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VOLUME 7 – NUMBER 4

Recording engineer producer

INCLUDING: CONCERT SOUND REINFORCEMENT

THE SYNTHESIZER AS A SIGNAL PROCESSOR . . . page 27

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R-a/p 2

Circle No. 101

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Dual capstans	Yes	No	No	No
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BOUSE APD 1600 with AUTO-REPLAY, digital counter, and complete remote control: $18,750.

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*Manufacturers published features and prices at time of publication.

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'RECORDING engineer/producer' is published bi-monthly (six times a year) by RECORDING & BROADCASTING PUBLICATIONS, 1850 No. Whitley Avenue, Suite 220, Hollywood, CA 90028, and is sent to qualified recipients in the United States. Subscriptions for other than qualified individuals or companies may be purchased at $7.50 per year (6 issues). All foreign subscriptions Surface Mail — $8.50, Air Mail — $15.00.

Controlled Circulation postage paid at Los Angeles, California.

Address all correspondence to:
RECORDING engineer/producer
P.O. Box 2449
Hollywood, CA 90028
(213) 467-1111

AUGUST 1976
VOLUME 7 – NUMBER 4

The Cover: Symbolic of both the sound reinforcement and synthesizer articles in this issue is Gary Davis' photo of the Neil Diamond troupe keyboard and effects kit awaiting set-up for the night's concert in Tempe, Arizona

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LEADER IN ADVANCED TECHNOLOGY

LETTERS and LATE NEWS

from: MARTY W. EGGERSS
3—M Company
St. Paul, MN

Re: Understanding Magnetic Tape Performance Specifications by Peter Butt. R-e/p, June 1976.

First of all let me express my regards for having the fortitude to delve into a subject so often misunderstood (and usually ignored) as that of magnetic tape specifications. I think that nearly everyone involved with magnetic recording should come away at least a bit enlightened after reading the article.

For the sake of clarity (or controversy) you and others may be interested in my experiences concerning some of the subjects covered.

You mention (page 32) that the figure given for modulation noise and print-through will change, depending on the reference level used. I have found that both of these parameters are independent of any reference level. That is, the signal-to-print or modulation noise ratios (in db) will always be the same no matter what reference level signal is recorded.

I hope by now that it is commonly known that, as you point out, the record head geometry (particularly gap length) has a major impact on tape performance. Probably the most apparent differences do occur in the shape of the bias (sensitivity) curves and the long wavelength maximum output. I will suggest that the long wavelength (low frequency) output for a given distortion level will tend to be *higher* for a narrow record gap, i.e. more "headroom", and the sensitivity peak will be *broader* with a narrower gap. These two facts are contrary to what you have expressed. I believe the two Ampex data sheets you reference will bear out the latter. Again, I will not attempt to explain this based on the physics of the recording process — only on my findings after running many rolls of tape on mastering machines with different record gaps. This would suggest that there is little sacrifice in recording performance with narrow record gaps, while still maintaining a good sync response.

You also mentioned that print-through will be less on a machine with a narrow record gap. I have not found this to be the case. The two Ampex data sheets you refer to have different tests for obtaining the signal-to-print ratio. One has a 150°F test, the other is a 70°F test. This, rather than the record gap, would tend to explain the difference in print-through shown on these data sheets. As you point out, higher temperature gives higher print. This is only to encourage one not to rush out and buy a new machine or head stack with a narrow record gap, just to eliminate print-through.

I hope some of our discussions will provoke further thoughts and investigation into variables affecting magnetic tape performance.

Reply to Marty Eggerss: from Peter Butt:

Re-examination of the points mentioned leads me to concede that Marty Eggerss is correct. The matters of print and modulation noise ratios I would attribute to oversights of proof reading that should not be excused. The statements I made regarding the increase in medium wavelength recorded flux with increased record gap length were based upon experience with gaps in the 50 to 200 microinch range. In those cases, the statements made are true. I was previously not particularly aware that the effect apears to peak somewhere around th 250 micro-inch region, declining as the gap dimension is increased beyond that. I wish to thank Mr. Eggerss for taking the trouble to bring these points to my attention. This illustration of the jeopardies of extrapolation will not be lost on this writer. Marty's response evokes mixed emotions on my part: embarassment at having erred and pleasure at having stimulated informed commentary.

from: BRAD S. MILLER
Sutton/Miller Ltd.
Los Angeles, CA

While Mr. Barry Schlosser's June 1976 R-e/p article, "FyF System — A quad Alternative" was certainly valid in two respects; (1) as a wonderful promotion for FyF Studios Inc., and (2) as an alternative for eccentric audiophiles who like to move furniture around (no wives to complain is an absolute prerequisite), his evidence to support his theory contained in a paragraph headed "Marketing Considerations" simply does not hold water.

While the music industry (software and hardware combined) as a whole has performed rather miserably with quadraphonics, it is not the fault of the music industry entirely, nor can the blame be laid at the doorstep of the basic 4-channel concept as we have come to know it and accept it, either.

A parallel. The first commercial stereo disc was made in 1956. The music industry (all inclusive) became an all stereo industry in 1964, eight years later. Now what do you suppose took place during those long eight years? We hasseled over the lady of the house — was she going to allow another speaker in her living room, let alone where was it going to be placed? We fretted over the fact that we had been listening to music through one speaker for twenty years monaurally, so wasn't that good enough? We argued over systems, yes systems; for I am sure that most of us remember the fake stereo records that were finally required to carry proper labels such as *re-channeled* or *electronically processed to simulate stereo.* And the clincher? STEREO FM RADIO, September 1961, which allowed the public in general to accept or reject this technological advance, this radical departure from the norm, and thusly created the all stereo music industry that we have enjoyed since 1964 until the present.

Now, regardless of these eccentric points of view, doomsayers I say, we are in fact nearing the point in time when discrete 4-channel broadcasting will be a reality. In order to provide the FCC with the data necessary, the National Quadraphonic Radio Committee, of which I was a member, conducted weeks of subjective quadraphonic listening tests, in San Francisco and Syracuse, New York. The general public was invited to participate in these tests as official "program auditors" complete with hearing tests. The primary purpose of these subjective tests was to compare listener preference for discrete vs. the various matrix systems proposed. The conclusions reached by the NQRC Panel 6 found absolutely no evidence to support Mr. Schlosser's unfounded claims that sounds coming from behind a person were either disturbing or upsetting.

Environmentally speaking, with ten years of 4-channel recording and production experience including The Mystic Moods (my turn for a plug), a properly recorded sound or music environment has no front or rear to begin with. The listener should be free to roam freely within the speaker area, in any direction, and obtain a sensation equal to that of wearing head phones at the very least.

It is a proven fact that humans hear sound from all directions and positions. So let's not blame the concept of quadraphonics as we know it today, 4-speakers, one in each corner of a listening area for lousy circumstances, poor economical conditions, inept marketing skills better described as a genuine lack of interest, until FM QUAD RADIO has an opportunity to demonstrate fully the potential of 4-channel sound to the American public!

Once the market demand is created, you will find the interest of the music industry, both record makers and equipment makers, to be focused clearly on quad. Right now, the CB'ers have their attention. Multi-media, multi-channel home entertainment centers are the wave of the future. Stereo AM radio will force

FM stereo radio to maintain a competitive edge. That edge will be 4-channel.

I am hopeful that your readers were not bored with this speech; I haven't made one in so long that I quit only when the paper ran out.

Ed: . . . And an equally wonderful promotion for discrete quad!

Ed: Through an oversite a notice of copyright was inadvertently omitted from the article by Mr. Schlosser. Such notice appears here for the record:
THE 'FyF' SYSTEM © 1976

. . . and more on the discrete quad scene —

from: JAMES GABBERT
K-101
San Francisco, CA

On July 24, 1976 K-101 was the flagship station for a discrete quadraphonic stereo network broadcast. This unique program was aired througout most of Central and Northern California and involved the cobined efforts of K-101 and KBRG in San Francisco, and KZAP and KSFM in Sacramento.

K-101 broadcast two of the four channels over its stereo signal on 101.3 mc and the remaining two channels over KBRG's signal on 105.3 mc. The show will be rebroadcast in Sacramento over KZAP (98.5 mc) and KSFM (102,5 mc). It will take two stereo receivers to listen to this experimental broadcast. The four stations promoted [the show] to their combined listeners to have "Quadraphonic Parties", invite a friend with another stereo system.

. . . At present the Federal Communications Commission does not allow a single station to broadcast discrete four channel stereo by itself, however a recommendation placed before the Commission by K-101 is currently being considered and a final decision is expected shortly.

In the final analysis, this first attempt at a network broadcast of discrete four channel stereo is both innovative and exciting and we felt that you [your readers] would be interested in knowing of it.

DICTAPHONE RENAMES SCULLY/ METROTECH AUDIO/ELECTRONICS DIVISION

MOUNTAIN VIEW, CA. . . . Dictaphone Corporation's Scully/Metrotech Division here has been renamed the Audio/Electronics Division.

E. Lawrence Tabat, President and Chief Executive Officer, said the new designation more fully defines the scope of the division's manufacturing and research operations.

The company will continue to use the Scully trademark on its line of broadcast and studio recording instruments and the Metrotech trademark on its broadcast recorders and reproducers.

Think of them as your musical instruments.

The audience can't see you. But they can sure hear you.

They don't know it, but they're depending on just one person to get the music to them. And that guy is you.

It's not something an amateur can do. It's an art. And that's why Yamaha has designed 3 superb mixing consoles with the qualities and range of controls that the professional sound reinforcement artist needs.

For instance, our exclusive 4x4 matrix with level controls gives you more exacting mastery over your sound than the conventional method of driving speaker amps directly from the bus outputs.

Features like that are years away except on the most expensive mixers. On the Yamahas, it's standard equipment. And so are transformer isolated inputs and outputs, dual echo send busses, an input level attenuator that takes +4 dB line level to −60 dB mike level in 11 steps, and 5-frequency equalization.

Whether you choose the PM-1000-16, the PM-1000-24 or the PM-1000-32, Yamaha gives you the flexibility you need to turn your job into an art. And because they're designed from the ground up to perform on the road, more and more professional sound men around the United States and the world are depending on Yamaha, night after night, gig after gig.

If you've never thought of your mixing console as a musical instrument, we'd like to invite you to stop by your Yamaha dealer. Once you've checked out the operation manual and tested for yourself what the PM Series can do, we think you'll come away a believer.

YAMAHA

Box 6600, Buena Park, CA 90620

Circle No. 109

by chris foreman

The typical sound reinforcement system is a maze of dozens of different kinds of components, connected with hundreds of feet of audio cable. It may be spread out over a stadium-sized area. With this kind of complexity, the potential for trouble is high. No matter how carefully the system has been engineered, ruggedized, its connectors optimized, the problems will occur . . . even after seemingly endless hours of burn-in time in the in the shop.

Quite unavoidably, the sound system traditionally shoulders the burden for maintaining the mood of the concert. When the problems do happen, it is usually at the most critical moment. Buzzes, distortions, and noises seem to wait until the emotional part of a soft number where the artist has the audience completely enraptured

troubleshooting the concert reinforcement system

. . . then the glitches pounce! To Avoid becoming the center of unwanted attraction, and to avoid the wrath of the artist, the producer, the road manager, and even the audience, the concert sound man had better be able to fix any problem . . . and, fast. That means finding the offending factor . . . isolating it . . . fixing it . . . or working around it. And, that is what troubleshooting is all about.

GETTING STARTED

I was initiated into the friendly troubleshooting fraternity on my first concert tour; in the South-West . . . Texas and New Mexico . . . 21 shows in 13 days . . . with the SUPREMES. The SUPREMES were sensational . . . the sound system, well, lets just say we had a few challenges. Anyway, by the third show I was deemed capable of setting up the system by myself with the . . . aid of a fellow truck driver and a couple of stage hands. We got it set up, patched it together, and turned it on. The distortion level was about 105%. I was lucky enough to have several hours before show time, a good thing, because it seemed like eons before I found the one bad cable that was causing the distortion.

On the last show of the tour, we weren't so lucky. During the last number (you guessed it - - a soft, smooth, romantic piece), as we were patting each other on the back for a successful show, the system began an imitation of machine-gun fire. Beads of sweat popped out on my forehead, and I froze. My boss, who had obviously been through this sort of thing before, kept his cool and began to go through an amazing set of seemingly random contortions, ripping out patch cables, pulling down sliders, till in about 10 seconds, he had isolated the problem. He was the hero; I was the one who got to go up on stage and replace the bad microphone . . . in my 13 day unwashed jeans.

It's from this kind of experience, out in the field, that the concert soundman really learns to troubleshoot. In my case, I'm sure that the learning process would have been easier if I had known a few things about system design, and at least the rudiments of troubleshooting logic.

. . . the Author:

CHRIS FOREMAN is an electro-acoustical consulting engineer, associated with Gary Davis and Associates of Reseda, California. He earned a B.S.E.E degree from the University of Nebraska at Lincoln.

The material in this article came from his experience as the co-owner of an electronic repair business in Lincoln, and his three and a half years as the senior engineer at Stanal Sound, a concert sound company in Kearny, Nebraska.

THE SYSTEM

Its Components

Before you begin to troubleshoot, its very advisable to have an understanding of the components of the system. It isn't necessary, at first, to understand the circuitry, but there is a need to understand function, input and output levels, impedances, and so forth. It may sound elementary, but you have to know what a component is *supposed* to do before you can determine whether or not it is malfunctioning. Good sources of information about components include owner's or service manuals, text books on specific types of components, and the articles that appear in this, and other technical journals.

