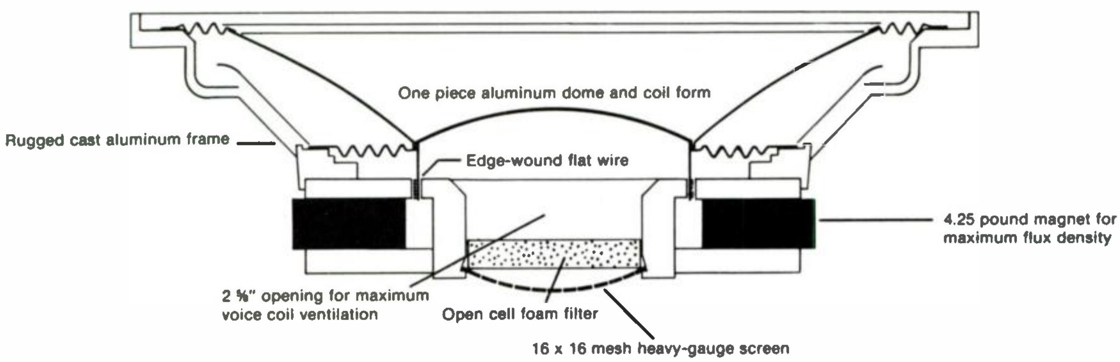
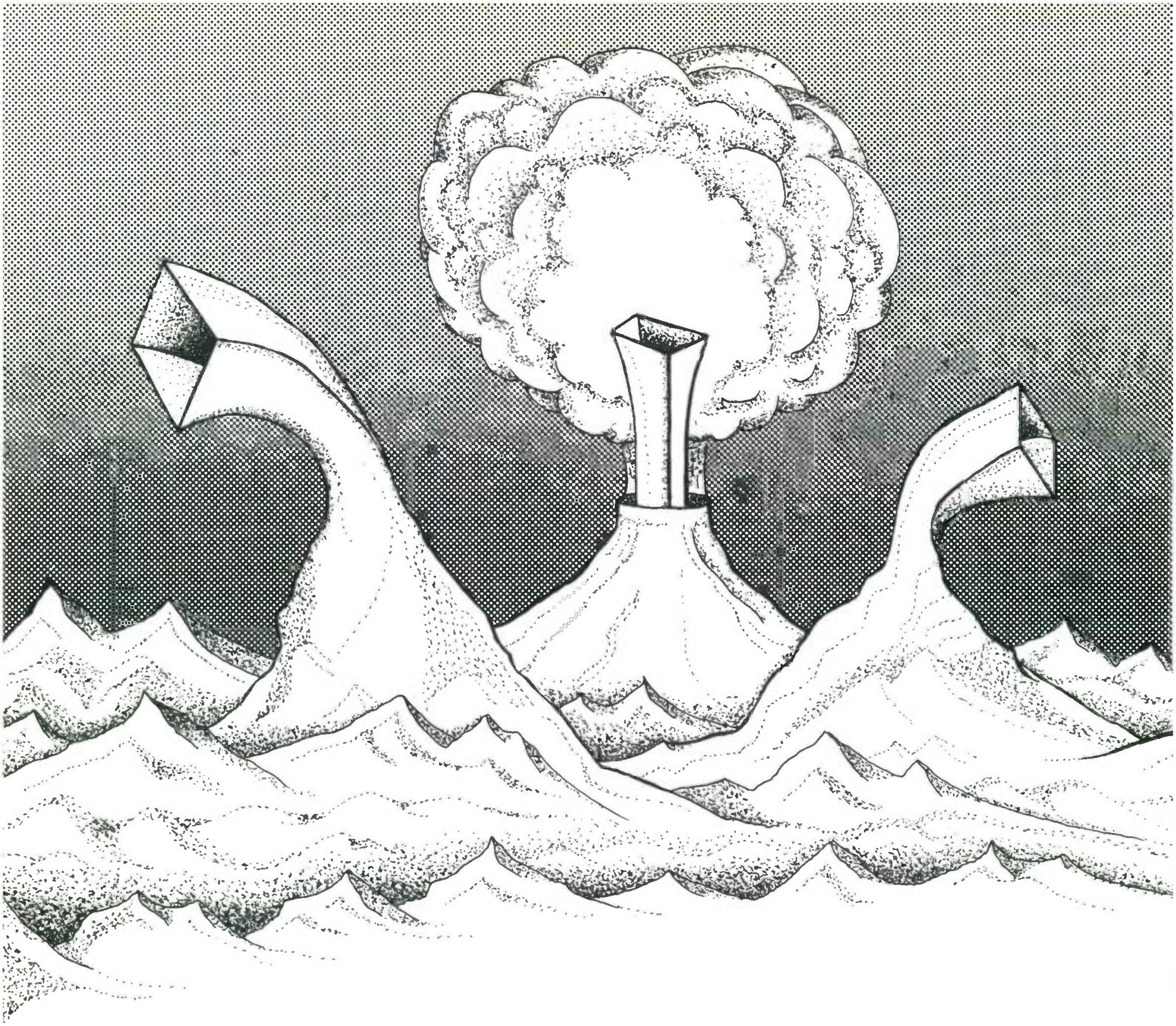


MODEL NO.	DIAMETER	NOMINAL IMPEDANCE	POWER HANDLING CAPACITY CONTINUOUS	PROGRAM	SENSITIVITY 1w, 1m on axis	VOICE COIL DIAMETER
1201	12"	4/8	150W	300W	101 dB	4"
1501	15"	4/8	150W	300W	103 dB	4"
1502	15"	4/8	150W	300W	101 dB	4"
1503	15"	8	150W	300W	102 dB	4"
1801	18"	4/8	150W	300W	99 dB	4"



The screen door slammed, leaving the old lady in her faded blue housedress, brown arms bare, standing on the dusty porch. Her eyes traveled slowly across the empty farmyard and the quiet fields just beyond. He wasn't in sight. She sighed once and stepped gingerly from the porch, her knee creaking with the wood, and walked slowly across the yard, scanning the horizon as she went. Nothing. She wiped her flour-dusted hands on her apron and lifted them to her mouth. Frequency range: oh, 800 to 2.5 K. Flare rate: 400 to 500 Hz. She angled her hands precisely, looking around as she thought despairingly, adjusting her coverage to take in another thirty degrees. She took a deep breath.

"Harry!" she called, enunciating the vowel sounds.

A small boy raised his head from the crevice by the creek. Spotting him instantly, she narrowed the angle of her cupped hands and filled her lungs again.

"Harry!" This time an octave higher. He jumped to his feet.

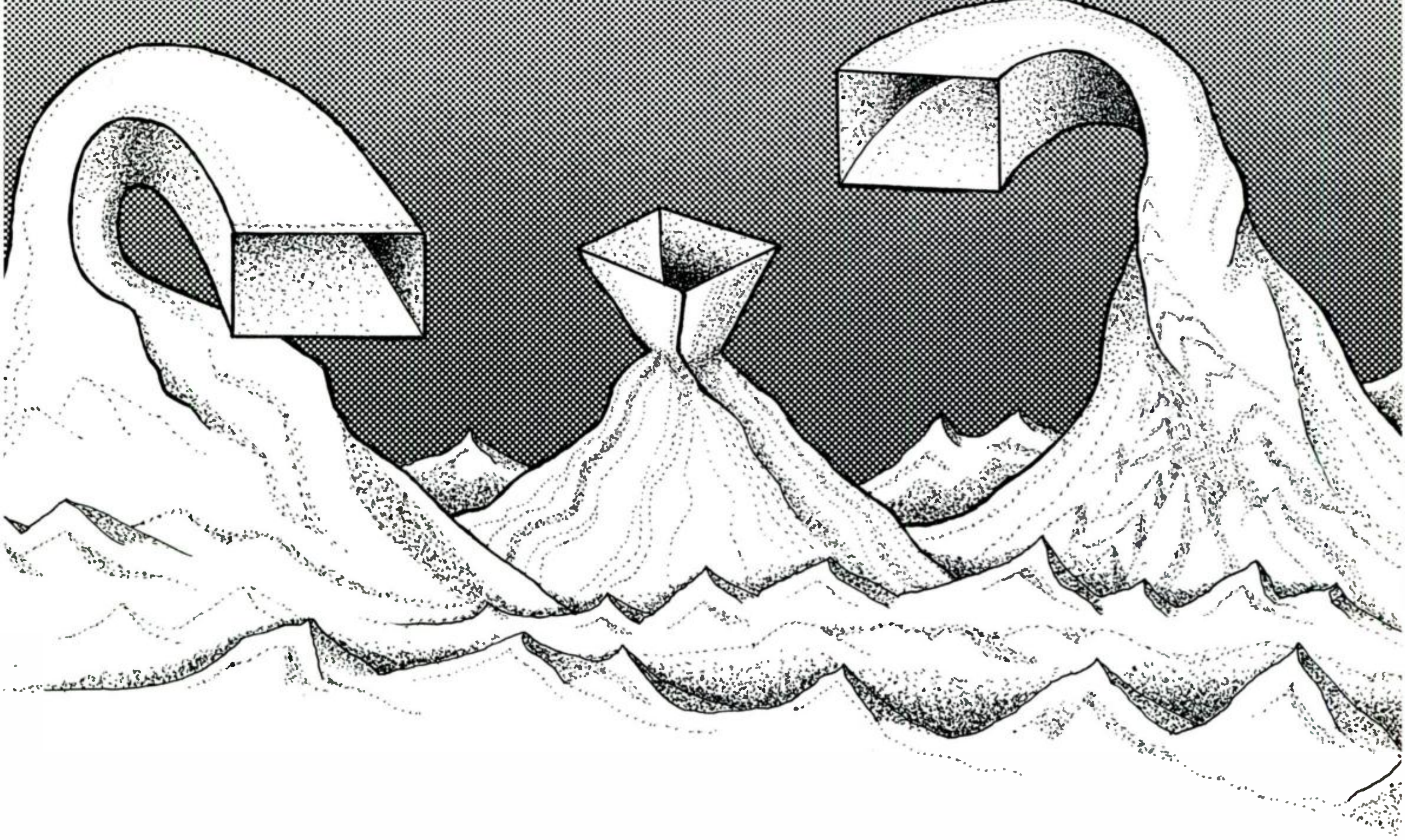
"Dinner!" He came running.

Everybody knows what horns are. You've heard them keeping score from high over the end-zone of football fields, urging you to get out and vote from the tops of sound trucks. You demonstrated a working knowledge of horns the first time you called across a distance and cupped your hands to your mouth: a natural reflex in human beings exhibited by even the most avid proponents of all-direct-radiator sound systems. Everybody knows what horns are. But how do they work? What kinds are available to the professional user and how are they applied in a sound reinforcement system?