Its Block Diagram

It is also important to understand the block diagram of the system, which takes you from an understanding of the components to an understanding of the system.

Understanding a block diagram is really pretty easy, once you get into the symbolism and conventions. Figure 1 is a chart of common, block-diagram symbols. Figure 2 is a block diagram for a basic sound reinforcement system.

Block diagrams are also known as "one-line", or "single-line" diagrams; the purpose of a block diagram is to show signal-flow between the components of the system. Omitting return wires, balanced lines, grounding, and AC power wiring, makes it clear how the components fit together to form a system. A block diagram could be compared to a city map of major streets and freeways; both show an overall view of a complete system. *Schematics* are like detailed street maps of individual sections of the city, they may help when you get close to where you're going, but they just get in the way when you have to get from major point to major point in a hurry. To follow a block diagram, begin at the left side. All signal paths are "one way" streets; no matter where a signal enters the system, it can only move toward the right (the few exceptions are

continued . . .

Circle No. 110

explained a couple of paragraphs later). Low-level signals (microphones or other sources) start at the far left side of the diagram and are amplified and processed by the various devices in the system as they move toward the right.

It might be advantageous to study the block diagram in Figure 2, and imagine what would happen if the signal flow were interrupted at any point along the line. Would the entire system be affected, or just part of it? Imagine that a noise source is placed at some point along the signal path. What effect would it have on system operation, and at what point could the signal flow be broken to stop the noise? (Don't worry about cutting out program material at this point.)

It's common practice in systems that are more complex than the one in Figure 2, to let some signals flow up or down, or even from right to left. Usually, however, the *general* flow is still from left to right, and a signal always flows *away* from its source. To study a good example of a complex system, with a complete block diagram, read "the Universal Ampitheatre Sound System" by Gary Davis, which appeared in the December, 1975 issue of "Recording Engineer/Producer"

BRASS TACKS

The first step in troubleshooting, quite obviously, is defining the problem. The following problems are classified by *symptom*, not cause.

Type-A Problems create *new*, unwanted sounds like hum or noise.

Type-B Problems create distortion, or they interrupt existing sounds (program).

TROUBLESHOOTING TYPE "A" PROBLEMS

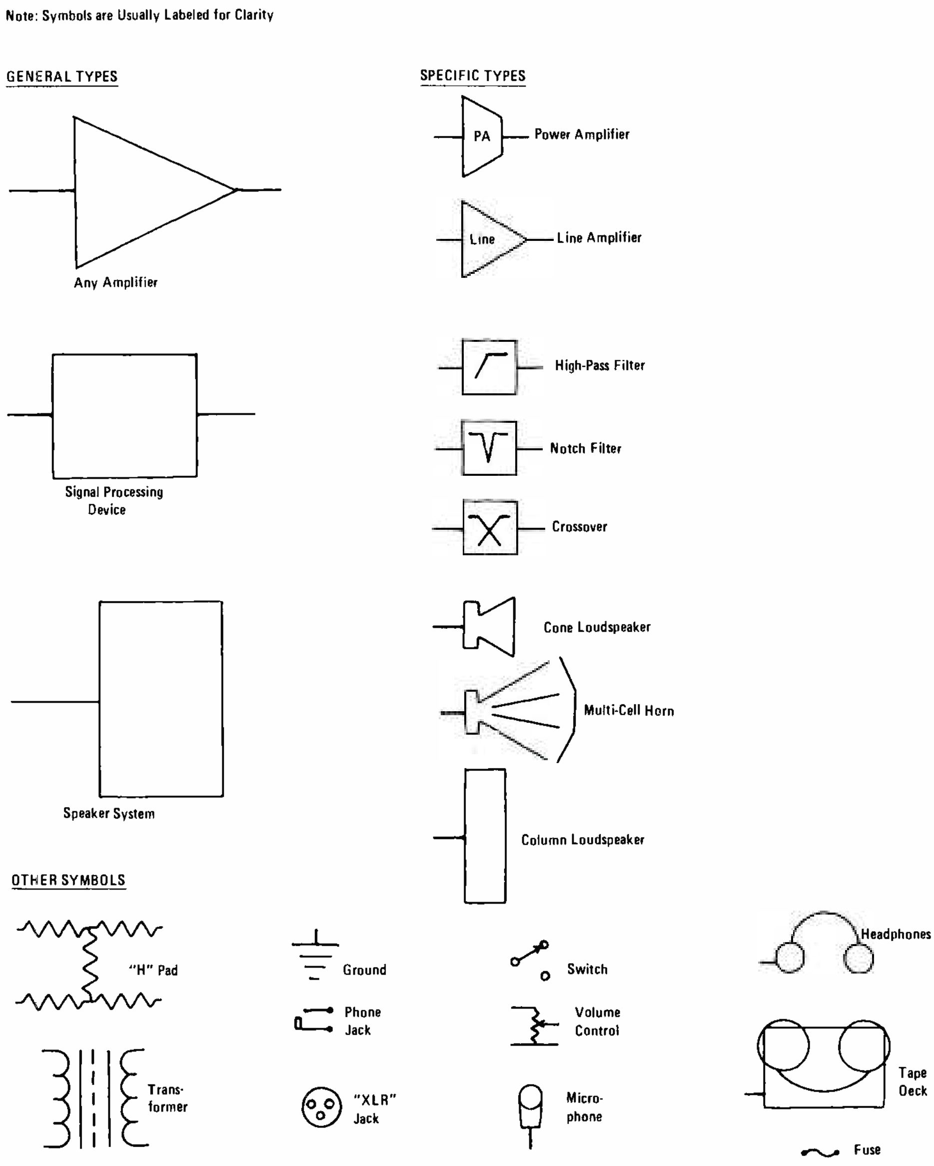
Many type-A problems are easy to locate. Using the block diagram as a guide, break the system (unpatch a cable, pull down a slider) at some convenient point, about half way through the system. If the symptom disappears, the problem is in the left half of the system, if it stays, the problem is in the right half of the system. ("Left" and "right" refer to component positions in a block diagram as discussed earlier.) Continuing this technique, further subdivide the system until the specific component or cable causing the problem, is located. Replace, or patch around it as necessary.

Figure 3 is a flow-chart describing the process of troubleshooting type-A problems. A flow-chart is similar to a block diagram. Start at the top, and follow the arrows.

Suppose the misbehaving sound system resembles the system of Figure 2, and it develops a problem like *hum* (dotted source). Using the flow chart in Figure 3, it might be an excellent exercise to attempt to troubleshoot the problem. We have found it an excellent training procedure to set up a sound system in the shop and have someone create type-A problems for some hands-on practice in troubleshooting.

However, there are a couple of faults with the whole approach to troubleshooting type-A problems. First, unpatching the system (or pulling down some appropriate slider) at a critical point immediately shuts off the entire program. Second, some systems will hum loudly or even go into oscillation when they are unpatched randomly; this can mask the original problem. Pulling down individual input faders (or listening on a "cue" bus if the console has one), may be a better first step.

FIGURE 1 COMMON BLOCK DIAGRAM SYMBOLS (From Template Set)



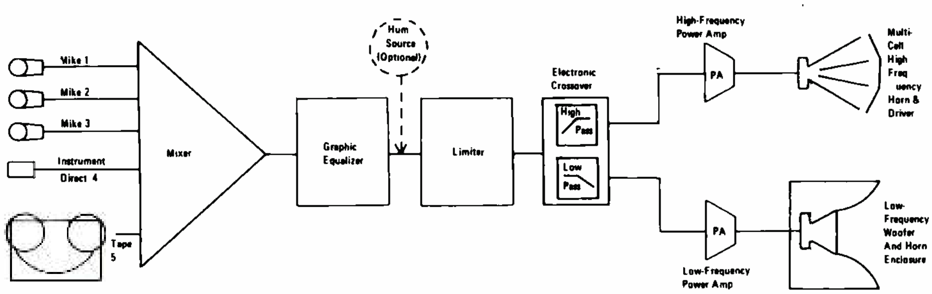
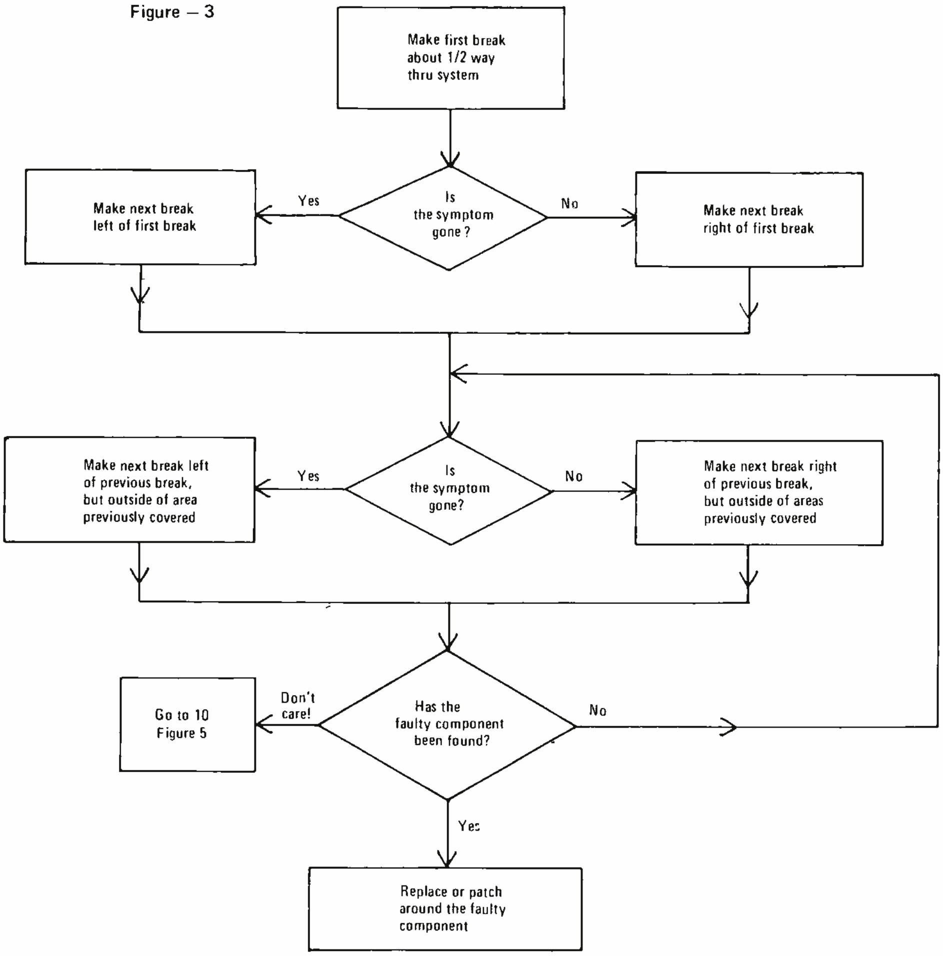


FIGURE – 2: BLOCK DIAGRAM OF A SIMPLE SOUND REINFORCEMENT SYSTEM

Figure — 3



Unpatching is a good way of isolating type-A problems, but before anyone begins ripping out cables, it would be better to be sure that the actions won't be causing additional system problems.

Another fault in this approach is that some problems, notably electronically-caused oscillations, and ground-loop hum, are of a nature such that unpatching the system at just about *any* point will eliminate the symptoms (and the program!). No amount of component or cable replacing will solve this type of problem. There are a few tricks, that can help, and these are covered in subsequent paragraphs.

TROUBLESHOOTING TYPE "B" PROBLEMS

To troubleshoot a type-B problem, a signal source and a signal tracer are needed. Figure 4 lists conventional signal sources, and tracers, as well as some unconventional ones. Conventional test equipment is often expensive and seldom roadable without a specially-built case. There are exceptions, but generally for the quick yet noncritical testing you can do during a performance, one of the unconventional sources or tracers will do just fine. Additionally, the signal source and tracer should be battery-powered (if they require power) to avoid compounding ground-loop problems, and to make it easier to move them from point to point around a large hall.

Adapters and Patch Cables

Since tracers, and signals sources aren't much use if they can't be plugged in where you need them, this is a good point to discuss the importance of a good set of adapters and patch cables. Since each sound system is different, specialty adapters will vary, but the soundman can rely on the need for plenty of standard types with XLR's, phones, RCA's, etc., in all combinations. With this selection, when a problem component is located that needs different input and output connectors, even input or output pads or transformers, you're prepared. If the faulty component can't be replaced, you can patch around it, provided the necessary adapters are available. A good adapter kit can help too, when there is a need to borrow a replacement device from the house, or some local band, or sound contractor; it will almost invariably have different connectors and input and output parameters from your original. Similar concepts apply to building and maintaining a tool kit.

Now, when a type-B problem appears, select an appropriate source and tracer as suggested by Figure 4, and follow the steps outlined in the flow-chart of Figure 5.

In many cases, the easiest signal source to use will be a microphone or one of the mike-level tone generators, like the Shure A15TG, with a battery-powered preamplifier; a combination that can be connected at various points along the signal path. The most convenient tracer is sometimes already part of the sound system. If the speakers and power ampli-

Figure — 4

SIGNAL SOURCES

CONVENTIONAL:

Sine/square-wave generators, function generators, noise of tone burst generators.

UNCONVENTIONAL:

Tape deck, turntable, microphone, (Say: "Check, one, two."), transistor radio with monitor output, transistor radio taped to a mike, electric guitar, other electric instrument, noisy preamp, your finger, (touch hot end of low-level signal cable to create hum).

SIGNAL TRACERS

CONVENTIONAL:

Oscilloscope, conventional signal tracer, VTVM, etc.

UNCONVENTIONAL:

High-impedance headphones, small speaker with a 70 volt transformer, tape deck with source monitoring, any pre-amp, etc. with a VU meter, the power amp and speaker section of your system, an entire duplicate system, or other system, guitar or instrument amplifier, transistor radio with auxiliary input (or wire an input to the volume control).

Note: A telephone pickup coil plugged into an appropriate signal tracer can locate magnetic fields (a common source of hum). A transistor radio tuned between stations can locate the source of RF fields (which can cause hum or buzzes, as well as radio station interference).

After a flurry of bold claims and brash statements from several manufacturers, regarding their purported automation systems, the Turtle has a question: Where are they?

One ad, for instance, shows photographs of a fully automated console and suggests that it is run by the advertiser's own automation programmer. WRONG! The console in question is powered by the Turtle's first generation system.

Another, claims to have been first to create automated mixdown, a claim of dubious importance in the first place. Actually, a company called Olive deserves credit for being first. The Turtle, of course, has the honor of the "first successful system".

But what about the Turtle's second generation system? Several months ago, the Turtle astounded the business world by actually turning down orders for its successful first generation system. It did this because it had proven to itself that a far superior system was in its laboratories. Being a Turtle, it took its time in packaging this new system, while orders for same filled its desks. (Of course, it didn't worry about someone else grabbing the orders... there was no one else.)

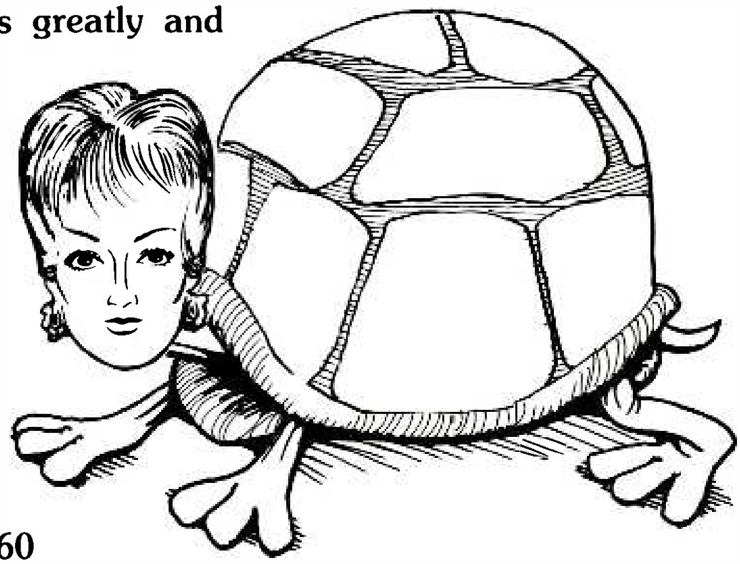
Finally, on July 4, the world had its first chance to view the Turtle's new triumph. Some 200 Turtle controlled lights and 10 channels of Turtle powered audio were trained on the historic Old State Capitol in Springfield, Illinois. It was a tribute to Abraham Lincoln, the United States and the Turtle. Every night since, some 500 people have watched and listened as the Turtle flawlessly faded the faders, dimmed the dimmers and switched the switches.

A second triumph came on August 12, when Quadrafonic Studio in Nashville, Tennessee, turned on 32 of the world's first centrally controlled programmable equalizers. (Of Turtle origin, of course.)

On both occasions, the Turtle's staff was there, making notes and corrections, so the subsequent production equipment might offer the best possible performance.

To those in the industry who have patiently awaited the Turtle powered systems, which they ordered, we apologize for the time we have taken in packaging (you know how turtles are).

In consolation, though, we do offer evidence that our second generation system does, indeed, do what we claimed it would. Our Fabulous Fader does fade fabulously, our Great Equalizer equalizes greatly and our 65K programmer does program 65,000 bits rapidly and reliably.



allison research, inc.

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fiers are working, the entire sequence of Figure 5 can be performed using the speakers and power amps as a signal tracer. (If the speakers or power amps aren't working, you'll find out by using the same process.)

The troubleshooting method outlined in Figure 5, has a number of variations. For example, the source could be connected at the far left of the system, and the tracer could be moved from the far right toward the left. Alternately, both the tracer and the source could start near the center of the system and move outward. Obviously, if any particular part of the system is suspect start at or near that point.

DEALING WITH INTERMITTANT PROBLEMS

It can be very difficult to fix something that refuses to malfunction. Fortunately, on the road, the worst intermittant problems are usually physical failures, like broken cables, loose connectors, or parts that are falling off of circuit boards inside of a component. For this type of problem, the symptoms can be recreated by shaking cables, or pounding on components with your fist. A rubber hammer is a valuable tool for finding intermittants on the road.

The more subtle intermittants, low-level (through high irritation) buzzes and noises, are the worst. A lab technician in the shop might use a can of "freeze mist" or a heat lamp to find these problems, but in the field, you've got to locate the individual component first. Quick thinking and careful observation are about the only hope. Try to remember any other conditions present at the time of the intermittant, and correlate them with the problem. If for example, console lights dim at the same time an abrubt pop goes through the system, the problem may be in the mixing console, or its AC power supply. If a particular hum or buzz corresponds with the producer's call for a particular lighting set, the problem may very likely be a grounding/shielding problem, and possibly a noisy AC line.

While it should be all too obvious, instead it's all too easy to overlook checking fuses and pilot lights at the first sign of any problem, especially those that aren't intermittant. But check them even for intermittants, since a loose fuse cap can cause problems, and blinking pilot lights can display them.

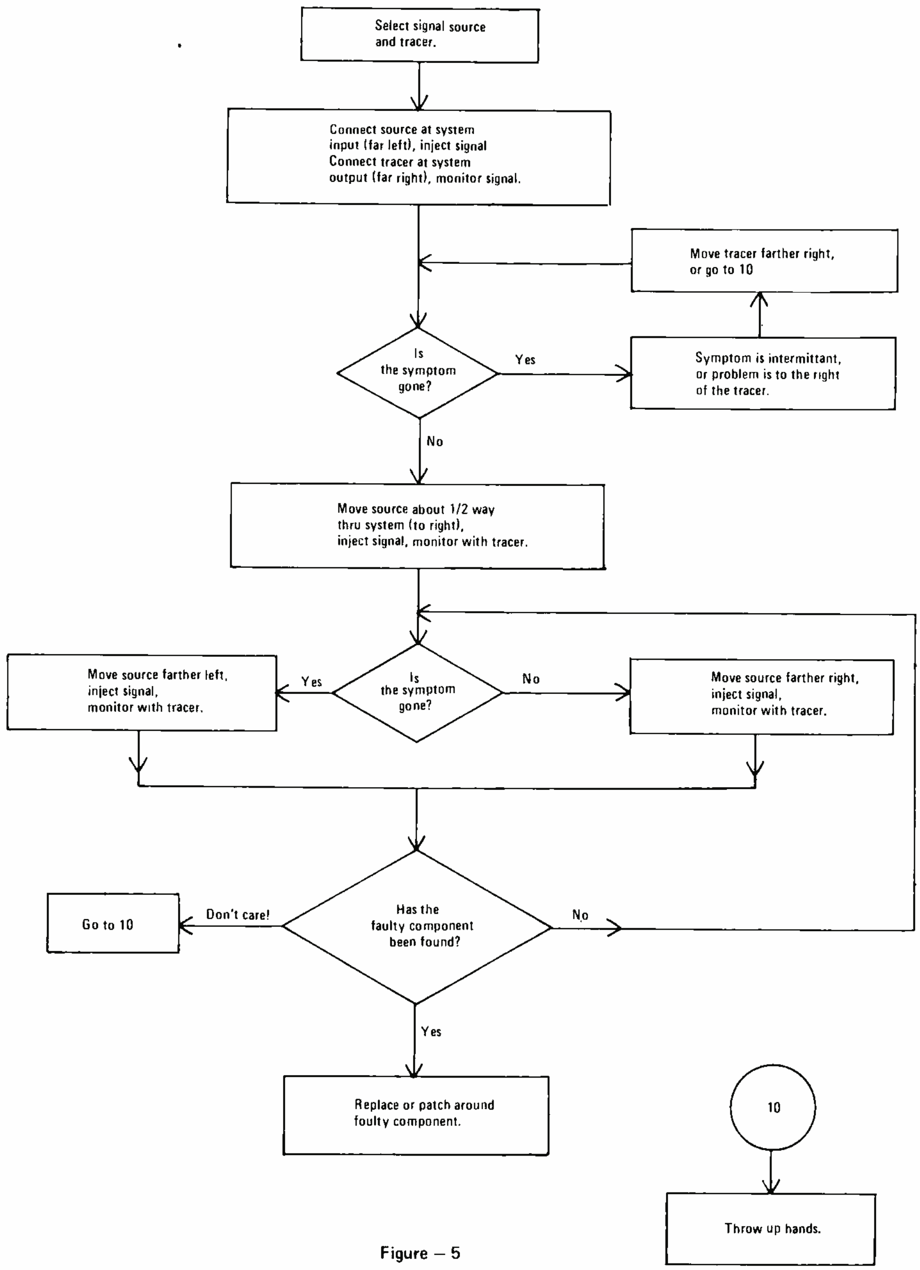


Figure – 5

PREVENTION

Now that we have all these troubleshooting skills under our belts, let's talk about a better method of dealing with problems, their prevention.

Even if you're a system operator, and you aren't involved in system design, or its construction, you should still be involved in problem prevention because problem solving, during a performance, is your task alone. One way to help prevent problems on the road is to help keep the people back home, in the shop, informed. Person to person feedback is invaluable to the people on the test bench. They may not get out in the field often enough to really have a "feel" for the problems on the road, yet their input to your system can make the difference between a successful performance and a real blow-out.

Careful system design, though, is still your best hope for problem prevention. System grounding, impedance-and level-matching, and cabling are at least as important as selecting the right mixing console or high-frequency driver. So, if you have the slightest inclination, learn as much as possible about system design. I recommend Don Davis' "Syn-Aud-Con" seminars (P.O. Box 1134, Tustin, CA. 92680), and his new book *Sound System Engineering*, by Don and Carolyn Davis, published by Howard W. Sams Co. The *Audio Cyclopedia* by Howard Tremaine, also published by Sams, is another good source of information.

One often overlooked design parameter is "neatness". A rat's nest of wiring around the console, or in the power amp rack not only asks for failures, it really complicates troubleshooting. Every sound company needs one of those characters who is a fanatic for neatness, and will sit in a corner of the shop all day, doing nothing but wiring equipment. Problem prevention is his game. You'll recognize the type immediately when you look at the back of a completed rack and find harnesses with cable ties every half inch, and shrink tubing on the end of every shield wire.

On the other hand, maybe there are only two or three people in your whole company. Yet, even if you buy your systems and components from someone else, system design and construction are still important considerations. Specify the components carefully, and if you don't know and approve of the supplier's cabling technique, cable the system yourself.

Spare Parts and Modularity

Carrying critical spare components (such as electronic crossovers) helps prevent a major disaster if one component fails, and seems to foil Murphy's Law as well. If you carry a spare component, somehow, the original is less likely to fail.

"Modularity" helps too. As an example, if you carry a 16-channel console, and two, 8-channel sub-mixers, instead of one 32-channel console, then one failure won't stop the show.

PROTECTION

Every problem can't be prevented, but the following protection schemes can help prevent a possible complete disaster when simple faults occur.

Capacitors

It is relatively common practice to insert a non-polarized capacitor in series with a mid or high-frequency driver to protect them against low-frequency signals, and against DC current from a faulty DC-coupled power amplifier. To review protection capacitors are chosen by value, voltage rating, and type:

$$\text{Value (in microfarads)} = \frac{500{,}000}{\pi f Z}$$

where "f" is the crossover frequency divided by two, and "Z" is the impedance of the driver.

Voltage rating must be greater than the *total* peak-to-peak voltage output capability of the power amplifier.

Type: Capacitor must be *non-polarized*, and AC rated for continuous duty. Non-polarized electrolytic capacitors are a good choice.

Fuses

Fuses protect electronic equipment against AC over-voltage, and internal faults. Placed in series with a loudspeaker, they can help prevent damage to the loudspeaker caused by excessive average power or DC current from a faulty DC-coupled amplifier. Choose a speaker protection fuse by current rating and type. Current rating: (the following formula is approximate:) current rating in amperes =

$$\sqrt{\frac{P}{Z}}$$

where "P" is twice the rms or pink noise rating of the loudspeaker, and "Z" is the loudspeaker minimum impedance.

Dilemma

Murphy must have invented fuses, or at least been the first one to specify them for speaker protection. Fast-blow instrumentation fuses may blow on program peaks, disrupting the concert, but standard, or slow-blow type fuses, may not blow quickly enough to prevent loudspeaker damage. If you choose the current rating to protect the loudspeaker against any power greater than its rated power, you will probably protect it against working at all; yet at twice the power rating (as used in the above formula), the speaker *may* sustain damage before the fuse blows. It may be better to lose a couple of speaker cones occasionally than to have to replace fuses in the middle of a performance just because the kick-drum is a little overbearing. If it is decided to use fuses, fuse each loudspeaker separately. That way, it's less likely that several loudspeakers will all go out at once.

Relays

Relays have been a common feature on the output stages of consumer or Hi Fi type, power amplifiers for some time. Now at least one manufacturer is installing them on the output of a professional "*super-power*" amplifier. If the design is reliable and the time constants are right, relays can be the answer to the fuse dilemma. Relays with the proper sensing circuitry can protect loudspeakers against DC current, excessive average program power, and amplifier turn-on, turn-off transients.