A horn can be thought of as an acoustical transformer; its purpose: to increase the coupling, or bite, between a source of sound and the surrounding air. It accomplishes this by acting as a wave guide, or transmission channel, progressively engaging successively larger layers of air from throat to mouth, gradually increasing the effective radiating area of the diaphragm

HOMING IN ON HORNS

BY L. RICHARD FELD & CHRISTINE KOFOED



and providing an efficient change of mechanical into acoustical energy. It "sounds louder" not because the electrical-mechanical efficiency or power capability of the speaker itself has changed, but because the transfer of energy is greatly increased.

There are certain basic parameters of a horn that determine its frequency range and dispersion characteristics: mouth area and flare rate set the lower limits of the horn's operating range; mouth dimensions and horn shape will govern dispersion patterns within that operating range. Flare rate is derived mathematically from the equation used to produce a particular horn; physically it represents the lowest frequency to which the device would function if the mouth were extended to infinity. It is difficult to provide an accurate rule for mouth size versus lower limit that properly represents the performance of all types of horns, but in general, the larger the mouth the better the operation toward the lower end of the driver's range. Mouth area also plays an important role in pattern control—a particular horn may be designed to give a certain dispersion pattern, but if the mouth is not sufficient to support and project the waveform, the control will fall apart in the lower portion of the system.

There are many ways of coupling loudspeakers, or drivers, to the surrounding air, only a few of which are suitable in professional use. Any good professional system must be efficient and should exhibit superior pattern control and projection ability. These three basic requirements, in our opinion, rule out all of the low efficiency direct radiator hi-fi-type systems and leave two time proven designs: horns and bass-reflex cabinets.

Reflex cabinets attempt to reinforce the lower ranges of a cone speaker by inverting the phase of the back wave by means of a tuned port working with the enclosed cabinet volume. Compactness is one of the chief advantages this design has over bass horns. A reflex cabinet will be considerably smaller than a bass horn capable of the same frequency range. Unfortunately, the cabinet suffers a loss in terms of projection ability and pattern control as well as an efficiency loss of several dB. Reflex cabinets are short-throw devices ideally suited to stage and studio monitor use, instrument amplification, PA in small-to-moderate sized rooms, and in some instances, PA in larger rooms having excellent acoustics. Situations requiring precise pattern control, greater projection ability and/or maximum efficiency are best handled by full horn systems.

There is a measurement used, called "Q," which indicates how well a device projects forward the sound energy available to it. "Q" is directly related to directivity. If all of the sound energy produced by a device were radiated equally in a sphere; if, for instance, a cherry bomb were exploded in a free field, it would have a "Q" of one. It would have no directivity. If that same energy were exploded (radiated) into a hemisphere, the "Q" would double. Project the energy into half *that* space and it would double again. The more energy directed forward at a given frequency, the higher the "Q" at that frequency. The more energy directed forward, the longer the distance the device can project direct sound pressure. Loosely translated, this means that high "Q" devices are more intelligible, more understandable from a greater distance than low "Q" devices. Horns as a general rule will have a higher "Q" than a direct radiator, and, also generally, long horns of narrow angle will have a higher "Q" than short, wide-angle horns.

It might seem that there is a practically infinite variety of horns, but actually they fall into a few definite groups with common design principles and operating characteristics.

There are three categories of bass horns: folded, straight, and combination straight horn and bass-reflex. In large sound systems where size is not a factor to be considered and maximum efficiency and long distance projection is required to get the job done right, the ultimate device to use is a straight bass horn. These giants are used in most of the large, efficient touring systems in use today, in theaters and large hall installations. Outdoors across any appreciable distance a straight bass horn is an absolute necessity, as the lack of room acoustics and the devastating effects of wind increase the need for pattern control and projection ability. When the size of a bass horn is a consideration in your system, as in a large club with limited stage room, a large folded horn might be used.

Folded horns are intended to reduce the bulk of a bass unit by folding the horn path back on itself one or more times, enabling the structure to be shorter than an equivalent straight horn. At very low frequencies these folds cause little problem and the horn will function correctly. As the frequencies go up, however, folded horns run into difficulty because higher frequencies do not go around corners well, consequently reducing efficiency and transient response and increasing distortion—the cause of muddy midrange in folded horns. A large-mouthed folded horn will work well up to 150-200 Hz, but not beyond, so that the system would require the addition of a straight mid-bass horn for proper reproduction above 200 Hz.

The combination horn-reflex system, like the folded horn, is an attempt to reduce the size of a straight bass horn, which requires a large mouth and a low flare rate, by coupling a smaller mid-bass horn with a bass-reflex cabinet. Such a system will function as a reflex cabinet in the LF ranges and as a straight horn in the midrange. It will not suffer the midrange deficiencies of a folded horn; on the other hand it will lack projection, pattern control and efficiency in the lower frequencies. Systems of this combination represent a halfway point between straight bass horns and reflex cabinets—certainly a workable compromise in many situations, such as small clubs, that do not require the extreme projection and efficiency of a straight bass horn.

The choice of which type to use for a particular application will become apparent by considering the size of the system, the size of the typical room in which it will operate, the SPL requirements, and the nature of the program material. This last item would include frequency range and transient capability requirements as well as an overall understanding of the type of sound desired. Typically, horn systems excel at high transient, hard-edged sound, whereas cabinets tend to be soft-edged and more "mellow."

High frequency horns come in a slightly wider variety than bass horns: radial, straight, acoustical lens, multicell and re-entrant. The HF horn that is most popular and enjoys the widest range of applications is the radial horn. Radials are designed to give uniform horizontal coverage over their side-wall angle at all frequencies of operation. Their horizontal pattern has a sharply defined edge, dropping off rapidly beyond the horn angle. Vertical stacking of radial horns raises their "Q," providing a definite increase in projection ability, particularly in the midrange.

Straight HF horns are probably the simplest of all designs available. Straight horns differ from radial horns in that they cannot maintain wide-angle horizontal coverage over their entire frequency range. They tend to narrow their pattern in both planes as the frequency goes up. In some cases this might be a disadvantage, but in other situations the opposite is true. Straight horns have a better on-axis high frequency response than radials because the high frequencies are concentrated in a tighter pattern—an advantage when the application calls for narrow angle of dispersion. Straight horns tend to be more compact than radials of similar flare rate and their flat fronts make box installation very simple. They are ideal for long throw applications. In paging situations where bandwidth is usually limited to the vocal range, straight horns are capable of fairly even coverage over wider angles, and their low distortion levels and excellent projection ability result in high intelligibility and good penetration, even in acoustically adverse environments.

An acoustical lens is not technically a horn in itself, but rather a device attached to a small straight horn in an attempt to broaden its coverage angle. A lens will greatly increase dispersion at high frequencies. Adding a lens to a horn limits its projection capabilities, essentially making it a short-throw, low "Q" device. The fact that a lens is removable can be used to advantage in portable systems, enabling the "Q" of a system to be varied to suit the requirements of the room.

The most direct approach to dispersion control is taken by the multicell—essentially an array of straight horns coupled to a single driver. Multicells provide excellent control and projection throughout their operating range and are widely used in permanently installed systems both in large halls and outdoors. A well designed, well made multicell is an excellent horn

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Christine Kofoed is Executive Vice President of Community Light and Sound. L. Richard Feld is a professional audio retailer at Dimension Five Sound in Philadelphia.

Hot sound vs. hot air.

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

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

Billboard 1978-1979 International Recording Equipment & Studio Directory

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capable of very good performance in a wide range of applications and should be considered whenever a large system is being designed.

Re-entrant horns are essentially folded high frequency horns. As mentioned concerning bass horns, midrange frequencies do not go around corners well; high frequencies do it even less well. Re-entrant horns are prone to high distortion and extremely ragged response. They are typically used in commercial paging situations and even in this undemanding role they do not even come close to the performance of a decent straight horn.

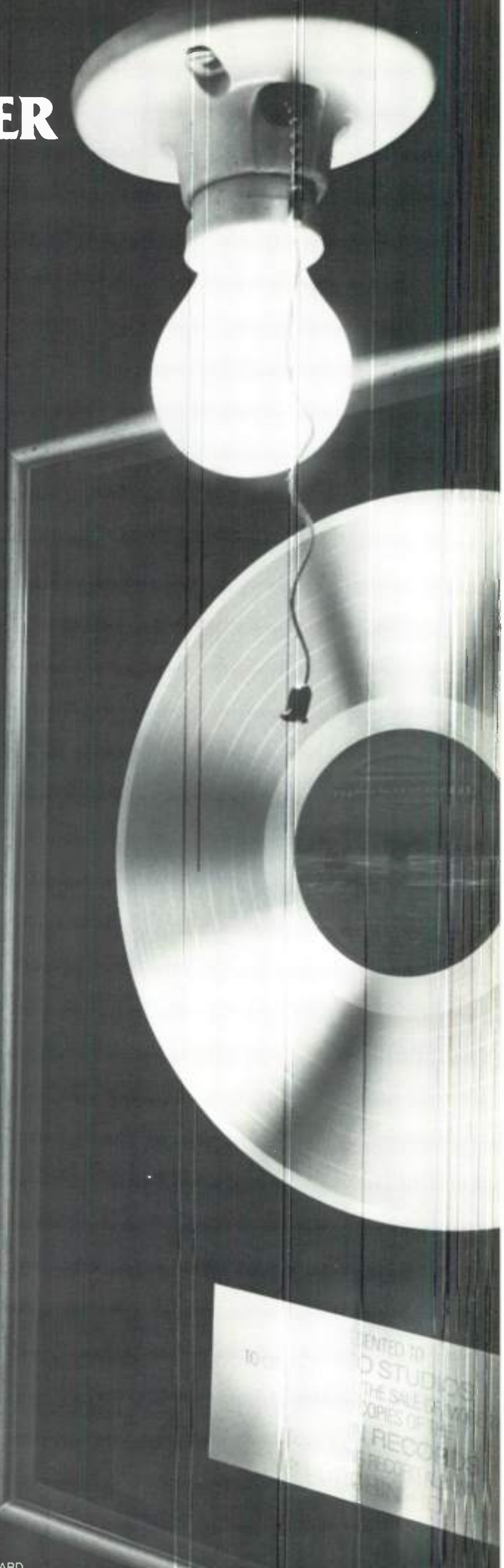
When selecting HF horns for a system, the points to consider are crossover points, SPL requirements, diaphragm size, "Q" or directional characteristics, frequency range, projection distance, and the ratio of HF units to LF units. Crossover point will dictate the lower limit of a horn's operation: It must be remembered that just as with LF horns, the lower the frequency, the larger the horn must be to operate properly. A system crossed over at 500 Hz will need to employ substantially sized HF horns to retain good response and control at crossover, whereas a system crossed over at 1500 Hz can get by fairly well with small HF horns. Just as the low frequency limit dictates horn size, it also dictates driver diaphragm size. Good response at low crossover points requires drivers with large (typically a 4″ diameter) diaphragms. The "Q" and directional characteristics required are determined by several factors: the size and shape of the area of operation; the projection distances involved; the reverberant nature of the area; and the type of LF unit used. Large areas, long throws and reverberant environments require high "Q" horns, such as large sixty degree or forty degree radials and large, straight long-throw horns. Small, non-reverberant rooms usually indicate lower "Q" devices such as ninety degree radials and straight horns with lenses. The "Q" of the HF units should be fairly well matched to that of the LF at crossover. Obviously you would not mate a long-throw HF horn with a reflex cabinet, or put a small ninety degree radial on a large bass horn. In large systems where both near and far field coverage is necessary, as in a large concert hall, a variety of HF and LF horns are used so that the entire audience gets its share of the performance. The ratio of HF units to LF units is usually determined by the efficiency of the LF driver and the crossover point. The more efficient the LF unit, the greater the HF capability required to keep up with it. With LF reflex systems the usual ratio is two LF drivers per HF driver. With LF horn systems the ratio is typically 1:1 or more. Low crossover points require a higher ratio of HF capability, because a greater portion of the energy spectrum is being handled by the HF section.