Limiters

Limiters are not usually considered loudspeaker protection devices, but they can be a valuable addition to a concert sound reinforcement system. A "*squared-up*" or "*clipped*" waveform causes a loudspeaker cone or driver diaphragm to move to one position and stay there. Because there is still power flowing through the voice–coil, but there is no voice–coil movement, the power becomes heat. If the system is set up so that the power amplifiers are the components with the *least* headroom, then they will be the first components to square-up when a high-level transient comes through. A limiter, placed just before the power amplifiers, (or the electronic crossover) can be adjusted to prevent these peaks from clipping the power amplifiers, which may save a few voice coils.

Patching

Many electronic devices create severe turn-on and turn-off transients. A good way to prevent these transients from reaching the speaker system, is to turn on all of the electronics, including the power amplifiers *before* you make the final speaker system to power amplifier connections. Similarly, disconnect the speaker system from the power amplifiers *before* you turn off *any* of the electronics. If this is impractical, follow the same procedure with the feed cables to the power amplifiers.

Some systems however, cannot be unpatched at random when they are operating. As an example, most guitar amps will hum loudly if you unplug the guitar cable from the guitar, before you unplug it from the amplifier. Since each system will be different, be especially careful with unfamiliar systems.

COMMON TROUBLE SPOTS

Cables and Connectors

By far the majority of all system problems will be caused by bad cables and connectors. Even when heavy duty, rubberized cables, with professional connectors are used, and they are treated with care, the abuse of connecting and reconnecting, along with repeated coiling and yanking, will ultimately cause failures. Hums, noises, distortions, and drop-outs, can all be traced to bad cables.

Physical Problems

Cables aren't the only components that get abuse on the road. Amp racks get dropped, mikes and connectors get run over by fork–lifts, "*g–forces*", believe it or not, cause parts to actually fall off circuit boards. A close physical inspection of the system before the performance will spot a lot of these failures, the ones you can't see will usually show up when the system is first turned on.

Power Supplies

AC power is a hassle almost everywhere you go. One engineer I know did a whole Elvis Presley show on a single M68 mixer after he plugged in his main console to an innocent looking 110V AC outlet only to find out that someone had connected a 220V AC circuit to it. Don't let it happen to you! Carry an AC checker, and use it faithfully. In an attempt to solve AC problems, one sound company has put "AC crowbars" (an electronic circuit which senses an AC overvoltage condition and short circuits the AC line) at the AC input of all their equipment. If the AC voltage reaching the equipment exceeds 135V AC, the crowbar shorts the AC line, blowing the house fuse or circuit breaker. That's one possible solution. Another company carries their own AC distribution system. The distribution system connects directly to the house AC service entrance, avoiding all house wiring completely. If you can afford the money and setup time, this is an excellent way to go.

Batteries

Batteries, are part of the power–supply problem. That bad mike on my first tour (an electret condenser mike) turned out to be a bad battery. Obviously, use good quality batteries in everything that requires them, and avoid batteries whenever you can (using a phantom power supply for condenser mikes is much more reliable). On the other hand, sometimes severe AC noise, or grounding problems can be helped by battery powering the mixing console and any other low–level equipment. Most of the Shure 'M' series mixers, for example, and the Altee 1220A console can be battery powered, If your console can be battery–powered carry a set of batteries heavy enough to see you through a show. As mentioned before, signal sources and tracers should be battery–powered.

SOLUTIONS

Hum and Noise Problems

Hum can be caused by a faulty power supply in a component. You can locate the component by the troubleshooting method for type–A problems, and replace it. Similar solutions apply to hum caused by broken shields. Unfortunately, a lot of hum problems are caused by poor grounding design, and such problems aren't solved, *they're prevented.* If a different bank of light dimmers comes on in the middle of the show, and all of a sudden, your program to hum ratio is about 1 : 1, or if a local radio station changes its broadcast pattern and your system suddenly gains a DJ, you've got real troubles. Trying to chase down grounding problems in the middle of a show is a random process, at best. If it happens, try eliminating condenser mics, guitar amp and instrument feeds,*any* unbalanced lines, and AC grounds on signal-level equipment.

One solution to many ground-loop problems is to carry a handful of AC two-prong to three-prong adapters and run around shoving them into every AC cable in sight when a grounding problem shows up.

This is a dangerous practice (as discussed under AC safety), but because it lifts the AC ground it can eliminate ground loops. If you want to keep everything AC grounded for safety, make up a bunch of signal cable ground-lift adapters as shown in Figure 7.

Grounding problems can show up in the strangest places. An engineer I know says he finally solved a particularly elusive grounding problem by removing a chain from a speaker system handle. The speaker connector had been located in the metal handle. When the chain connected the handle to the metal scaffolding, a ground-loop to earth was formed....

The best solution to grounding problems is to prevent them in the first place by having a carefully-designed, *versatile* grounding scheme for the system, and setting it up carefully before each performance. <u>SOUND SYSTEM ENGINEERING</u> and THE <u>AUDIO CYCLOPEDIA</u> have good sections on grounding. If you want some of the deeper theory of grounding and shielding, I recommend <u>GROUNDING AND SHIELDING IN INSTRUMENTATION</u> by Ralph Morrison, published by Wiley.

Oscillations

Ultrasonic oscillations (not caused by acoustic feedback) may be the problem if your system suddenly sounds extremely distorted, and all the VU meters on the console suddenly peg. A faulty component or cable can cause oscillations and the troubleshooting methods for type-A problems can help. Sometimes oscillations happen when there are high (line-level) and low-level (mike) cables running next to each other for long distances (common, but bad practice, is to run high-level power amp feeds back to the stage through the mike snake). During high-level program material (rock-and-roll levels), the mike signals are so high that

Circle No. 114

there aren't any problems, but when you get to the critical soft passage you may have to push up the console faders, increasing the loop gain, and causing the system to oscillate. Just pulling down the sliders a little can often stop the oscillation, but for a permanent fix, the high and low-level cables should be separated.

Acoustic Problems

How many times have we all walked into a hall only to find that face to face communication at more than 20 feet is a practical impossibility? The proverbial "acoustical watercloset" is all too common, and usually, there's not much that can be done about it. Equalization can sometimes help a little. Since most halls with high reverberation times are worst at low frequencies, rolling off the bass a little can help. So can *"stacking"* the high frequency horns vertically since this creates a *"column"* speaker effect (narrow vertical pattern) which helps keep the sound away from the ceiling, thus reducing the reverberation. Careful speaker placement can not only reduce reverberation, it can help reduce severe echos, and acoustic "phasing" problems, too.

Acoustic feedback or "howling", may be the most common acoustic problem. Whenever there's time, try eliminating feedback by changing the mike (try one with a different response pattern), or by using a different monitor speaker, or by changing the relative placement of mikes and monitors. Feedback elimination filters should have as narrow a bandwith as possible so that they take out feedback frequencies, but as little program as possible.

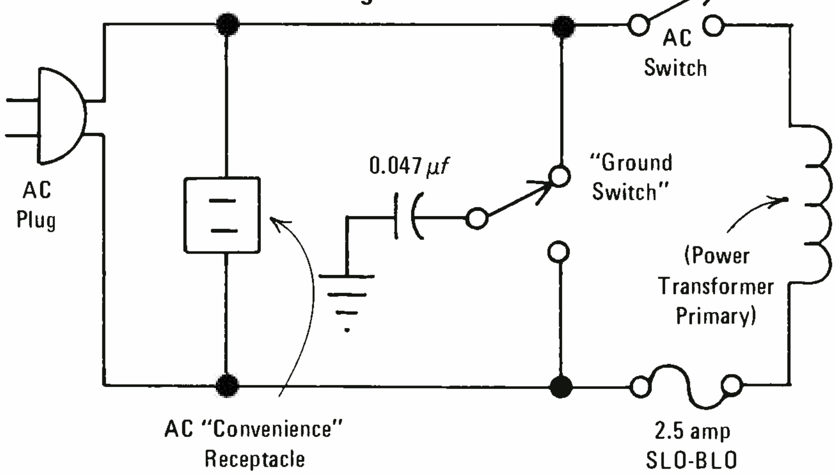
The true professional will *always attempt acoustic solutions to acoustic problems*. Electronic solutions (especially excessive equalization) can often cause more problems than they solve.

Outdoor Problems

It might be thought, that with no echo or reverberation problems, the great outdoors would be the answer to a lot of sound system troubles. Instead, it creates problems of its own. There obviously can be echos, usually from far-away hills, and for extremely large audiences, where a remote speaker stack has had to be used, *artificial* echos will be created (unless you use an electronic time-delay device). The less obvious problems, like moisture and dust can cause failures, and grounding then can be a problem. AC power, outdoors, is extra dangerous. Use your AC voltage checker, and ground the system carefully to avoid shock hazards.

Ground-loops can form between connector (or snake box) shells and moist dirt. Acceptable practice is to isolate them with tape or physically move them away from the ground. Tape over cracks in connectors can help prevent dust and

Figure – 6



moisture from causing problems, and be sure to check with the local weather bureau. One engineer I know, found himself and his system in the middle of a flash-flood during a concert in Utah.

Unfamiliar Systems

This is where a grasp of system design fundamentals, and the ability to read block diagrams really comes into play. If there is a system block, study it carefully, and have someone familiar with the system "walk" you through it before the show. Find out if there are any common trouble spots, and how to access critical cabling, etc. If house, or union rules keep you from working with the system, make sure that whoever *can* is there throughout the concert.

SAFETY

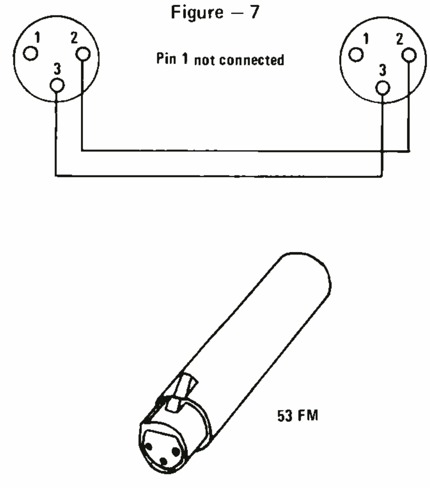
It's pretty well established that excessive SPL levels (above 90 or 100 dB) over a long period of time, can cause hearing damage, though there are opinions to the contrary. If you believe that concert SPL must average 100dB SPL, or higher in order to be exciting, don't bother to invite me; I'd embarrass you with my ear muffs.

AC safety is another story. Opinions don't differ on the need for AC protection, though techniques do. Lifting AC grounds with two-prong to three-prong adapters, it must be emphasized also lifts the shock protection of the AC ground. It's always a good idea to AC ground the mixing console since this also grounds the chassis of all the microphones. Then all you have to worry about are the potential shock hazards from any other piece of equipment.

The "polarity" or "hum" switch on a lot of guitar amps throws a capacitor from the chassis to one side of the AC line. Figure 6 is a diagram of an actual guitar amp "hum" switch circuit. The problem, is that the lowest "hum" position can connect the chassis of the amplifier to the "hot" side of the AC line. Since the strings of an electric guitar are connected (through the shield of the guitar cable) to the amplifier chassis, the performer is connected to the "hot" side of the AC line, with only the series impedance of the capacitor between him and serious shock hazard. A voltmeter or even a neon AC circuit tester connected between the guitar amp chassis and AC ground will show the presence of any dangerous voltages. Ground the chassis, flip the polarity, or use another amp; don't take short-cuts; lives are at stake. One musician I know uses his guitar as an AC safety checker. He plugs the guitar into his amp, turns the amp on (shock hazard exists even when the amp is off), and holding the guitar by its wooden parts, touches its strings to the sound system microphone. If sparks fly, he reverses the polarity switch on his guitar amp. It's hard to believe, but one night, he burned five strings off his guitar! (How much liability insurance do *you* carry?)

The Altec 1220A mixing console has a circuit that I'd like to see on every piece of AC powered sound equipment. When you first plug the console into the AC line, a neon light warns you of any potential shock hazard. Flip a switch, the ha-

Figure – 7



SIGNAL-CABLE GROUND-LIFT ADAPTER
(For XLR-Type, 3-Wire, Balanced Lines)

zard is eliminated, and the light goes out. It's a simple circuit, and it works, but you still have to check every other piece of AC powered equipment.

TROUBLESHOOTING TYPE–C PROBLEMS

If bad components represent hardware problems, and bad design is a software problem, then people troubles must be "mushware" problems; Type–C.

Airline people mis–treat equipment, some stage-hands should be called "stage-thumbs", even your own roadies will plug things together wrong; and, while most artists are considerate, a few will come up with tricks like the following:

The system is at the verge of feedback, but the artist is holding the mic a good 18 inches from her (his?) face, and singing softly (compounding the feedback problem).

The first three rows are emptying fast, but the artist wants the monitors louder and louder.

The producer, who is used to studio mixing, asks you to mix everything "up" into the "red".

So what do you, the system operator, do about this kind of "engineering problem? One answer is to stay out of the political hassles as much as possible. One company I know has a full-time hassle-solver, a "company diplomat" of sorts, who can get on the telephone and sooth furious unions, or fly out to solve a "sound problem" that was really rooted in politics. If you can find one of these types, use him (or her) to the fullest extend possible. That kind of talent is rare in the sound business, but indespensible to it.

Some of the truly technical type–C problems, however, can be solved by inspecting the system yourself before you turn it on. Making the scene just before the show, expecting everything to be set up right for you, is asking for trouble. Check it yourself, and if possible, be there during set up.

THE BEST TEST EQUIPMENT

In the end, it's the built-in test equipment that solves the problems. Your ears, for example, are incredibly sensitive. They have a dynamic range of about 120 dB. They can tell you qualitatively (and with suprising numeric accuracy) the same things that you could learn from a spectrum analyzer, a frequency counter, a distortion analyzer, and a host of other instruments. Sophisticated test equipment is becoming more and more common on the road, but if you carry test equipment, try not to rely on it for the kind of quick troubleshooting we've been talking about. Your ears, eyes, and a little logical thinking will get you to the root of a problem a lot faster.

In fact, as you become familiar with a system, and its block diagram, and you learn to think logically about system problems, a lot of solutions will show up in your mind as soon as a problem appears. Even for unfamiliar problems, logical thinking speeds up the troubleshooting process. As an example, it's much more likely that a bad feed cable from the high side of the electronic crossover is the cause of a sudden loss of high frequencies in the program, than that all of the tweeter diaphragms have suddenly blown.

Logical thinking seems easy.....but try it when the problem happens during the middle of the performance, and the artist, the producer, even the audience are jumping down your throat. One engineer I know, says that when the going gets tough, he slows down more to compensate for the increased tension. It may be hard, but if you keep your cool, the problems will get solved faster.

So now what? You've solved the problems, the show was great, the artist, the producer, and the audience are happy, but who gets the credit? Well, you've always got your own internal satisfaction for a job well done. Think about your pay-check, and the vacation at the end of the tour. And, there can be other compensations; one engineer I know, received a standing ovation after a particularly good concert. Not bad . . . for a roadie!

The SYNTHESIZER as a SIGNAL PROCESSOR

by JAY PETACH, Audiocraft Recording, Cincinnati, Ohio and Fultz Recording Studio, Fairdale, Kentucky

Today synthesizers are commonly found in recording studios, and aside from the usual musical applications, most synthesizers are capable of some types of signal processing. This article will describe how to use the synthesizer to produce some interesting effects.

The synthesizer is actually a number of electronic components which can function as a unit to produce sound. These components are typically voltage-controlled oscillators, amplifiers, and filters, along with envelope generators, ring modulators, noise generators, envelope followers, etc. When the recording engineer considers the synthesizer as a rack full of these components instead of a complicated musical instrument, certain signal processing possibilities should come to mind. By using one or more of these components it may be possible to duplicate the functions of existing signal processing equipment or to create new and unusual effects.

Besides voltage-controlled oscillators, voltage-controlled amplifiers are probably the most common component found in synthesizers. The voltage-controlled amplifier or VCA is also the basic component of all compressors, expanders, noise reduction devices, and automated mixing equipment. Fundamentally the VCA is an amplifier that increases or decreases signal output in proportion to the control voltage applied. The control voltage can be DC, the output from an oscillator, the amplitude envelope generated by an envelope follower, or noise, plus a variety of other possibilities. The limits on the absolute value of the control voltage may vary from brand to brand (e.g. ARP uses ± 10 volts, MOOG uses ± 5 volts, etc.). If the control voltages are generated by the synthesizer itself (as in the following examples) it is safe to assume that they will be within the usable range of any of the synthesizer's components. If for some reason it is necessary or desirable to generate a control voltage from some external source, care should be taken to insure that the control voltage is within limits of the particular synthesizer being used.

By using the VCA as shown in Figure 1, the synthesizer can be made to function as an expander. The threshold can be adjusted by varying the amount of control voltage. The

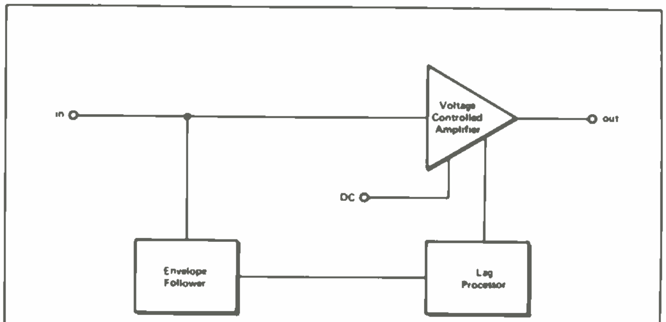


Figure 1: THE EXPANDER

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control voltage in this case is composed of the amplitude envelope from the envelope follower and a certain amount of DC. The synthesizer has a number of sources of DC. Some of these are: The voltage processor, the keyboard control voltage, and the initial gain control on the VCA itself, if there is one. This DC control voltage may not be necessary, but may be helpful in adjusting the expander threshold. The attack and release times can be adjusted if the synthesizer is equipped with a component that can control the attack and decay of the control voltage. One such device found on many synthesizers is a lag processor. Unfortunately with the lag processor attack and release times may not be varied independently, that is they both increase or decrease together. However, this is usually not a great disadvantage.

The expander may be used to produce a gating or Kepex-like effect that works fairly well for cymbals and other percussion. Another application of the expander is in the reverb send or return line. When used in the reverb send line only the peaks trigger the reverb. The effect is similar to that of a delayed reverb with a short decay time on the reverb unit and a lag processor in the control voltage circuit of the expander. With the expander in the reverb return line the reverb is gated. The effect here is that of an unusual reverb decay that can be quite interesting and especially noticeable during stacatto passages.

Substituting a tape machine for the reverb chamber, the expander can produce some unusual effects with slapback. Still another effect can be produced by routing the signal direct to one channel of a stereo mix and routing the expanded signal to the other channel. The result is a location change in the stereo field as the signal increases and decreases in volume. In every case care should be taken not to exceed the dynamic range of the expander and therefore produce distortion.

By routing the control voltage through an inverter the VCA now acts as a gain reduction amplifier (see Fig. 2). In this case, the DC component of the control voltage is necessary. The net control voltage, which is the difference between the positive DC and the negative amplitude envelope, must be positive in order for the VCA to pass any signal. Again the threshold can be adjusted by varying the components of the control voltage, and the attack/release times can be controlled to some extent by the lag processor.

This effect is helpful for creating a very extreme compression that can produce dynamic reversing, among other things. This dynamic reversing is especially noticeable on percussion. It can actually change the rhythm pattern by completely eliminating certain beats. Like the expander this dynamic reverser can be used in conjunction with reverb, echo, or other outboard equipment. Percussion, especially, can be panned automatically by routing the signal to one channel and the dynamically reversed signal to the other channel.

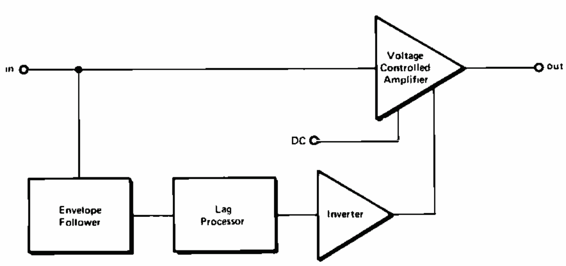


Figure 2:
THE GAIN REDUCTION AMPLIFIER

The voltage-controlled filter or VCF is another component found on all synthesizers. Typically these filters are low pass. In other words they will pass all frequencies below a certain cutoff frequency and will not pass any frequency above the cutoff. The VCF cutoff frequency is not fixed, but will vary in proportion to the amount of control voltage applied, the greater the positive control voltage the greater the bandwidth.

By patching together the various components as in Figure 3 the synthesizer now become an active filter that can be useful in helping to decrease high levels of surface or tape noise. By adjusting the initial filter cutoff frequency and the control voltage, the amount of filtering can be adjusted. In this case also the lag processor may be used to vary the attack and release times.

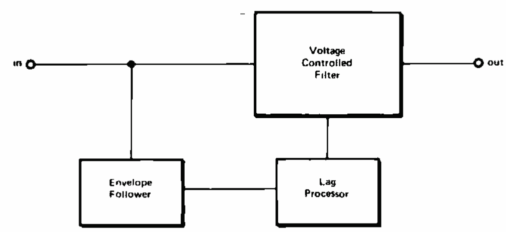


Figure 3:
THE ACTIVE FILTER

The VCF can be used in series with a VCA as shown in Figure 4. When used in this manner it may be possible to achieve more noise reduction with less noise modulation. Note that the lag processor may be used anywhere in the control voltage circuit. It should be mentioned that commercially available active noise filtering devices are extremely more complex in design and generally work better.

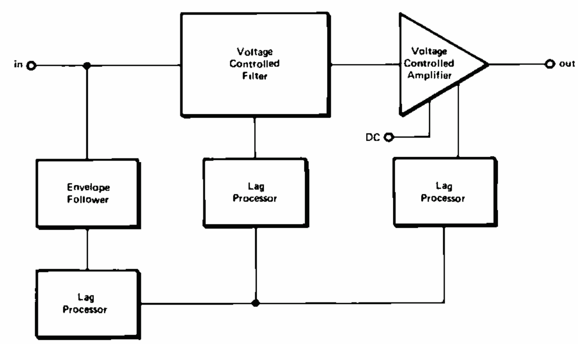
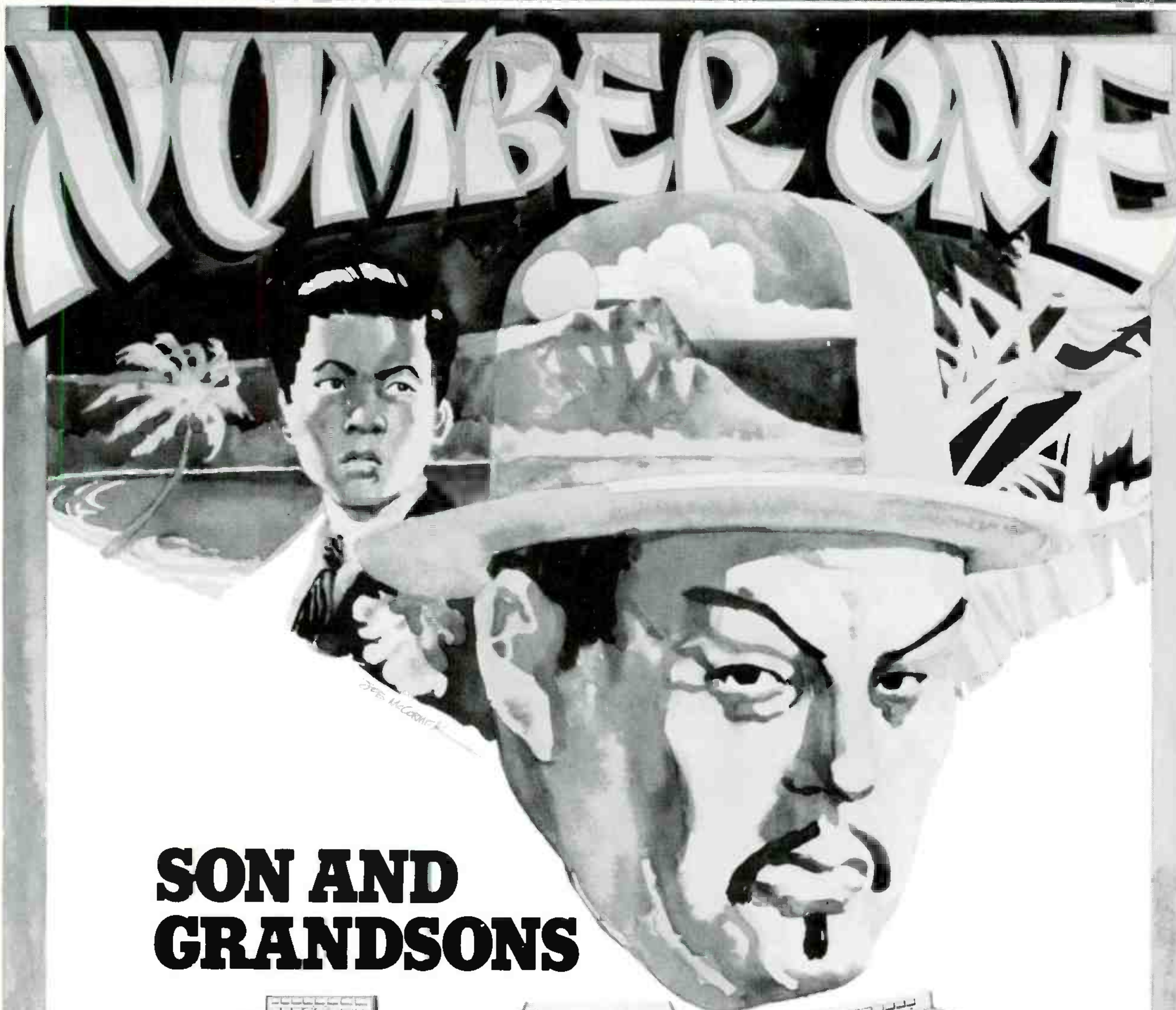


Figure 4:
THE ACTIVE FILTER and EXPANDER

Using the VCA as an expander as shown in Figure 5, one instrument may be gated by another. In this example instrument number one, say a bass guitar, could be gated by instrument number two, a bass drum. The resulting effect is either that of a very tight bass/bass drum section, or the illusion that the bass drum is actually tuned to the proper pitch. The synthesizer's voltage controlled oscillators might be substituted for instrument number one to produce some very unique timbres.