Horns are never found in very cheap systems because horns are not cheap. It is possible to install an inexpensive 15″ cone, an 8″ cone and a piezo-electric tweeter in a wooden box, creating somewhat of a full-range reproduction system without reaching the cost of a good HF horn alone. On the other hand, you could not possibly do large scale sound reinforcement with your box—or with any number of such boxes. What initially appears to be a bargain quickly depreciates as the scale of the task goes up. In large scale concert systems the cost-effectiveness of horns, not to mention their performance advantages, is universally recognized. Put in simple terms of power efficiency, a good horn will be 5 to 8 dB more efficient than a direct radiator. This means that you would need three to six times more direct radiator power to achieve the same SPL. If only the increase in amplifier power were considered, a typical 10,000 watt horn-type concert system would become a 30,000 to 60,000 watt direct radiator system. At about a dollar a watt the expense mounts up.

In the vast middle ground between the $100 system and the $100,000 system, there will be situations where one or the other approach will be most cost-effective, and often some balance between the two is the right answer. Obviously, sound systems cannot be designed solely on a "dollars in—SPL out" basis, but when considering that aspect of the problem, smaller systems will tend more toward direct radiators and larger systems toward horns.

It is interesting to note that all of the designs discussed were developed and in production before 1930. Some materials have been developed since then that duplicate the original physical design equations more exactly than their forerunners, methods of manufacture have improved and the price has gone up to be sure, but the basic physical process of "moving air" has not changed, and the horns remains.

FOR THE SMALL STUDIO OWNER WITH BIG IDEAS.



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UNLOCK YOUR EARS

A Chip Off The Old Circuit Block

By Mike Beigel

Digital system design is not, by itself, the main factor in the approaching revolution in audio and musical products. The real driving force is Large Scale Integration (LSI), the technology of packing thousands of electronic circuits on a quarter-inch chip of silicon for a couple of dollars. The magic combination of digital techniques and LSI microelectronic methods is due to the repetitive and homogeneous nature of digital circuits and the predisposition of LSI towards multiple similar building blocks. Applying standardized system designs and mass-production economics to the formula yields the science of "Techonomics," which has already given us the pocket calculator, the digital watch and the "TV Game."

Digital products and techniques have already penetrated into the music industry to a large extent, especially in relatively simple applications. The circuitry involved in most electronic organs and pianos, rhythm machines, etc. represents a level of technology and system design not too different from the analog circuits they replace. What I would like to focus on in this article is the "high technology" of digital audio, in which the models and techniques are still developing, and the products which have just begun to penetrate the studio and professional level markets. Essentially, this means digital computer technology applied to audio signals.

THE DIGITAL AUDIO META-SYSTEM

Since computer techniques are being applied to every aspect of audio processing, I will describe a generalized system which would be capable of performing virtually all possible functions on audio signals. Far-fetched? Actually, systems of this kind already exist as research tools in laboratories and universities throughout the world. Though we can't hope to cover more than a few superficial aspects of each component in the system (each subsystem represents a whole field of technological study by itself), we can at least derive an overview of the field. (Fig 1).

The eight major classes of digital audio processing functions, and their relationships, can be seen in the system diagram. In order of appearance, not importance, they are:

Analog audio processes controlled by digital signals. This section might be a computer-controlled mixing board, a programmable analog synthesizer, or digitally controlled analog filter.

Analog-digital conversion. We refer here to the digitization of audio signals (as opposed to control signals). Time-varying analog voltages are converted into discretely sampled digital numbers, and vice-versa.

Digital signal processing. The digital number sequence representing an audio signal is manipulated by "computer" processes and transformed in some way. Simple delay, or reverberation, or pitch change operations are some of the available digital processes.

Digital synthesis. Musical or audio signals are generated in digital format. The available techniques of digital synthesis include harmonic synthesis (of overtone series), FM synthesis (discovered by Chowning), and speech synthesis, among others.

Storage. This includes a particularly wide-ranging conglomeration of devices and techniques: short-term memory system, digital tapes and discs, and solid-state permanent "memories" for storing frequently needed data.

Analysis systems: These can find the pitch and frequency spectrum of a musical tone, identify people by their "voice print," and extract other useful information from the digitized audio signals.

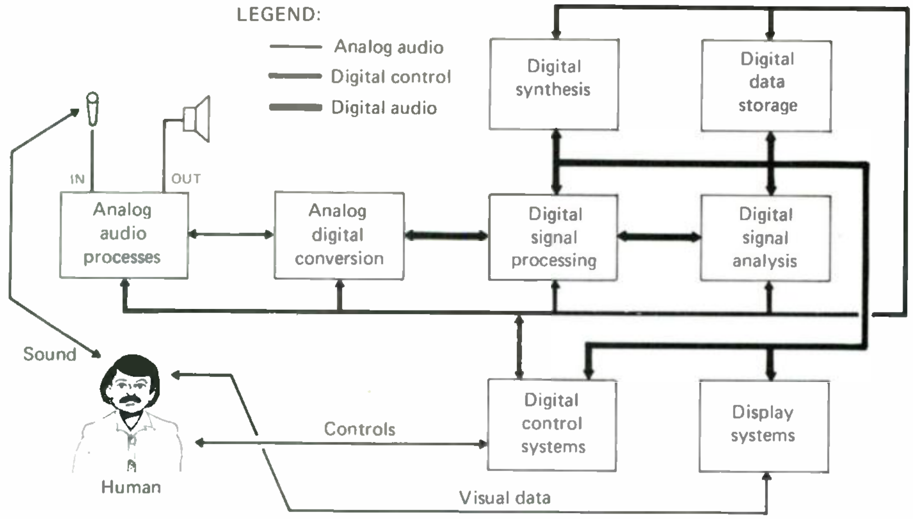
Display systems. A CRT screen, printer, specialized LED bar graph, or numeric readout transmits relevant information to the human operator.

Control systems. Providing an interface between all the processes and a human operator, the control system is likely to be a general-purpose or specialized digital computer attached to all the systems and a meaningful set of controls (e.g. a set of knobs and buttons, a musical keyboard, or other interactive system).

The lines and arrows in the diagrams delineate the flow of digital audio signals, control signals, analog signals, and perceptual data between the various subsystems and the human operator.

EXAMPLE: DIGITAL DELAY

Although the system diagram is generalized (and likely to evoke images of a room full of computer equipment), a close examination of any digital audio product will reveal that it can be



DIGITAL AUDIO META-SYSTEM

Figure 1

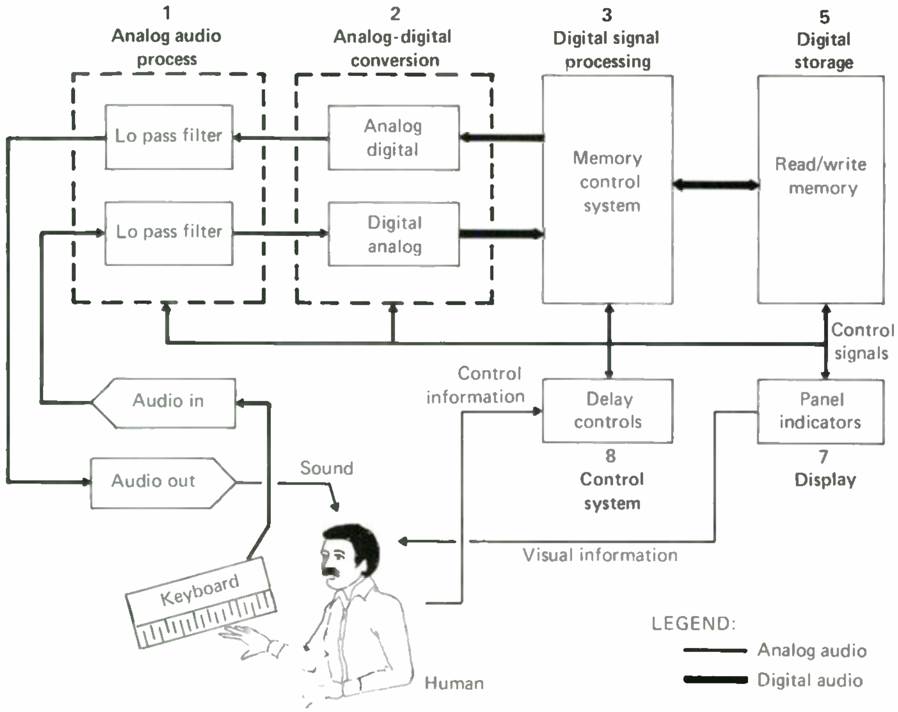
modeled this way. Let's take a basic digital delay system. (Fig. 2)

The audio input signal goes into a low-pass filtering system (1). Since many digital delays don't have enough storage capacity to provide full delay time and full frequency response at once, the filters are controlled by the delay time control (8) to provide frequency response appropriate to the delay time. Likewise, the analog-to-digital and digital-to-analog converter systems (2) may have their conversion rates altered by the control system to provide an appropriate number of conversions per second.

The digital audio data are now managed by the memory management system (3), which is a simple digital processor. It takes each successive input from the analog-to-digital converter, stores it in the memory, keeps track of when to take a delayed sample out of the memory and send it to the digital-to-analog converter. The random access memory (5) provides the storage capability for holding enough samples of digital audio data to create the desired delay length.

To coordinate the operation of the whole product, the front panel controls (7) interface with the control system (8), which in turn coordinates the functioning of the rest of the system elements. As a result, the operator is able to provide an audio signal to the device, control the delay, and listen to the results.

Knowledge of the technical aspects and problems associated with the eight subsystems involved in digital audio processing will be worthwhile. We will re-examine these subsystems in detail in an article to appear in the near future.



DIGITAL DELAY SYSTEM

Figure 2

DIGITAL CONTROL OF ANALOG PROCESSES

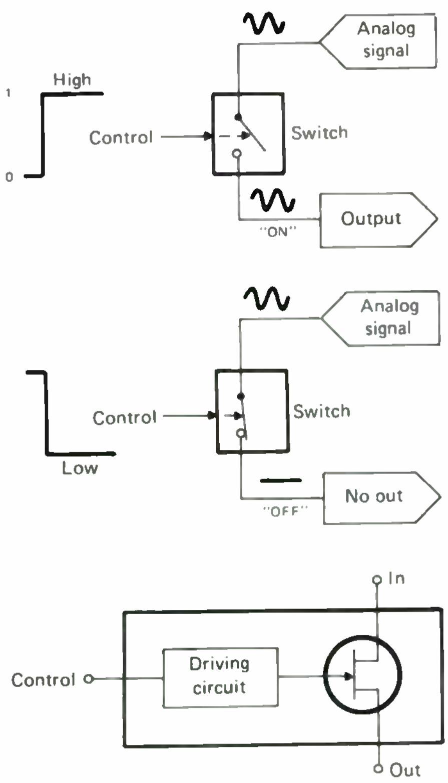
The "synthesizer age" introduced voltage-control to many audio processes. Instead of controlling the whatever-it-is by a switch, potentiometer, or patch cord; a voltage-controllable element (VCA, VCO, VCF, etc.) is used. Digital technology makes use of these voltage-controlled elements by means of analog-digital conversion techniques, and also introduces some new "digital-controlled" analog signal processing elements.

The major classes of digital control generators are: analog switches and multiplexers; sample-and-hold circuits; analog-to-digital (ADC) and digital-to-analog converters (DAC); and multiplying DAC's.

The first category, analog switches and multiplexers, provides the function of "turning things on and off, or sending them someplace." Depending on their price and specification, analog switching devices can transmit positive or negative control signals as well as audio signals. Analog switches are "voltage controlled," but in this case (and almost all cases where digital processing is concerned) the control voltage is either *on* or *off* (*high* or *low*). It may seem like a waste of many good control voltages between *high* and *low*, but it is precisely this waste of information that gives digital processing its utility. Digital information is unambiguous: It is designed not to lose or distort information. The price is putting everything into Highs and Lows, On's and Off's, or 1's and 0's: binary numbers. For our analog switch, a binary "1" closes the switch and a binary "0" opens it. What goes *through* the switch is arbitrary.

Analog switches come in price ranges from $.20 to $20.00, and in all conceivable switching configurations (SPST, SPDT, DPDT, etc.). They differ from mechanical switches mainly in that they have a finite resistance in the *on* position (as opposed to a virtually zero resistance for mechanical switches), and operate over limited ranges of voltage and current. Looking inside, we will find a Junction-FET or MOS FET switching element controlled by some kind of "driving" circuit. (Fig. 3)

An extension of the analog switch concept is the analog multiplexer. This is basically a collection of analog switches controlled by a digital decoding device. Its function is to select one signal from a number of possible choices. Instead of sixteen dif-

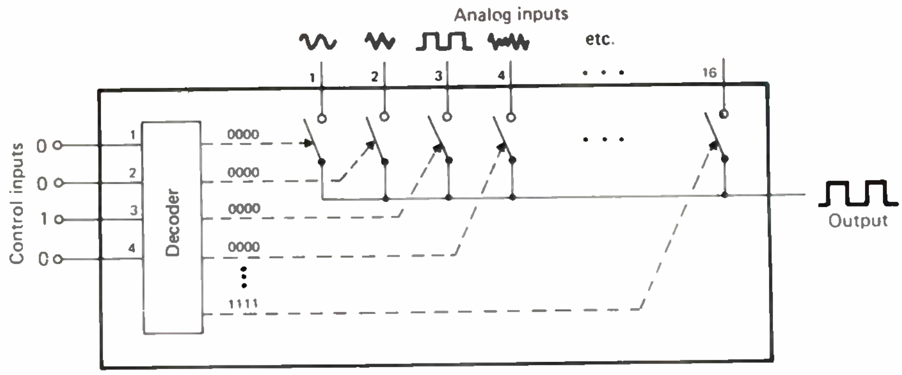


ANALOG SWITCH

Figure 3

ferent switch controls for a sixteen channel unit, the multiplexer requires only *four* digital control signals. The sixteen possible combinations of *on* and *off* each correspond to one of the switches being closed. This means four wires controlling sixteen switches, fewer pins on the integrated circuit package, fewer packages in the system, and less cost all around: basic Techonomics. (Fig. 4)

A sample-and-hold circuit is an analog switch connected to a capacitor, usually followed by a high-impedance operational amplifier and occasionally placed inside an op-amp feedback loop for accuracy and speed. Besides its indispensability in analog-digital con-



SIXTEEN CHANNEL MULTIPLEXER

Figure 4

version, it performs many useful functions as a digital-analog controller. Its basic operation is contained in its name: when the "sample" control is *on*, it acquires the analog value at its input; when the sample control is *off*, it holds that value until another "sample" command is received (or until DC leakage drains the capacitor). (Fig. 5)

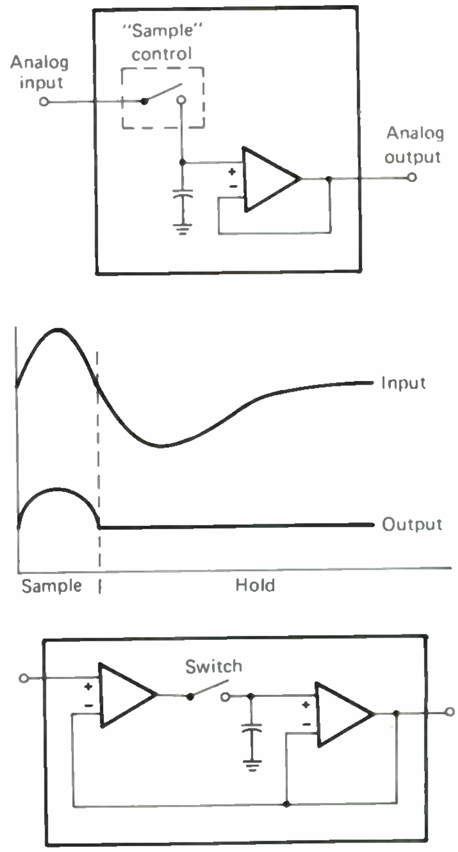
The digital-to-analog converter (DAC) is the fundamental interface component between the "computer" and the "real world." In principle, it is easy to understand.

Computer (binary) numbers are assigned to represent different analog voltages (or currents). The number of different analog voltages depends on the number of binary digits (bits) used in the representation. Let's assume that our DAC delivers an equal voltage step for each binary code; the number of possible voltages is then equal to 2 raised to the power of the number of bits:

Number of bits	Analog voltage steps
1	2
2	4
3	8
4	16
5	32
6	64
7	128
8	256
10	1024
12	4096
14	8192
16	32568

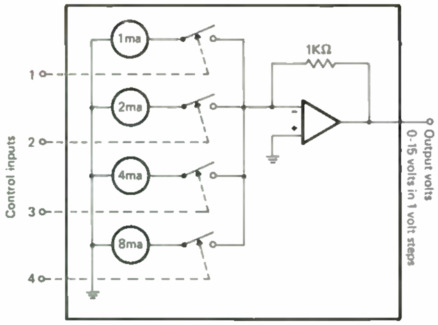
Increasing the number of bits allows the output of the DAC to more closely approximate a continuum of analog outputs. It also increases the cost (especially over 10 bits) dramatically, so DAC's are generally chosen with the smallest number of bits necessary for a given application. If a DAC is controlling a voltage-controlled amplifier, eight bits will be sufficient to provide practically inaudible gradations of loudness. For a voltage-controlled oscillator spanning the audio-frequency range, at least 12 bits will be necessary for acceptable frequency resolution.

The internal construction of a DAC is basically a set of switches and accurate sources of voltage or current. The code at the input activates the appropriate switches, and the currents thus selected are summed to yield the DAC output. A great variety of



SAMPLE HOLD CIRCUITS

Figure 5



SIMPLIFIED DAC SYSTEM

Figure 6

methods are used to provide this function, depending on the accuracy, speed and stability needed. (Fig. 6)

A DAC does not necessarily have equal-sized output-voltage increments. DAC's are available in which the output voltages conform roughly to fixed voltage *ratios*, rather than equal increments. They are commonly called nonlinear or companding DAC's, and are particularly well-suited for controlling VCA's. The DAC voltage increments would correspond to decibels, rather than linear output levels. (Fig. 7)

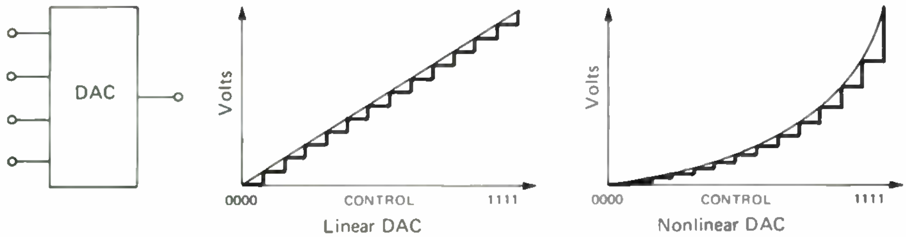


Figure 7

Another useful type of DAC, the multiplying DAC, does not provide fixed output voltages. The output voltage is a digitally variable percentage of an externally-supplied reference voltage. If we allow this reference voltage to be an audio signal, the DAC is essentially a digitally-controlled attenuator. Non-linear multiplying DAC's of this type are very effective digital VCA's. (Fig. 8)

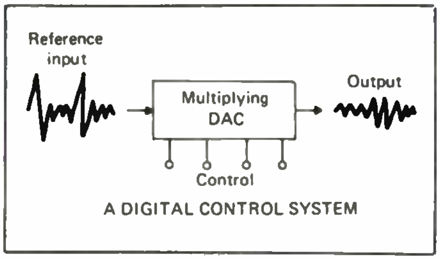
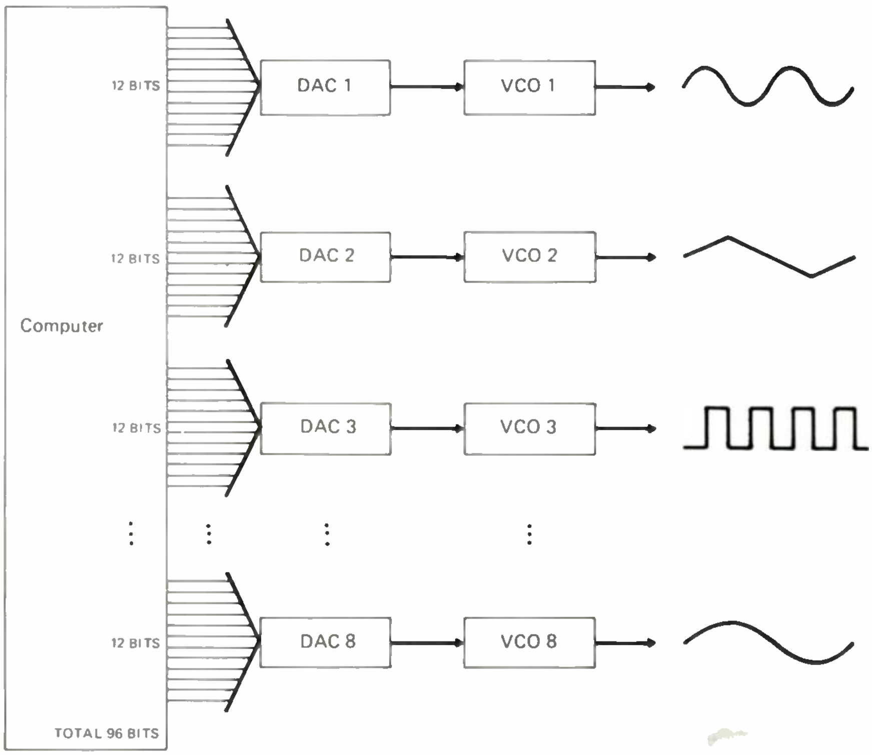


Figure 8

To illustrate the combined usefulness of all the digital control elements just mentioned, let's design a digital controller for some voltage-controlled oscillators in a synthesizer. Let's assume we have eight VCO's and a computer which will provide digital codes corresponding to the control voltage for each oscillator.