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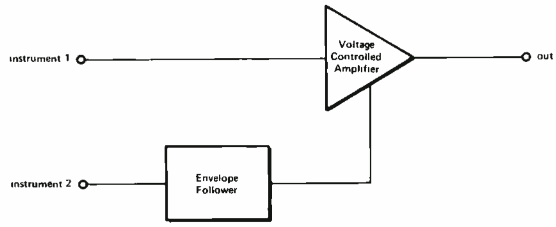

Figure 5:
GATING ONE INSTRUMENT WITH ANOTHER

Many synthesizers have one or more preamps that make it possible to patch an instrument or microphone directly into the synthesizer without first routing the signal through the console. However, since the inputs and outputs of most studio synthesizers are compatible with console line levels, it is possible to execute these effects at any stage of the recording process, including the final mixdown.

By using an oscillator as the source for the control voltage the VCA will produce a tremolo effect when the oscillator is generating a low frequency sine wave (see Fig. 6). The pitch or frequency of the oscillator now controls the frequency of the tremolo. Frequencies below about seven Hertz are suitable for producing the tremolo effect. Above about seven Hertz audible AM sidebands will be produced and the timbre will be changed drastically. The amount or depth of the tremolo may be increased by increasing the ratio of oscillator voltage to DC voltage in the control voltage.

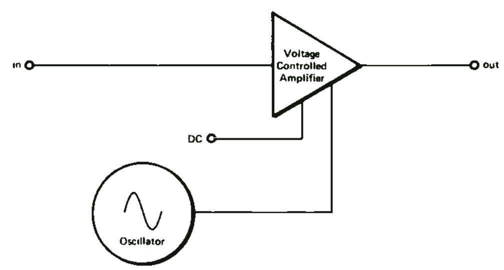

Figure 6: TREMOLO

This effect works especially well on guitars and piano and thus makes it possible to add tremolo, either during the recording session or the final mixdown. When used with an organ a Leslie-like sound can be produced. Unusual effects can be produced by adding tremolo to instruments or voices that would not normally have one.

By using an oscillator with the VCF as in Figure 7a, a

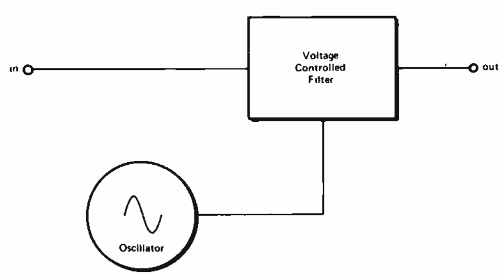

Figure 7a:
WAH–WAH EFFECT WITH OSCILLATOR

wah-wah effect may be produced. This makes it possible to add a wah-wah effect to a signal already recorded.

In Figure 7b the amplitude envelope of the instrument being filtered triggers the wah-wah effect, the louder the instrument the wider the output bandwidth. Several popular devices designed for performing musicians are nothing more than low pass VCF's triggered by the amplitude envelope.

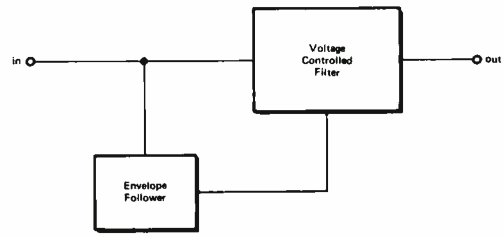

Figure 7b:
WAH–WAH EFFECT WITH ENVELOPE

Note that by adding an inverter in the control voltage circuit the filter will now act in an opposite manner. The VCF will now filter down on louder passages and open up on quieter passages.

The wah-wah effect may be produced by opening and closing the filter manually as a third option.

Many large studio synthesizers have high pass filters in addition to low pass filters. High pass filters will pass all frequencies above a certain cutoff and will reject all frequencies below this cutoff. If the filter is voltage controlled the cutoff frequency will vary with the control voltage. Like the low pass filter, the more positive the control voltage, the greater the output bandwidth. If a high pass VCF is available it can be used in the same manner as described for the low pass VCF. The characteristic sound is closer to a "sproing" instead of a "wah".

If one of each type VCF is available the signal could be routed to one channel of a stereo mix through the low pass VCF and to the other channel through a high pass VCF. With identical cutoff frequencies on the two filters and equal (but opposite in sign) control voltages a guitar or other signal could be wahed, sproinged, and panned in one easy operation.

Many other sounds can be created by using the two different type filters in series to form a band-pass or band-reject combination.

The most useful component of the synthesizer for creating unusual timbres is the ring modulator. This device frequency shifts by producing AM sidebands. The output fre-

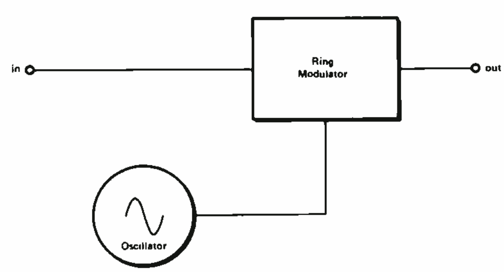

Figure 8:
THE RING MODULATOR

quency is shifted the same number of Hertz as the control voltage frequency.

Figure 8 shows the basic patching scheme for the ring modulator. When the control voltage is a slowly oscillating sine wave, the ring modulator will produce a tremolo like the VCA. The basic difference between the two components is that the output of the ring modulator contains only the AM sidebands. This makes this component ideal for creating mechanical and unhuman-sounding voices and other strange effects. Some mention should be made of the fact that the ring modulator with about a 15 Hz control voltage is ideal for making a $5000 grand piano sound like a $50 upright.

One additional effect that the synthesizer is capable of producing is that of a voltage controlled pan pot (see Fig. 9). By using two VCA's or one VCA and a ring modulator or two ring modulators the input can be panned continuously from channel to channel at the frequency of the controlling oscillator. There are limits as to the speed, since both the VCA and the ring modulator will produce noticible AM sidebands as the oscillator's frequency goes beyond about seven Hertz.

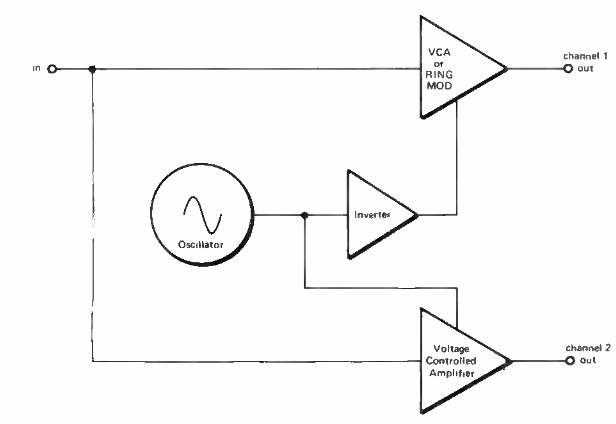

Figure 9:
THE VOLTAGE CONTROLLED PAN POT

Some mention should be made of the fact that the synthesizers designed for live performance are generally not as useful as signal processors, since flexible patching schemes are generally avoided in order to facilitate rapid signal rerouting. However, most studio synthesizers, regardless of brand, are capable of producing the effects described in this article. Most studio synthesizers are also compatible with typical studio signal processing equipment with respect to S/N ratio and bandwidth.

It was not intended that this article represent the synthesizer as a panacea for all signal processing needs. Generally devices designed to perform a specific function, e.g. gating, limiting, noise reduction, etc. work more successfully. However, the synthesizer's signal processing potential should not be overlooked, especially if there's one sitting in the control room just a patch cord away. In addition to duplicating or approximating the functions of commercially available signal processing the synthesizer may be the only way to create certain effects, e.g. adding a wah-wah or tremolo to a guitar during the final mixdown or frequency shifting a voice or instrument during the final mixdown. The engineer/producer who is aware of the capabilities may be able to use this tool to augment existing equipment or to create new and unusual effects.

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Ken Scott

by Howard Cummings

Diverstity. That would probably be the best word to describe the work of KEN SCOTT. He has worked with people ranging from Jeff Beck and the Beatles to Billy Cobham and Stanley Clarke to David Bowie and the Tubes.

With three Grammy nominations and numerous Engineer/Producer credits to his name, Ken doesn't have to say, *'Crisis? What Crisis?' 'Happy In Hollywood'* would be more appropriate.

The interview took place at A&M, Howlingwood, California.

Howard Cummings: I'd like to find out something about your background, electronically . . . musically?

KEN SCOTT: Electronically, none whatsoever. I know virtually nothing about electronics. I know if I turn this knob this way so much, it'll change the sound - why, I don't know and I don't particularly want to know. I've known too many engineers who know everything technically, and they think if they try to make an electronic change, it'll mess up the sound. So it gets in the way of the aesthetics.

Musically? I first got interested at the age of 9 or 10. At school a friend of mine brought in some Bill Haley and Elvis 78's. I borrowed 'em, loved 'em, that was it! From then on I wanted to be a recording engineer. I used to record plays at school and such. The main problem was how to approach recording companies and TV companies. The majority of them said they wanted university degrees so I decided to try for one. I tried seven "O" levels (a degree of educational proficiency in the U.K.) and got only one. So, I thought "I've got to try harder." I stayed for an extra year . . . I *intended* to stay for an extra year. Half-way through the year we took mock exams. I think I sat for two of them on a Friday morning. Friday afternoon I decided I couldn't take any more, so that Saturday I wrote off to record companies. On a Tuesday I heard from EMI, had an interview on a Thursday, was accepted Friday morning, and left school Friday afternoon.

Howard Cummings: Was this around the Jeff Beck TRUTH period?

KEN SCOTT: Way before then — 13 years ago.

Howard Cummings: So you might have worked with Norman Smith? (Beatles engineer from 1962–1965)

KEN SCOTT: I did. I tape op'ed on HARD DAY'S NIGHT within a week after starting at EMI. Also worked on RUBBER SOUL, BEATLES FOR SALE, and HELP. Geoff Emerick started MAGI-

CAL MYSTERY TOUR and I finished it.

HC: Which you were not credited for. I hear it was EMI policy.
KS: Right. I was credited a little on the *double "white"* LP along with a group of other people, business advisors and such.

HC: Could we talk about your involvement on the "white" album? I know you did some tracks with Barry Sheffield. (an engineer at Trident during 1968) HEY JUDE . . .
KS: As far as I remember, they did about three basic tracks at Trident. HEY JUDE, DEAR PRUDENCE, possibly SAVOY TRUFFLE, maybe some others.

HC: What was George Martin's approach to recording the Beatles? Was there a preproduction meeting?
KS: It was impossible with them. (The Beatles). Whoever had written the song would come in that night and teach it to the others. I remember on SEXIE SADIE it took us three whole nights to get the basic track down. One of the things about them was they probably tried every variation possible within themselves until they got it the way they wanted it.

HC: Chording, instruments, fast versions, slow versions?
KS: Yeah. "STRAWBERRY FIELDS" was two completely different versions for example.

HC: I know. Great matching. Were you in on REVOLUTION No. 9?
KS: NO THANK YOU! (Laughter) That was Geoff. (Emerick).

HC: How did "YER BLUES" come about with the double-tracked drums in the solo?
KS: Hmmm . . . it's hard for me to remember that, but on the album, George (Harrison) recorded a song that was never released called "NOT GUILTY". We were experimenting with the kind of effect you get in a theater with a P.A. He put the vocal on in the control room standing between the two speakers. We were feeding *him* the sync head, and *we* were listening to it off the playback head, so we had a form of tape echo which was acoustic as opposed to electric. Anyway I thought it was funny at the time and happened to make a comment to John (Lennon) about "The way you're going you'll be asking to record next door." (In the machine room, set to one side of the control room at EMI, London) So John says "What a great idea!!" We went on and recorded "YER BLUES" *in there*! The room was just large enough to get all the instruments inside, but since it was so small, I couldn't hope for much separation.

HC: Where's the bass drum coming from in the solo of "MOTHER NATURE'S SON"?
KS: Outside of the control room is a long (60') corridor with a set of stairs. When Paul (McCartney) would show up, we would overdub him on bass drum at one end of the corridor, with a 47 or 67 at the bottom of the stairs, so there's this huge long delay to it.

HC: What sort of special mikes would you have used on this LP on things such as strings, guitars, and vocals.
KS: The strings would have been the *old* tube-type U-47. I would have used them for vocals most of the time also, but I vaguely remember them (The Beatles) requesting different mikes to obtain a different kind of sound, so that led to my using a really old mike, of which the number escapes me now. It was something we had also used on bass drum.

HC: Could we talk about the Jeff Beck TRUTH sessions because that was a personal favorite of mine. For me, Beck and that band haven't topped themselves.
KS: Oh yeah, they were amazing! It was strange the way they split up. I do remember that album was very straightforward.

HC: What about the Mickie Most approach to recording?
KS: He knew what he wanted. Ah . . . as far as I remember when we were putting it down on four-track he wasn't there very much at EMI No. 3, but when it came to the mixes, he was there.

HC: Did he leave it up to you to do the lay-down?
KS: No. Peter Grant was there, who was a partner of Mickie's.

HC: Peter Grant and Mickie Most???
KS: Yeah! They used to be very close.