One way to accomplish this would be to have the computer provide digital codes to eight different DAC's, each controlling a VCO. This would work, but would be quite expensive and cumbersome. Each DAC requires 12 bits, so the computer has to have 96 wires to supply all the information to the DAC's. Computers, especially the microcomputers which are used to control processes like this, just aren't built that way. Also, 12 bit DAC's cost about $20 each. (Fig. 9)



STRAIGHTFORWARD BUT CUMBERSOME DAC CONTROLLER

Figure 9

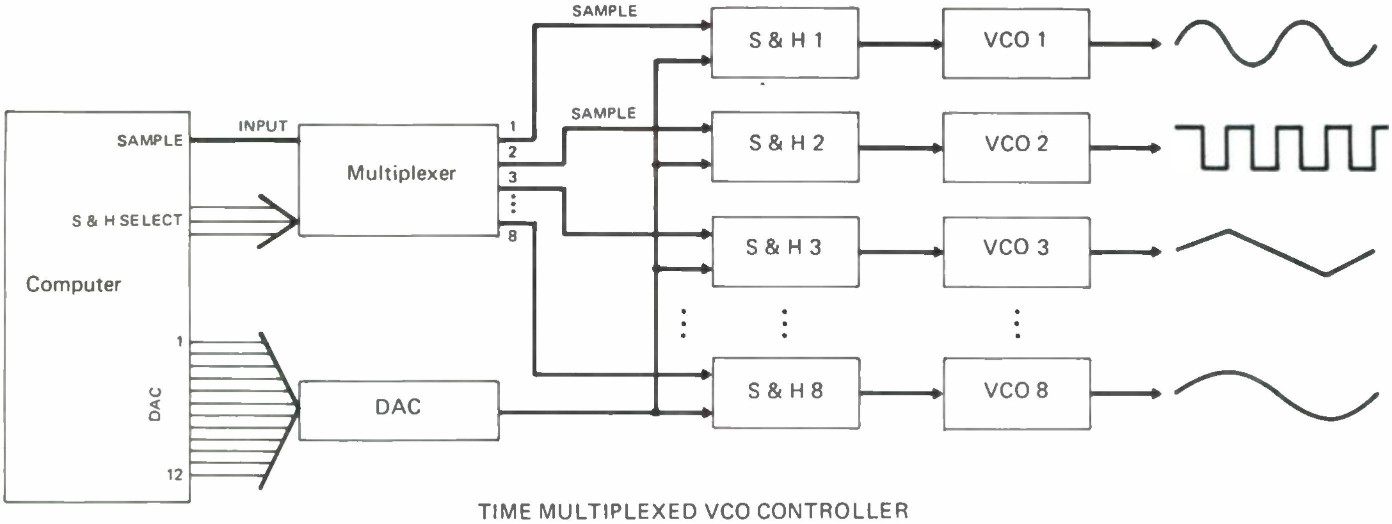
By using a multiplexer and sample-hold amplifiers, a much more efficient and economical system can be made. Multiplexers and sample-holds are much less expensive than 12 bit DAC's (this year), but more important—they simplify the system.

In the new system, the computer needs only sixteen output lines: twelve specify the output code to the single 12 bit DAC, and the other three control the inputs to an eight channel multiplexer. The extra line supplies a "sample" signal from the computer to whichever sample-hold is selected by the multiplexer.

In this system, the computer outputs a sequential series of codes to the DAC, corresponding to each control voltage for the VCO's. At the same time, the computer puts a code into the multiplexer, selecting the sample-hold corresponding to the appropriate VCO. A "sample" pulse is then sent to the sample-hold, and the output from the DAC is "latched" into the sample-hold. The "sample" pulse is then removed, and the computer and DAC supply the rest of the sample-holds in the system.

The sequence is repeated at a rate fast enough that each VCO is perceptually "up to date." Usually a repetition rate of 250 system updates per second is sufficient. This type of system is called a "time-multiplexed" system: a number of different channels of information are transmitted over a single data line, each in its own "time slot." A "sequential circuit," so to speak is created. (Fig. 10)

This article, and the article to follow, are attempts to acquaint SOUND ARTS readers with all the aspects of



TIME MULTIPLEXED VCO CONTROLLER

Figure 10

digital audio technology. The impact of digital electronics will affect every area of the musical electronics field: new kinds of products, specifications, performance features and applications. New equipment will be required in the repair shop, as well as knowledge of how to operate and service these new "wonder-boxes."

I remember when synthesizers first came out. The only people who knew about them were the manufacturers and the customers. The music stores were in the middle of it, wondering what was going on and whether they could make money on it. That was almost a decade ago, and you know the results. The same thing is happening again, technologically, with digital equipment. This time, I believe the music retailing industry is ready for it and ready to learn.

Next month we will continue with the "Digitization of Audio Signals."

THE RETAILER COPES WITH DISASTER

By Allen Hester

Saturday night, July 1, 1978: The paging system at the Orpheum Theatre in Memphis crackled: "Chris Lovell, you have an emergency phone call in the front office. Chris Lovell, come to the front office, please."

As the local rock concert continued, Chris and his friend, Tommy Stinson, ran to the phone. Chris's wife, Bonnie, was on the line.

"Chris, the store is on fire. You'd better get over there in a hurry."

As Chris and Tommy got in the car and headed east on Union Avenue, they realized what was happening. The Memphis Fire Department union members were out on strike. As Chris and Tommy sped down Union, it seemed to them that the entire city was on fire.

"It looked like the city had been fire-bombed," Chris said later. "Everywhere we looked, there were huge flames flaring in the air. I felt like I was in another world or something. I had never seen anything like that before, never even imagined it. But there it was . . . fire everywhere."

When Chris and Tommy got to 205 S. Cooper, Chris's store—Strings and Things—was burning completely out of control. Of course, because of the strike, there were no firemen to fight the fire, and the volunteers from a neighborhood bar nearby were struggling with fire hoses in a brave but vain attempt to quench the blaze. The National Guardsmen who were in charge of manning the fire trucks couldn't do very much either, since they had no fire-fighting training to speak of, and their main concern was to keep people from going near the building and getting hurt.

Chris made his way through the crowd and identified himself to the guardsmen. With the help of someone in the crowd he managed to break in the front door of the store, but the heat was so intense that Chris could only go a few feet before turning back.

Meanwhile, at the northeast corner of the store, several people were trying to pry the safety bars off the display windows in front of the antique guitar collection, one of the best collections of old and rare guitars and amps in the entire south, and the pride of Strings and Things.

Chris, after failing to get inside through the front door, ran down the sidewalk and joined the group of people in front of the collection. When he reached through a broken window to grab an old Epiphone archtop guitar, a piece of jagged glass fell across his wrist, cutting it to the bone. Immediately, a guardsman grabbed him and put Chris in an ambulance headed for the hospital. Fortunately, the people at the scene were able to get some of the old guitars out before the fire got to them.

Ninety miles away in Dyersburg, Tennessee, Charlie Lawing had just finished playing a gig with his band when the bartender called him over to the phone and said, "The sheriff's on the line. Wants to speak to you." Charlie answered it.

"Mr. Lawing, your wife notified us that your business in Memphis is on fire, and advised that you come home right away."

"When I got home," Charlie said, "the store was burning pretty good. At first I was told that only the warehouse was burning, and that it wasn't so bad; but when I got there Chris had already been taken to the hospital, my wife Beth was in a state of near-hysteria, and there wasn't much I could do. So I stood there and watched six years of my life go up in smoke."

The business that Chris Lovell and Charlie Lawing had built from scratch was totally destroyed. Only a few of the old guitars in the collection were salvaged.

THE AFTERMATH

There are many more stories about the fire that night, but the most important thing about the disaster as it pertains to other dealers across the country is what happened *after* the fire. How did Strings and Things survive such a devastating blow? What, and who, got them back on their feet?

The total loss of inventory was in excess of $300,000 in new and used merchandise (including many new products just brought home from the summer NAMM show in June), office equipment and supplies, personal items, and $14,000 in cash that was to be deposited the following Monday.

An entire city block, which housed not only Strings and Things but a restaurant, an insurance office, and a printing shop, was left in a smoldering heap that burned off and on for a solid week before it finally died out. Several weeks later, fire department investigators determined that the cause of the fire was arson.

As soon as Chris was released from the emergency room on Sunday, July 2, he went home and began compiling a list of everything the store would need to stay in operation. All the manufacturers were contacted the next day; the situation was explained; and each manufacturer, without exception, agreed to send the necessary merchandise on open account. Most of them did not even ask if the store was insured. Their confidence in Strings and Things was so strong that they began shipping immediately.

Temporary headquarters were set up in Charlie Lawing's home. For two weeks, deliveries, phone calls, and visits from friends and company representatives did not cease. Meanwhile, a search was underway for a new location for Strings and Things.

Commerce Union Bank official Robert Montgomery was immediately notified of the situation, and he extended an open line of credit to Strings and Things so that all immediate expenses could be covered pending the settlement from the insurance company.

Jeff Michaels, the Memphis insurance agent through whom the Strings and Things policy was purchased from Insurance Company of North America (INA), contacted INA and requested that a company representative be sent to Memphis immediately to assess the situation. Gene Smith, a large claims specialist from the Atlanta office of INA, came and talked to Chris and Charlie.

"Gene was very straightforward with us," Chris said. "He told us that as soon as his company could investigate the claim and verify the losses we incurred, INA would pay the claim. He also cautioned us that if there was any evidence of wrongdoing on our part, he wasn't going to pay us a dime! But Gene gave us straight answers to all our questions, and simply said that INA had rules to follow, and that they would be followed. Mr. Smith made it clear that INA would play the game fair, but that Strings and Things had to verify its losses before anything could be done.