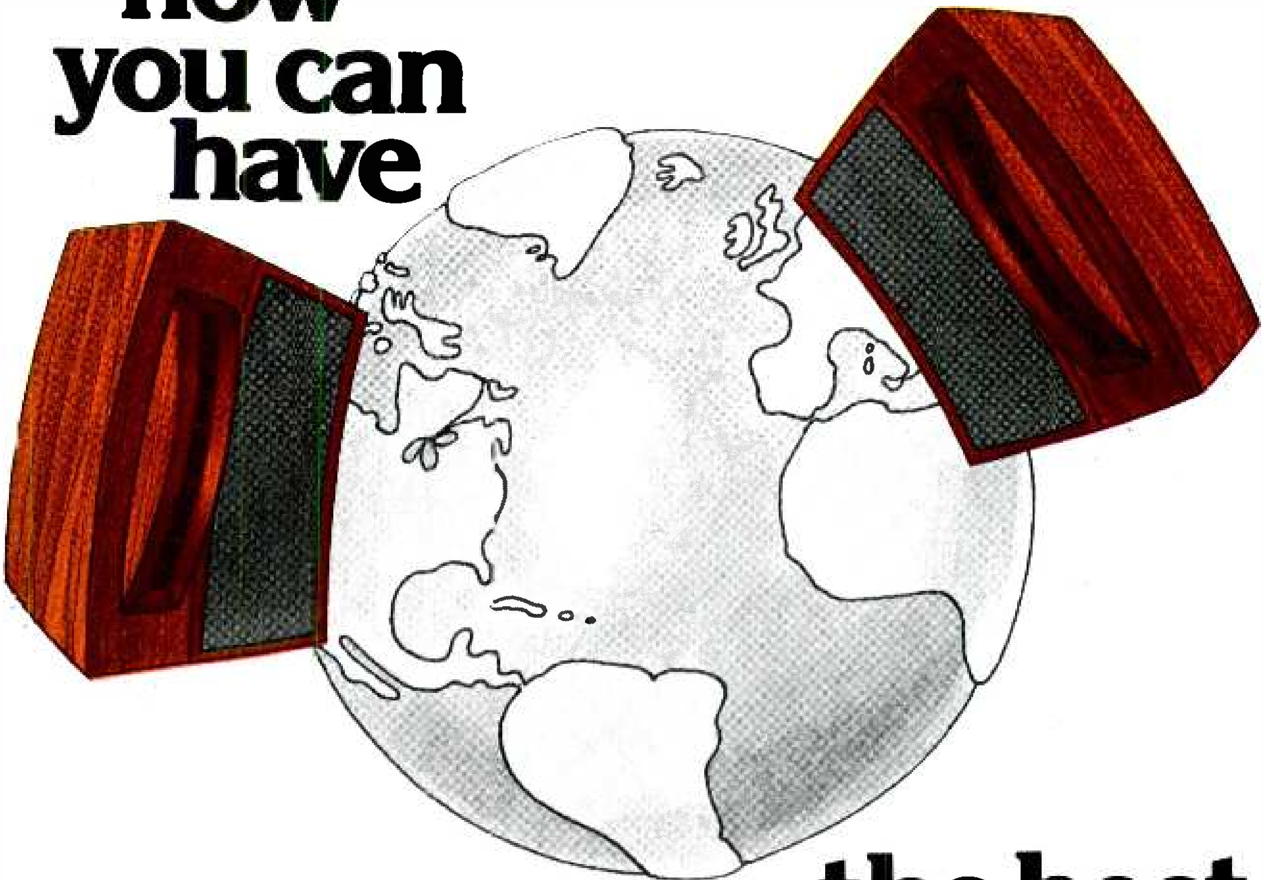
HC: Who suggested material? Did Jeff have things in his head and say "I've always wanted to do AIN'T SUPERSTITIOUS and make my guitar growl"?
KS: Yeah. (laughter) They virtually had the songs prepared. The vocals were done *"live"* by Rod (Stewart) who wanted to move round a lot. I remember the mike was a KM54 with lots of foam.

HC: BLUES DE LUXE–Was it "live" or studio?
KS: Studio. How was that credited?

HC: They said "thanks to the help of boom-boom we can give you a sample of some live blues" or something to that effect. What else was a buzz?
KS: On OLD MAN RIVER, the band was joined by Keith Moon on tympani.

HC: It was credited: tympani by "you

now you can have



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know who". Some people thought that was Beck.

KS: No. One night Keith arrived in his Rolls Royce at EMI with Tannoy P.A. attached to the roof and as he was leaving at three or four in the morning, fully drunk, he started swearing out the window to the whole neighborhood! (laughter all around) EMI was inundated with complaints for a week afterwards! You know, he's even registered at hotels by driving his mini-car up stairs into the lobby. (laughter)

Beck also did one album which was never issued.

HC: The "JIZZ WHIZZ" album?

KS: No, the one he did over here at Motown. Old Four Tops and Stevie Wonder songs. *We* tried to mix it in England.

HC: That must have been his 1971 trip prior to ROUGH AND READY.

KS: Possibly. It was the Motown rhythm section with him.

HC: We're moving into a Bowie phase.

KS: The way that came about was I worked on the first two albums. I think the first was MAN OF WORDS, MAN OF MUSIC which later became SPACE ODDITY and the other was THE MAN WHO SOLD THE WORLD, with Tony Visconti producing. Bowie didn't do much for a couple years afterward ('70–71) but did try producing some singles for other people. I was getting fed-up with engineering and he said "I'm going to be starting a new album soon, would you like to co-produce it with me?" I lept at the opportunity and it became HUNKY DORY.

HC: How did you feel about the PIN UP'S Album? Did you think — "flashes of nostalgia" or, it might be a bonb in the bad sense of the word?

KS: To start off — a little apprehensive, but then as it went on I thought "great". Before we started off we reviewed the original songs. Some of Bowie's versions I think turned out better.

HC: Which? ROSALYN, any of the Who or Yardbirds stuff?

KS: I loved the way "CAN'T EXPLAIN" turned out. That was weird. It was exactly the same as the original. Did you ever hear of Johnny Kidd and the Pirates from England?

HC: SHAKIN' ALL OVER?

KS: Right. Well the guitar over the sax solo in "CAN'T EXPLAIN" is the same as "SHAKIN". We didn't want to copy it exactly so we slowed it down and it became more of a Phil Spector thing. "HERE COMES THE NIGHT" and "SORROW" I liked a lot. It was an album of English big-hits that were obscure here in the States.

HC: I heard there was enough material left over for a "PIN–UP'S — Volume 2".

KS: (Puzzled look) ???

HC: No??? Wild rumor!!!

KS: Yes! (laughter) There were as many cuts as appeared on the LP.

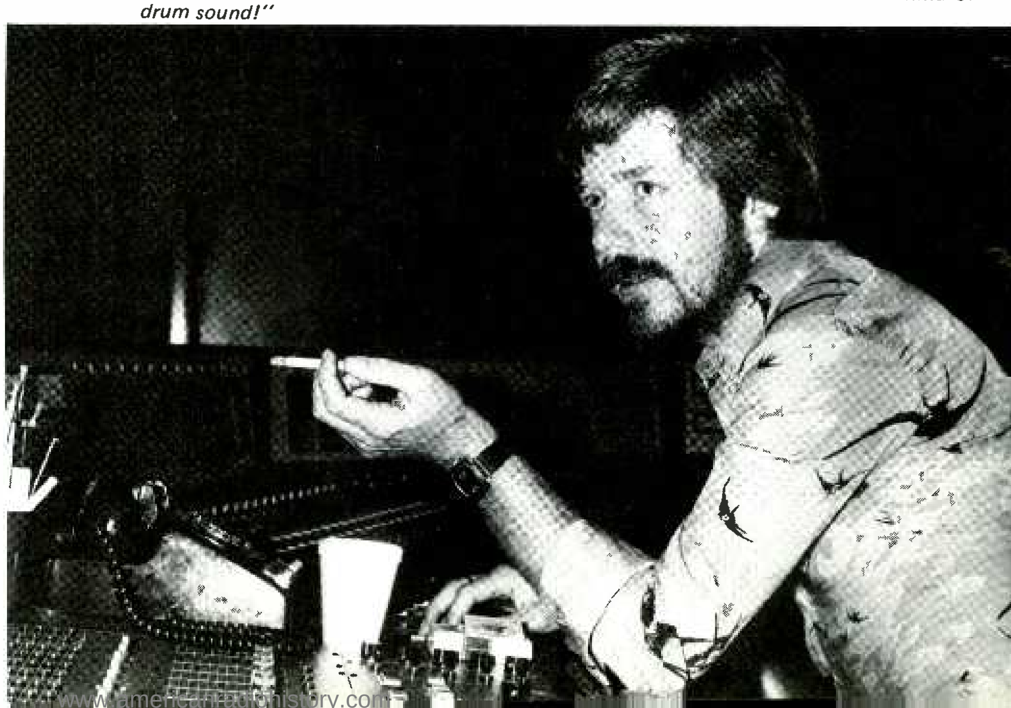
HC: What about your vocal mike selection for Bowie.

KS: With David it was normally U67 for vocals.

HC: Did you find that he would like to experiment a lot with vocal mikes for special effects?

KS: No, not really. The one thing we sometimes did was to mike him some distance away, and more so we did that

. . . the wire suspended mikes . . . "its something I had to do when I was recording Tony Williams. He doesn't like to have his drums damped down . . . suspended the mikes from the nuts inside the shell . . . it worked well . . . it was a different kind of drum sound!"



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HC: Like on the sax solo of "CAN'T EXPLAIN"?
KS: More than likely. More than likely.

HC: Could you talk about your use of drum mikes on PIN–UP'S with Aynsley (Aynsley Dunbar - Bowie drummer 73-74) versus that of Woody (Woody Woodmansey - Bowie drummer, 71-73) on the ZIGGY album, because I notice more definition, better clarity, and more of an "up-front" mix.
KS: Exactly the same mikes – exactly the same placement. As far as I remember when I was doing ZIGGY I was into what most L.A. drummers seem to be into at the moment and that's putting lots of damping on the drums. By the time I recorded Aynsley I'd already recorded Billy (Cobham) a couple of times and I'd got a much *"liver"* drum sound so we used virtually no damping for Aynsley. The size and brand of their kit also had something to do with it plus we worked in two different studios. (Trident and Chateau)

HC: What sort of special processing did you go through in developing the sound for the DAVID BOWIE – MIDNIGHT SPECIAL TV broadcast? (NBC–1973)
KS: The Marquee Club (London) has a recording studio behind the club itself. I was recording the actual performance while David was being filmed in the club. Once it has been recorded I took it back to Trident and mixed it there. It was a 16–track recording and I think I used about 20 or 25 mikes total; 8 or 9 on drums, bass was direct, one mike each for the two guitars, vocal mike, and backing vocal mikes.

HC: Did you mix on small speakers and what sort of EQ did you use for TV consideration?
KS: I didn't take any EQ into consideration for TV. I may have mixed on small speakers, I don't remember, but just the fact of doing it on small speakers would take care of the EQ. I remember we had just returned recently from the Chateau from doing PIN–UP'S so the instruments of the group were pretty together. I feel that to a degree the show was a farce because the TV production crew was far more into the visuals trying to get by the censor than they were about the sound. There were instances where, musically, the take was terrible, but on occasion we were able to persuade them (the crew) to re-do those bits. There was another occasion where the music was great but Bowie was wearing a costume that was cut very low and you saw a fraction of his pubic hairs. The crew thought the censors might object to it and we had to do another take. Towards the end of the song, knowing it wasn't nearly as good, Bowie totally messed about so I think in the end they had to inter-cut between the two performances.

HC: What next? You got involved with Elton John's MADMAN and CHATEAU.
KS: I had been getting on rather badly with a new manager at EMI and was talking to Gus Dudgeon at a session. He said "Look, I use Trident a lot. Barry Sheffield wants to get out of engineering. Move over there and I'll have a built-in engineer." I went over there.

Robin Cable recorded all of MADMAN and I mixed it.

HC: How was Gus to work with?
KS: Very easy. He knows what he wants.

HC: Do you think it's an engineers' job to throw ideas around or keep the sound-end down pat?
KS: I think it depends entirely on the engineer. I've seen some who make suggestions while they sacrifice the sound-end.

HC: Do you usually follow the whole recording chain from lay-down to mix to refs to lacquers?
KS: Oh, you *have* to. I have a say in the final product.

HC: I always used to wonder about George Martin; the ALL YOU NEED IS LOVE session on the guitar solo. I wondered why he approved it.
KS: It might have been because of the TV broadcast.

HC: Preserve the "authenticity" I guess.
Were you kind of apprehensive about recording in France for "Chateau" because the sound is pretty respectable.
KS: No. Gus had been an engineer and had looked over the place before we went in. It was done for tax reasons primarily and it was something different.

HC: Were the sessions Dolby?
KS: Yes.

HC: Do you notice of Dolby affects cymbals on drums, like a loss of harmonics on Nigel's kit? (Nigel Olssen was the drummer with Elton John at that time)
KS: When I record drums and then play them back, agreed, there isn't as much top as there was. But then you use tones and it reads perfectly.

HC: How about the choice of vocal mike for Elton?
KS: Elton would have been an AKG C12A.

HC: How was Richard Perry to work with? (Son of Schmilsson)
KS: HELL. One of his things is his

amazing memory. When I worked with him there was never a perfect basic "take". So he used to say "Let's use take 3 or reel 7 for the verse and take 5 of reel 2 for the chorus" and he was right on!

HC: Could he be hard on you?
KS: Yes.

HC: Is there anyone you admire as far as engineers or producers?
KS: I find I rarely like people for specific albums. I'll listen to an album and say "that's nice", then here something by him again and say "that's nice". Consistency is the thing. My favorite producer at the moment is Arif Mardin.

HC: How did you get involved with neo-jazz-rock people?
KS: It's very weird. I have to now relate back to Elton John in France. In the dining room of the Chateau, we had a record player set up. During meal breaks, we used to play Mahavishnu records. I used to hear snatches of it and thought they must have been doped out of their heads! I got a call from England later from CBS that Mahavishnu wanted to work with me. I listened to an album, was totally knocked-out and got involved with BIRDS OF FIRE, Billy Cobham, and John McLaughlin. With them, everything's bigger. Cobham's drums - enormous! 13 toms on the new Stanley Clarke album we're working on now.