That verification was a long and tedious job, because there were no duplicate records of inventory other than the ones inside the store at the time of the fire. So the secretaries had to call each company and request copies of invoices from the past several months. That took a lot of time, but the open line of credit from Commerce Union Bank kept the ball rolling.

After two weeks of operating out of Charlie's house, the crew moved into a vacant building across the street from the old store. There, for seven weeks, Strings and Things began to make its comeback. A strong advertising campaign, centered around the slogan "We Got Off Our Ashes!", was undertaken to let the public know that Strings and Things was not dead. Drum Stand manager Dave Patrick had sets of drums set up out on the sidewalk . . . he even made a few sales right there on the street!

During this seven week period, Chris and Charlie had been searching for a new permanent location. When they found the building they wanted at 1492 Union Avenue, the building was for *rent only*, but the owner of the property offered to let them buy the property at a very good price. With help from their attorney, Robert A. Wampler, and Bob Montgomery at Commerce Union, Chris and Charlie were able to purchase the property.

Remodeling the new building began the day after the purchase agreement was made. A makeshift but very dedicated crew of carpenters, friends, customers, and local bands worked long and hard to get the new building ready for opening day. Charlie supervised the remodeling, while Chris directed sales at the temporary store.

On September 9, Strings and Things held its grand opening, just nine weeks after being burned to the ground.

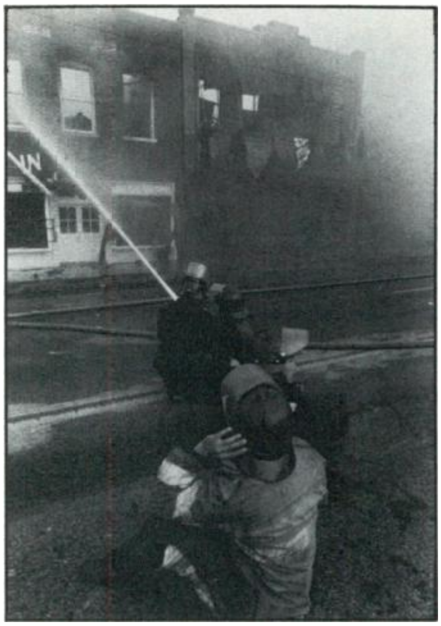
IN RETROSPECT

Looking back at the fire and the aftermath, Chris and Charlie realize how fortunate they were to have a good rapport with their lawyer, banker, insurance agent, employees, and the manufacturers whose products they stocked and sold.

"Bobby Wampler, our attorney, really held us together," Charlie observed. "He was there fighting the fire; he was with us the next day; he told us what we had to do and made sure we did it. He is much more than just our attorney, he is a true friend. We realize that now more than ever."

Chris praised the banker, Bob Montgomery. "We had just moved our business account over to Commerce Union. I feel that if we had been with our old bank, we would not have gotten the personal attention that Bob gave us. We were in constant touch with him, and he backed us all the way, even when people were screaming in his face and ours, too."

"Don't forget (insurance agent) Jeff Michaels," Charlie said. "He did as much as anyone. He helped us negotiate our claim with INA and kept us informed as to their position regarding our insurance coverage. It would have been easy enough for him to say, 'Look, it's out of my hands,' but he



didn't. He stuck with us."

"It is really hard to say who or what was *the* most important element in our comeback," Chris added. "Without the help of the manufacturers who extended open accounts to us, we wouldn't have had anything to sell. And without a group of employees who were willing to work for almost nothing there for a while, we surely would have gone under. With no merchandise and no crew, it would have been all over. The 'down time' would have killed us. Our competition could have stepped in and taken our business away. We were just very lucky that everything fell into place as it did. If one, just *one*, of those elements we mentioned had not been in our favor, it's quite certain that we wouldn't be in business today."

LIVE AND LEARN

What did these two guys learn from their experience? Plenty. Here is a list of things that Chris and Charlie feel are essential to any dealer who wants to stay in business and protect himself from disaster:

Cultivate dealer-manufacturer rapport. Take the time to show the reps around your place; let them know *you* and what you are doing to promote your business and their products. It takes a little time, but it's worth it.

Find dedicated employees. Sometimes this is hard to do, but you need people who will stick with you through thick and thin. Give them something extra when you can, make them feel like they are an important part of your operation.

Get a good lawyer. This can make you or break you in a hurry. Find a man with a sound reputation, not some slick talker who is only out for a buck. Every dealer needs a lawyer who is on his side at all times. Where would Muhammed Ali be today without Angelo Dundee in his corner giving good advice at the right time?

Find a responsive banker. It is essential that you always honor your word at the bank. Let the banker know that you are determined to succeed and that you want him to be a part of that success. If your banker believes in you, he will be willing to gamble with you, because let's face it, banks want to make money just like every other business. But nobody bets on a loser, least of all a banker.

Know your insurance policy! This above all else is imperative. Sit down with your agent and have him explain step-by-step what would happen if your business burned down. Don't wait until it happens and *then* say "What now?" Make sure that you understand *each clause* in your policy. Inventory coverage is not enough. You've got to have coverage on such things as signs and billboards; outdoor awnings; cash on hand; landscaping, trees and shrubberies; commercial vehicles; business interruption; and any number of other items that may not be a part of your general policy.

Strings and Things found these things out the hard way. Maybe you won't have to.

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

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

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

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

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

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

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CIRCLE 90 ON READER SERVICE CARD

The SOUND SH

Schecter is a California-based company that specializes in custom replacement parts for Fender Strats, Telecasters, Precision and Jazz basses. At the winter NAMM show, Schecter introduced the latest addition to their distinguished line of custom parts—a Les Paul-type wiring harness and pickup assembly, which the manufacturer has named the "Z-Plus Humbucking Assembly."

Dan Armstrong, a pioneer in pickup designs and guitar building, designed these new humbuckings exclusively for Schecter. The Z-Plus is a complete replacement electronics kit designed to fit into a Les Paul body with *no* extra routing, drilling, or body modification of any kind. No extra switches or knobs are required to make the Z-Plus fully operational in any Les Paul (or any electric guitar with similar pickup and wiring configuration).

The kit comes prewired with four Omni Pots (made by Allen-Bradley) on a brass grounding shield. The wiring harness is color-coded for easy installation by anyone with basic soldering skills.

Since each omni pot is a double-pole, double-throw switch, there are a total of 26 different tone combinations possible with the Z-Plus assembly. The player can choose between double and single coils, switch to series-parallel or an out-of-phase mode in various combinations, as well as the standard mode.

CIRCLE 1 ON READER SERVICE CARD

Sound Workshop has introduced the 262 Stereo Reverberation System, bringing "new performance standards to reverb systems in the under $1,000 category." Two channels of EQ are provided, with each channel allowing a plus and minus 15 dB range over the high and low frequency bands. Frequency selection is fully sweepable from 50 Hz to 1 kHz (low band) and from 500 Hz to 10 kHz (high band). The EQ bandwidth is optimized for proper contouring of the reverberant signal. The unit also allows dry/wet mixing, full drive level into 600 ohm loads, LED level indicators, active balanced inputs, and matched bi-FET preamps for ultra-low noise performance. It mounts in a 3½ inch rack space and sells for $700. It is also available with transformer-balanced outputs and XLR connectors at a suggested retail of $750.

CIRCLE 2 ON READER SERVICE CARD

Sound travels relatively slowly. This physical reality can be both a blessing and a curse. In a good concert hall, the hall's resonance and reverberation hang in the air after each note, thus enhancing the sound of the music. However, when amplified sound is used, time lags become a problem, because the amplified sound reaches the listener either ahead of or behind the direct sound. The result is a disturbing echo, and when several speakers are used, it gets even worse.

Electronic digital delay units synchronize the live and amplified sounds, but the usefulness of a digital only begins with solving this sort of acoustic problem; many other applications are available to the owner of such a unit. For example, the **DeltaLab** DL-1 boasts no less than a dozen uses: doubling, mono-stereo synthesis, hard reverberation, delayed echo send, Haas-effect image localization, chorus effect, enhanced spatial ambience, recursive filtering (tuned feedback resonance), comb filtering (fixed flanging), OPPE

By Charlie Lawing

multiple echoes, and vocal thickening and spreading.

The DL-1 has three separate outputs; thus it is capable of simultaneous delays of varying length. This is essential in a large room, one that has a balcony, or an outdoor concert arena. Short delays are used up close; longer delays are used in the rear of the area; and the end result is that everyone present hears the same thing at the same time. The DL-1 is

capable of delays up to 160 ms, enough for all but the largest concert environments. (Further delay can be achieved by using more than one delay unit). And the DL-1 is tamper-proof, because it has a set of internal controls that override the front panel controls. So the installer can measure the proper distance-to-delay ratio by consulting the owner's manual, set the DL-1 accordingly, and leave the area without having to worry about the house electrician going bananas over a new set of knobs to turn. Great for permanent installations in churches or theaters where customer satisfaction is essential and nobody knows what you are talking about when you say "acoustics." Been there before, have you? Well so have I, and that's why I like the tamper-proof feature of this unit.

Some of the other applications of the DL-1 will interest you. In a disco system, adding delayed side speakers to the main system can really "open up" the sound, making it more spacious, and adding a punch to otherwise dry sounding music.

For a vocal track, either live or in the studio, a doubling effect (achieved by using a delay of 15 to 30 ms) thickens the sound, adds depth and makes it louder without significantly raising the meter levels. By panning the image to the left or right of the original signal, the effect is further broadened. Triple-tracking is available also.

When recording guitar or string tracks, the double and triple-tracking effect will make five strings sound like fifteen (the often-heard-of "chorus effect" that is in demand among musicians today.) Slapback echo is no problem, either. Just delay about 50 ms or so and you've got it.

One of the little tricks that musicians have come up with when using a digital unit is to delay slightly the feed to the main speakers or to the monitors, which will reduce feedback and improve clarity at high sound levels.

DeltaLab will sell the Owner's Manual for the DL-1 for $5, which is refunded when the unit is bought.

CIRCLE 3 ON READER SERVICE CARD

Polyfusion's FP-2 is a battery operated active volume control pedal. Features include a high input impedance buffer, an output line driver amplifier and a low noise potentiometer. The unit is constructed of heavy gauge steel and operates on one 9 V alkaline battery. Suggested retail price is $69.95, with a one year limited warranty.

CIRCLE 4 ON READER SERVICE CARD

As I've said before in this column, many of my customers are sacrificing huge amps in favor of improving their overall sound reinforcement gear. The trend is definitely toward a quality sound system and a knowledgeable person to operate it, but of course, sound gear is expensive and bands often have to buy their systems a piece at a time. That's why my partner and I chose **Tangent** mixers as one of our lines. Tangent offers good specs and features at a price people can live with. The new 1602a is no exception.

The 1602a has a Solo Function which enables the sound man to monitor any input or preview an entire grouping, such as vocals or a horn section. By depressing the solo button, the channel is automatically fed into the headphones, regardless of what had been there beforehand.

Three separate effects sends—Effects, Monitor, and Reverb—give the operator added flexibility by acting as three independent mixers within the unit.

On each channel, a pair of access jacks lets the user patch in external effects to that channel alone. This is ideal for using echo units or delay lines on vocalists, guitarists, or drums for special effects controlled from the console. Any time you can eliminate a wasted motion on stage by manipulating effects controls from the console, you decrease the chance of an on-stage error by a musician whose concentration may be affected by frequent knob-turning.

Another good feature of the 1602a is its modularity. Each channel can be separately removed from the console in the event of a malfunction, thereby preventing the entire board from being inoperative while one faulty IC is being replaced.

Rather than relying on pads, the Variable Gain Control varies the gain of the microphone pre-amp over a 40 dB range.

Other features such as internal reverb, balanced inputs and outputs, expandibility, and phantom power capability are standard on the Tangent 1602a.

Spec-wise, the unit delivers a noise level of −128.5 dB and has a THD reading of less than 0.004% at 1 kHz. The slew rate is a minimum of 10 volts per micro-second at any point in the audio chain.

CIRCLE 5 ON READER SERVICE CARD

The **MXR** effects line, we have found at Strings and Things, has a terrific performance record. MXR units are consistently durable and clean, and the fact that several other companies have marketed similar effects that were almost carbon copies of the MXR line should tell you a lot. MXR units are in high demand everywhere.

Last year, MXR introduced its Digital Delay Line, which provided studio flanging and time delays in a portable, rack mountable unit that retailed for $995. The customer response was very favorable, but a lot of people couldn't afford the digital delay at that price. So, being the responsive young designers that they are, the MXR people went back to the drawing board and designed a similar but less expensive unit, which they call the Flanger/Doubler.

This new signal-processing device can produce several varieties of flanging, hard reverberation, doubling, and some subtle chorus effects, as well.

The time delay range in the flanging mode is from .25 to 5 milliseconds; time delay in the doubling mode is from 17.5 to 70 milliseconds. The Flanger/Doubler can be instantly switched from one mode to the other.

The Flanger/Doubler features a manual delay time control, a mix control (between the dry and delayed signal), width and speed sweep controls, and a regeneration control.

It is equally useful in the studio or on stage. It has instrument level inputs and outputs on the front panel for on-stage use, and line level ins and outs on the rear panel for permanent studio installation.

Easy-to-see LED indicators on the front panel indicate whether the power is on, the

The SOUND SHOPPE
REAR ENTRANCE

effects in or out, and the length of the sweep being used. On the rear panel, voltage-controlled terminals make external delay control possible from an outside source such as a synthesizer, as well as the capacity to hook two units together for stereo.

The Flanger/Doubler was introduced to the industry at the NAMM Winter Market in Anaheim, California in late January. At a list

price of $425, the MXR Flanger/Doubler is well within the range of the working musician. Many echo units now in production cost more than the F/D, and to my way of thinking the MXR unit is far more versatile and useful than a simple echo.

I have had nothing but good luck with the MXR line in my shop; there has only been one unit they ever built that I wasn't completely satisfied with, and the only complaint I had about that unit was the noise it generated. I've sold thousands of their phasers alone, and I expect to do well with the new Flanger/Doubler.

CIRCLE 6 ON READER SERVICE CARD

Teac has just released a new 4-track recorder, the A-3440, which is essentially the same as the older model 3440 as far as appearance goes, but the A-3440 is improved in several significant ways.

One of the most exasperating and time-consuming aspects of home recording is the endless battle that the musician wages with cords and cables of various description. It takes time to set up for one function, then more time to replug and re-adjust for something else, and even more time to play back what has just been put on tape.

Teac, having realized how frustrating all this wire wrangling can be to a musician, has internalized a variety of operations into a Function Select Panel on the A-3440. This panel of push-buttons effectively eliminates all the back-and-forth manipulation of wiring, thereby cutting down on the time it takes to record, as well as the unnecessary effort.

With the output select button in the "sync" position, the Function Select button takes control. Put the Function Select button in the "out" position and you can monitor the tape in the sync mode. Push the Function Select button in, activate the "Record" button, and you can monitor your source for any given channel while recording.

The A-3440 has a new monitoring setup, also. The user can listen through the headphones to one or all of the four tracks, in any combination, while recording. Although the mix is mono through the phones, there is an independent level control for the loudness of the monitor mix furnished.

Another very convenient feature of the A-3440 is the Pitch Control. With this feature, the recording engineer can speed up or slow down the tape enough to put a fixed-pitch instrument in tune with other instruments already on the tape. This eliminates having to stop in mid-session to re-tune an electric piano, just for an example. Also, the Pitch Control can add some nice special effects to a track when used judiciously.

The A-3440 is built to accept any professional quality dbx noise reduction unit, which can enhance the S/N ratio by up to 30 dB.

Of course, each channel has its own VU meter and mic line switch, as well as a 20 dB mic attenuator which prevents overload distortion. Individual input and output levels are included on each channel as well.

The A-3440 will accept four unbalanced high- or low-impedance microphones. Manual cueing control makes it easier to isolate any spot on the tape for cueing and/or editing.

An optional remote control unit (RC-70) is available for the A-3440.

CIRCLE 7 ON READER SERVICE CARD

By Tom Morse

DEALER DOSSIER

Daddy's Junky Music Store, New Hampshire

Daddy's Junky Music Store opened in 1972 in humble surroundings. Its growth since then has been fortunate as well as calculated, and now there are three locations. The Music Mall in Salem, New Hampshire, the largest of the stores, boasts a sound stage for in-store performances and demonstrations; a music museum; and a sound room. Begun on a very small basis, Daddy's represents a dream of Fred Bramante. We met with Bramante to investigate the philosophy and workings of Daddy's Junky Music Stores.

How many years have you been in the music business?

Bramante: Well, I opened my first store in 1972, about six and a half years ago. Actually, I've been in music longer than that.

Doing what?

Bramante: I first got into music in college. I was a bass player and vocalist with a band. Anyway, I was practicing with the band one night and one of the guys was talking about a friend who needed to sell a guitar pretty quick to raise bail money for someone. I had a little extra cash so I bought it. I fixed it up and sold it for a profit. Then I started looking for more instruments to buy.

Where did you go from there?

Bramante: After I graduated, I moved to Connecticut. I had been hired to teach science in Stamford, Connecticut. By then I already had a wife and two children, and my teaching salary didn't exactly allow us to live in luxury. And I kept buying instruments.

That small start obviously paid off.

Bramante: It was slow going at first. I couldn't afford to buy much. Sometimes I'd sell a guitar right away. Sometimes I'd have one sitting around for weeks. I was taking guitar lessons from Larry Coryell back then; he was living in Rowayton, Connecticut, not far from my home. Every week I'd walk into my lesson with a different guitar. He didn't realize I had a little business going and advised me to find one guitar I liked and stick with it.

How did all this actually become a business?

Bramante: I had always planned on opening a store of my own. But my money situation had made it impossible. But I kept an eye out for cheap rents just in case something would break. I saw an ad for a store location at $75 a month including utilities in downtown Norwalk. The time was right, the price was right, so I went ahead with it. I had saved about $600 to open the store with.

That was your initial investment?

Bramante: Yeah, $600. So I really couldn't afford much of an overhead. And the place I rented—mess is an understatement. It was a hole when we first started to get it ready to move in. Plaster falling from the ceiling, dead pigeons on the floor, mice and other creatures crawling all over the place. Every day after school, my family would go to the store with me to clean. Some days a few of my students would come along to help. And at the end of the day my kids, Michael, then 3, and Candi, then 2, would be absolutely filthy. One day Michael asked, "Are we going to Daddy's junky store today?" I loved the sound, so that's what I named my store.

With a $600 investment, what did your stock the store with?

Bramante: We opened in September, 1972, with eight used guitars, about four used amps, one set of used drums. We spent $15 on a cash register, $400 on inexpensive, new, folk guitars, $50 on a display case, $30 for a guitar rack, and we spend the rest on accessories. And we were only opened part-time. A high school kid that worked on commission would open up at two-

A great vocal microphone must do just two things:

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The result is honest, very musical sound. Vocals with punch and clarity—a direct result of our frequency-aligned response. The ATM31 curve takes into account every element in the chain...voice, amps, and speakers. It's the same kind of sound you hear on the finest recordings, but delivered on the road, day after day, in concerts and club dates alike.

As for survival, take a close look at one example of ATM31 "Road Tough" construction: the windscreen. Not simply woven wire, but *three* layers of screen. A heavy outer wire mesh, a finer inner mesh, and finally a fine brass screen. All soldered firmly in place (others use cheaper epoxy, but it can get brittle and fail at absolutely the worst times).

Every other detail of the ATM31 is as carefully engineered for performance and long life. This is one vocal microphone which will stay new-looking and new-sounding long after others are showing their distress.

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CIRCLE 98 ON READER SERVICE CARD

thirty and then after school I would come down until about six o'clock. Saturdays we were open from eleven to four. But I was convinced that my idea would work, and it did. We were little but we developed some really good customers who kept coming back again and again.

Where did you go from there?

Bramante: Well, on weekends and during the summer, I would visit my parents in Salem, New Hampshire. One weekend I went to listen to a fifties band, "Gunga and the Dins." They were fabulous. And one of the singers did a great DJ act, "Good Guy, Guy Goodly." I went back almost every week to see them and I got to know the band. "Good Guy," also known as Chris Gleason, is my partner today. He opened "Daddy's" in Salem in December, 1973.

Was this a "junky" store?

Bramante: Actually we'd come a long way, but we still needed to find an inexpensive location, so we were situated in half of a gas station building.

What next?

Bramante: The next year we opened a store in Manchester, expanded and moved the Salem and Norwalk stores, and in 1976, we opened our Portsmouth store. That was our first really large and renovated store. By that time the business was growing at an almost awesome pace. So I resigned my teaching position, sold the Connecticut store, and moved to New Hampshire to run the stores full-time. And then, last year we put up our own building in Salem. We call it the "Music Mall."

What do you attribute the growth you've achieved to?

Bramante: Our philosophy. I started the business with the philosophy that a good deal on a used instrument is a good deal, period. An instrument suffers its greatest loss in value the moment it is purchased by its first owner. It's no longer new. But a quality used instrument, if kept in fine shape, will rarely lose value, and may even increase in value.

Do you offer new instruments, too?

Bramante: Yes, but only a few select lines. As we grew, we ran into a supply and demand problem—too many customers, which we sure didn't mind, but too few instruments and amps. So we began researching guitar and amplifier companies to find the ones that would fit into our philosophy of offering the best dollar for dollar value. We have turned down some of the most famous names in the business because they didn't fit into that philosophy.

Who didn't turn you down?

Bramante: For the most part with the lines we carry, we did the chasing, because after our study we were so convinced that they were the best for the money, we wouldn't settle for anything less. We carry Peavey, Biamp, Ovation, Hondo II, Schecter Guitar Research, MXR, Electro-Harmonix, Shure, DiMarzio, Gurian, Conn, Intersound, a few others. All of these fit into our philosophy.

What else do you offer your customers?

Bramante: We have super electronics and guitar repair departments. Even when we were very small we had several excellent people doing repairs. Our Salem service lab for electronic repairs was custom-built by Mark Rogers, our chief technician. He has a ten-bay amplifier test setup which runs up to ten amplifiers after they have been repaired and sequentially tests each one. We have a unit which automatically tests sustain units. Another unique and very versatile unit is our central patch panel. This is a panel that is located at the service bench and has all of the test equipment inputs and outputs on it, plus all of the inputs and outputs from the two equipment rack consoles at the far end of the lab brought out to it. This allows quick and easy interfacing of any of the pieces of test gear with each other and the unit being worked on with a minimum of wires and cables.

We also have a separate shop set up with all our power tools and that's enabled our guitar repairmen to more efficiently organize the guitar shop. We also do brass and woodwind repairs. We have a total of three shops in the Salem store, two in Portsmouth, and we'll soon have one in our new Manchester store. We relocated in Manchester in October, 1978.

Let's talk about your Salem store. You've described your first store and the gas station. Tell me more about the "Music Mall."

Bramante: We designed the building ourselves, working with an architect. It was planned to be a music store from the word go. We labored over every electrical outlet, every light panel. The store has features you won't see in many stores and some you won't see anywhere else.

We have a beautiful instrument museum, beams throughout the store which display our guitars and string instruments, a set of 1930s cathedral chimes, a sound room, seats built around wooden posts with headsets so a customer can try an electric guitar without bothering to plug into an amp.

That way he doesn't disturb other customers.

Bramante: That's part of it. It works two ways. It's also good for beginning guitarists, buying their first electric. Sometimes a beginner would like to try out a guitar, really get the feel of it, but he's self-conscious because he doesn't play well yet. This headset allows him the privacy to do what he wants without that self-consciousness.

What other features are in the store?

Bramante: We have an amplifier demonstrator that has the capability of interfacing up to ten mixer amps, six stereo boards, six mono boards, six stereo power amps, and six mono power amps. It also lets us compare up to three different microphones at one time, through the same channel of any amplifier. It will also handle up to 22 pairs of speakers.

Instead of going through a big production to try our different combinations, all you have to do is plug in your guitar or mic and flip a few switches. We can set up any combination of equipment in less than 10-15 seconds, so it's great for letting customers compare units. In fact, the only problem is that our pro-audio department is growing so fast, we are outgrowing the demonstrator.

Right now, Mark Rogers, the head technician I mentioned, is designing and building a larger and more versatile unit. The new demonstrator will be able to handle three times as many power amps and mixing consoles. It will also have input-output buses for units like graphic equalizers, digital delays, and so on.

If you only carry a couple of lines, I don't understand why you need to be able to handle so many units.

Bramante: Well, Peavey and Biamp are not just a couple of lines. They are the best lines to have as far as I'm concerned. Peavey is so diversified that you almost don't need to carry anything else. And I feel that Biamp is "the" up-and-coming line. We also have some MXR and Intersound units that are rack mountable.

Do you plan on taking on any more pro audio lines?

Bramante: Only if they fall into our dollar for dollar philosophy.

Are you into heavy discounting?

Bramante: No. We give reasonable discounts, but we don't give stuff away. It's possible to find someone who will try to undercut our prices just for the sake of saying they beat us out of a deal; but when that customer needs service he'll bring his amp in and that store may tell him they don't have a service department, or their part-time repairman can't get it done in less than a week or two. A little something will click in that customer's head and he'll realize we were the better deal. We also don't offer different cash and trade prices. And we don't bicker about prices. We've been around, we know instruments, and we set fair prices. And those prices are usually the lowest you can find. We take good care of our customers, in price and service; we make our deal the best, because we want our customers to stick around.

Anything else particularly unique about the store?

Bramante: Sure! We have a product which is absolutely one of a kind. Well, actually three of a kind—we have one in each store. I designed it and Mark put it together. We're in the process of patenting it now. It's called the "Pedal Pusher." It's a special display which contains all of the models of MXR and Electro-Harmonix devices we carry. This unit has the capability of displaying 54 effects devices; we can sequence up to six in any order. You can also hear any devices that are in stereo because the unit is hooked up to two amps located on the unit itself. The unit has a local/remote switch. On local a customer can try out any of the effects using a guitar or mic plugged into the control panel. On remote the customer can play any of the keyboards in our keyboard department in

conjunction with the effects on the "Pedal Pusher." Besides the output from the keyboards to the "Pedal Pusher," the keyboards are also connected to one of two keyboard mixers and then fed into an amplifier. There is also an output to a Leslie speaker if the customer wants to hear a particular instrument through it.

Sounds pretty easy on the customer.

Bramante: It is. And it's a lot of fun for him too. He can easily try out anything he wants. We've created a juxtaposition of opposites. We want our customers to feel comfortable and unhurried as they try their instruments and amps, but we also want to be as automated as possible to make their experimenting as easy as possible to set up.

What is "Daddy's Junky Mail?"

Bramante: "Daddy's Junky Mail" is a newsprint tabloid that we publish quarterly. It averages 28-36 pages and our circulation is roughly 10,000, mostly Daddy's customers. We do articles on the various lines we carry, features on what's new on what's new with the stores.

Before we go on to something else, why do you call your new building the Music Mall?

Bramante: The building is a three-story wooden structure. The main floor houses Daddy's Junky Music Stores. The lower floor has our three shops, corporate offices, the Music Education Division, and storage. The top floor is being finished to house a non-competitive music establishment such as a record store or audio-component store, or possibly a recording studio. As well as our music school.

Tell me more about the Music Education Division.

Bramante: We've just set that up this year to handle band instrument and guitar sales to schools. Besides conventional retail services, we've designed a multi-faceted program to expand the service we can offer schools. This includes field trips, assemblies, workshops, unit plans, etc.

What about your music school?

Bramante: We already offer music lessons in our Portsmouth store, but before the completion of the Music Mall we had no space to do it in our other locations. Our school won't be your run-of-the-mill lesson program. We've put together some really innovative plans so that we can offer a total experience to our pupils that will extend far beyond his half-hour or hour long lesson. We hope to have the program in full swing by next fall.

You've been in the retail end of the music business for about eight years now, in some way or another, and you've come pretty far. Where do you go from here?

Bramante: Well, we've come a long way, but then you really can't start much smaller than I did. And we have a long way to go yet. I have my eye on a few towns in New Hampshire where I may want to open another store. It would be super to have a Music Mall in each of the towns where we have a Daddy's. I'd like to try manufacturing the "Pedal Pusher" sometime. I'd like to get into recording [illegible]uipment in the near future. T[illegible]ities are endless.

INDUSTRY UPDATE

Schroeder Sales of Cleveland has been appointed a sales representative organization for the 3M Company's "Scotch" recording tape line.

The Norwegian government has approved the necessary funding of $24 million in capitalization and long term guarantees for the continued manufacture and general operation of the profitable divisions of Tandbergs Radiofabrikk, A/S. This will specifically include the Tandberg line of high fidelity products. Tandberg of America, Inc. is conducting regional high fidelity dealer meetings.

Charles S. Grill has been promoted to the newly created position of General Manager-Marketing Communications of the Consumer Electronics Division of Sharp Electronics Corporation. Grill has been with Sharp since 1974.

Lou Melillo has joined Ferrofluidics Corporation as Audio Products Manager responsible for applications engineering and marketing the use of ferrofluids in loudspeakers.

Osawa & Co. (USA), Inc. has appointed **Howard Milstein** Marketing Assistant to Marketing Director **Edward M. Healy**.

Koss Corporation has announced that Sansui Electric Co., Ltd. will be the exclusive distributor of its stereophones in Japan. Sansui has over 2,000 retail outlets in Japan.

Rank Hi Fi, Inc. has appointed three new sales representatives: Simonite Sales, Inc.; Sinai-Johnson; and Lloyd Doctoroff & Associates.

Michael Karmazin has been appointed Products Manager-Audio of Sharp Electronics Corporation, reporting to **R[illegible]ert Miller**, the Audio Departr[illegible]ting Service Manager.

Ron Means has been appointed National Sales Manager/Professional Products at Altec Lansing. Means was previously Sales Manager for University Sound.

Otari Corporation has appointed **Steve Krampf** National Sales Manager. Krampf had been vice president at Express Sound, and had served as representative for Tascam and dbx.

Soundcraftsmen has appointed Pacific South Coast Marketing to handle sales in southern California. Paul Hayden Associates has been named representative for the southeastern United States.

Robert E. Morrill has been elected Vice President, Marketing of Phase Linear. **A.P. Van Meter** has been named Vice President, Engineering and Research. **Walter J. Mildridge, Jr.** has been named Vice President, Finance. **Patricia Baker** has been named Credit Manager for the company.

Sansui Electronics Corporation has appointed **T. Yoda** Vice President/Sales and Marketing, replacing **Ken Hoshino**, who has been named Executive Vice President in charge of Sansui's European operations. Yoda was previously western regional manager and general manager in Sansui's Los Angeles office.

Thomas Ishimoto has been appointed to the new position of National Product Manager for Nikko Audio. He was previously west coast regional manager for Onkyo.

TDK has expanded its national operations department. **John Greges** has been named Eastern Operations Manager. **Eugene Dunham** has been named Western Operations Manager. In addition, **Jerry Rader** has been promoted to National Credit Manager.

Ben Loughrin has been named Market Development Manager of Phase Linear. He was previously with JVC America. **Bruce H. Lowry** has been named National Sales Manager for the company. He was previously a Sales and Marketing Manager at 3M.

John Dale has been promoted to Vice President and General Manager of the Magnetic Tape Division of Fuji Photo Film U.S.A. Dale had been General Manager of the Division. **Fred Nakamura**, Executive Vice President and Chief Executive Officer has been elected to the Board of Directors of the parent company.

International Audio Marketing of Westwood, Massachusetts has been named the international marketing representative for Integrated Sound Systems, Inc. A subsidiary of the VSC Corporation, Integrated Sound Systems manufactures and distributes GLI sound equipment.

Maxwell Corporation of America has appointed **Steve Levine** Midwest Regional Sales Manager for consumer audio and video products. Levine was previously with Third Century Marketing.

The Professional Audio Products Division of Sony Industries has completed its national sales representative network, with the recent appointment of 2001 Enterprises; Kramerson-Randall Sales Corporation; Norpac Marketing, Inc.; and Furman-Goldman & Associates.

Sony Corporation and Superscope have jointly agreed to terminate their contract early. Sony has also agreed to purchase back all outstanding microphone and mixer inventories from Superscope. In addition, Sony has announced plans for the construction of two microphone testing and repair facilities to be located in New York and California.

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