HC: Considering the size of Cobham's kit, how do you deal with the quantity of his drums and how would you assign the tracks?
KS: As an example I'll use the new Stanley Clarke album. He had two bass drums, snare, 13 toms, and I don't know how many cymbals! Each of the bass drums would be on its own track. The toms and cymbals would be across four tracks, thinking of quad, so I could position it where each track would be in one of the corners. If he did a roll from the highest tom to the lowest, it would go from front-right, counter-clockwise, and end up just to the front of front-right. And then the snare would be on its own track so we'd use about 7 tracks of the 24-track. Oh, and Billy has an assortment of electronics gear for special effects, which he plays with his feet.

HC: Is it a synthesizer device like he used in his Roxy appearance?
KS: He's got an Echo-plex. He's got an Eventide Phase-Shift, and what's the one that makes the sound go backwards?

HC: Omnipresser?
KS: Omnipresser, that's the one!

HC: What about mikes?
KS: The same as I always use!

KS: The Norths?? (Manufacturer) North drums. Those were a bit of a problem 'cause there's two distinct sounds coming from the skin and that coming from the open end of the tom. The way it worked —we thought we'd have to use two mikes, but we miked those toms at the bottom and picked up enough sound from the North's along with the ordinary toms.

HC: Was the balance a particular hardship mixing all the variations in tom tonality?
KS: Not really. We just got the toms so they sounded equally balanced. We were lucky.

HC: On a 1974 Stanley Clarke album, one specific track "SPANISH PHASES" — knocked me out.
KS: Yeah. It wasn't decided — the title — until the album. We recorded the orchestra first one afternoon, just short pieces . . .

HC: Before Stanley???
KS: Right. When it came to putting the bass on, I told the tape-op to put a minute-and-a-half of tape between each orchestral section. Stan went into the studio and played straight through—it was as if he knew when each of the passages was coming up!

HC: The same treatment for his "curved" toms?

HC: Great timing. What specifically knocked me out about that was Stanley's bass -- the reality and dynamics! It was as if I could see the "windings" on his bass strings!

KS: Right. We tried it on three tracks; a Neumann 47, a direct from the Poly-tone pickup on the bridge, and a mike on his amp. We finished using the pick-up only. Great pick-up! The Stanley Clarke album we're working on now is going to be amazing, with Billy Cobham, George Duke, and a guitarist named Icarus!

At one point on Stanley's new album there is an acoustic track with Stanley on upright bass, and McLaughlin (Mahavishnu John) on acoustic guitar. We overdubbed some triangle and used, I remember, the Eventide Clockworks Harmonizer on it.

We fed part of the signal back onto itself, so you get the original signal plus a semi-tone plus a slight delay, then signal plus a semi-tone plus a semi-tone. So by putting the triangle through it, we obtained the sound of a bell-tree.

HC: You got involved with Supertramp. I hear they had recorded one or two albums, then you got involved in CRIME OF THE CENTURY.

KS: Right. I had heard some of their demos and said "no thanks", basically because I was hearing only bits and pieces on the demos. I went to hear them at a rehersal and all of a sudden -- it clicked! Hearing whole songs as opposed to bits and pieces.

HC: Was one of those songs SCHOOL?

KS: Right. SCHOOL was one.

HC: Great bass -- direct through the board?

KS: Right. I always take bass direct. I've now got used to the *clossness*. SCHOOL had some limiting possibly 5 dB, some EQ . . . but I always try to get it from the instrument being played. Sometimes I use an 8-band graphic with cut at 200-300 Hz. It's good for electric bass and bass drums.

HC: Do the guys get impatient when you tell them "EQ your own stuff."?

KS: They're used to it. Sometimes I strip down the drums for tuning -- it can be hard. We felt on "CRISIS, WHAT CRISIS?" we didn't want drum continuity. So the drums were tailored for each song, as opposed to CRIME OF THE CENTURY, which was basically all the same, except for two tracks that had to be re-recorded through the cans. Once on a Stanley Clarke LP, Tony Williams wanted to re-record his drums because he felt his kit was too damped. He couldn't *"feel"* it properly.

HC: Do you ever worry about a NAB–CCIR clash -- recording and mixing in different countries?

KS: I always *mix* in the same place, usually Trident. I wasn't used to Electric Lady acoustically and when I tried some mixes there, and played them back in England, I was so brought down.

HC: Frequency?

KS: Right. Not enough high-end. Had to re-mix at Trident. I've also worked out of Scorpio, (London) and now I'm here at A&M.

HC: Could you compare the facilities of Trident, Ramport (London), Scorpio (London), Electric Lady (NYC), and A&M. Boards, acoustics, monitors?

KS: It's something up there (points to his head) that links them. Towards my last days at Trident, the monitoring at Trident and Scorpio was the same, which is Cadac speakers and they're amazing!

HC: I hear they have them at Musicland – Munich.

KS: Have they?? Yeah, *they're* the ones. I don't know who the hell handles them here but they aren't distributed properly. Their desks are the best make of desk in England. I get on with them well. Ramport, Electric Lady . . . I get on well recording, but I don't like to mix there because I'm used to Lockwood's (speakers) and Cadac's. They're both very smooth speakers. JBL's are OK for recording but I can't stand mixing on them.

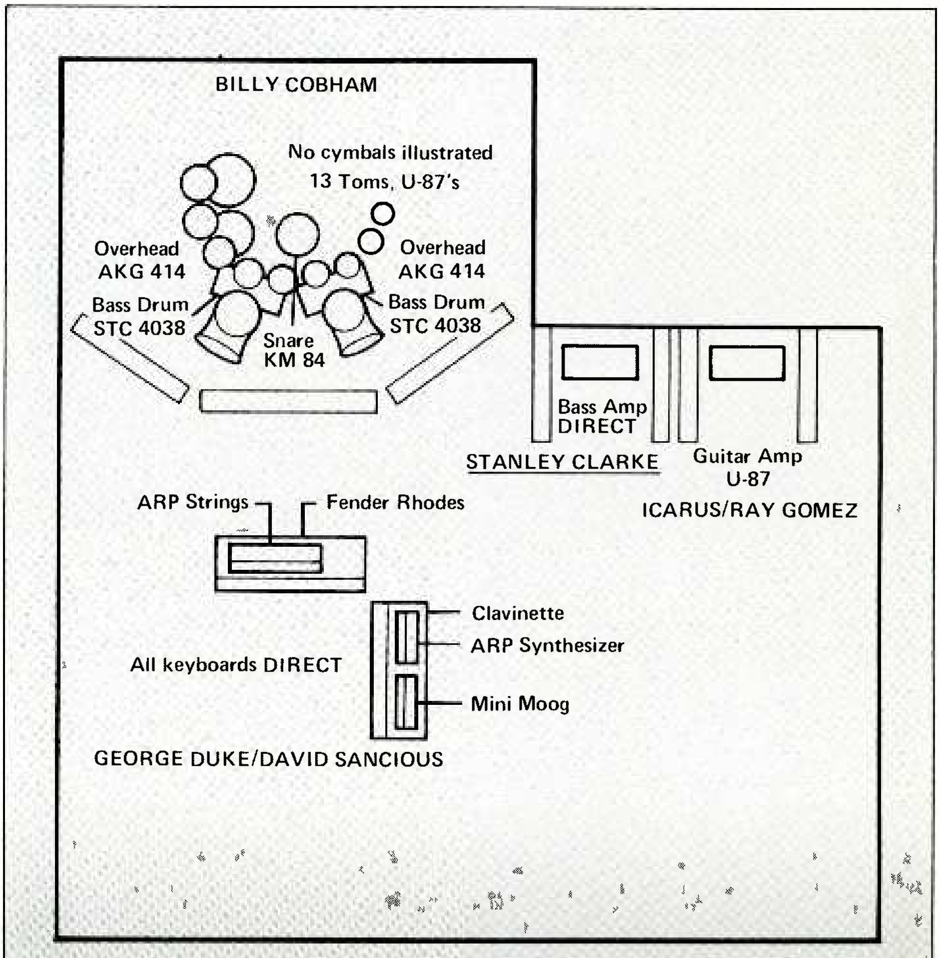
HC: Because they color?

KS: (Pauses) Yes, if you're not used to them. I hear it in the mid-range. Trident, Scorpio, and here (A&M) is much smoother for mixing. "Chateau" is Lockwood speaker.

HC: Does A&M still have the Altec 9846 speakers or the Tannoys?

KS: They now have 604E's with Vega bass units. Mastering Lab cross-overs, and UREI third-octave equalizers tuned the way they like them. But now, I totally re-tune them when I go in there. The blue marks on the equalizers are theirs and the red marks are mine. (laughter)

I was interested in your comments on Munich because I'm looking for a studio



STUDIO SETUP FOR STANLEY CLARKE LP (as yet unreleased) A&M Studio D, Hollywood for the next project, but all I'm familiar with is Deep Purple and the Rolling Stones from there.

HC: Electric Light Orchestra's Poker LP, Donna Summer's Love To Love You Baby, Zeppelin's Presence . . .
KS: Ah, starting to make a little more sense.

HC: Let's return to your description of the rooms of Ramport, Scorpio, and Electric Lady and how they relate to your sound.
KS: Ramport is an old converted church, very "live", it's a typical church hall.

HC: Stone walls?
KS: I believe it's wood. But it's very "*live*", which for some things is great. It's also large and rectangular in shape. I've only *recorded* some Tramp (Supertramp) over-dubs there. Scorpio is much squarer and much more dead. I've done a little over-dubbing there, but mostly mixing. Electric Lady has a very peculiar shape, kind of hard to describe. Some of it is curved, some of it straight, but it all relates to the acoustics and helps with the sound. When I lay out the musicians, I keep them fairly close so they can develop a *feel* for each other when they play.

HC: On the David Batteau album, what is the ethereal sound at the end of DANCING ON ATOMS? (At this time George Duke, keyboardist on Stanley Clarke's latest album, interupts. After exchange of greetings, we proceed.)
KS: Ah . . . Moog.

HC: Any special processing for that?
KS: Direct into the board. To handle the dynamics I may use the limiters a little bit harder.

HC: I noticed you used additional musicians on the TUBES album. Why was that?
KS: Certain numbers need certain things like sound textures.

HC: How did these guys like Don Randi, Wechter, and Paich inter-act with the band?
KS: Ok. They were a part of the orchestra. Jack Nitzsche also did some conducting. There's quite a difference between him and Paich. When Paich was conducting on the TUBES LP, it was very formal, at the podium and all that, whereas Nitzsche would sit in the corner and if there was a wrong note, the musicians would come over and tell him! (laughter all round) A more casual approach.

HC: Why the change from Al Kooper producing the Tubes to you?
KS: (pauses) Al Kooper knew very much what he wanted to hear to the point where it became very dictatorial at times. I first met them after seeing them at the Roxy while I was working on the CRISIS album with Supertramp. I was knocked out! I was wishing I could produce them. Kip Cohen arranged a meeting which I was terrified about but it came out all right. I didn't know what to expect after seeing their show.

HC: No Quay-Lewd at the meeting?
KS: No, they were great.

HC: Have you used as extensive a separation technique on albums other than the Tubes?
KS: Somewhat. Except I have to approach the jazz-rock things differently. It's not quite so easy because it's "then and there".

HC: Immediate capture.
KS: Right. You have far more musicians playing at one time. There are far fewer over-dubs and you can hear the outcome right away.

HC: As opposed to "building" something.
KS: Right. The way I mix is on most things I can't stand to go all the way through and try to get every move right. I normally get it in sections, verse, chorus, edit, etc.

Continued on page 44

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PARTIAL DISCOGRAPHY REFERENCE

Artist	Title	U.S. Release Date	U.S. Label
Jeff Beck Group	TRUTH	1968	Epic
The Beatles	"WHITE"	1968	Apple
David Bowie	HUNKY DORY	1971	RCA
Elton John	MADMAN ACROSS THE WATER	1971	MCA
Elton John	HONKY CHATEAU	1972	MCA
Mahavishnu Orchestra	BIRDS OF FIRE	1973	COL
David Bowie	PIN-UPS	1973	RCA
David Bowie	SPACE ODDITY/THE MAN WHO SOLD THE WORLD	1973/70	RCA/MERC
Billy Cobham	CROSSWINDS	1974	ATL
Billy Cobham	TOTAL ECLIPSE	1974	ATL
Supertramp	CRIME OF THE CENTURY	1974	A&M
Stanley Clarke	STANLEY CLARKE (Spanish Phases LP)	1974	Nemperor
Supertramp	CRISIS? WHAT CRISIS?	1975	A&M
The Tubes	YOUNG AND RICH	1976	A&M
David Batteau	HAPPY IN HOLLYWOOD	1976	A&M

GRAMMY NOMINATIONS

Elton John	HONKY CHATEAU	1972
Nilson	SON OF SCHMILSSON	1972
Supertramp	CRIME OF THE CENTURY	1974

Continued from page 41

HC: And you cut everything together?

KS: Right. The reason I got into it was because of Bowie on HUNKY DORY. There was only me in the studio so I tried it in sections.

HC: I noticed some of your mike techniques on this Tubes album also.

KS: On YOUNG AND RICH we suspended mikes inside the bass drum from wire.

HC: How did you prevent vibration being transmitted from the bass shell?

KS: It worked fine. There was a slight sway, but no problems.

HC: Why did you use wire-suspended mikes?

KS: It's something I had to do when I was recording Tony Williams on an earlier Stanley Clarke album. (SPANISH PHASES LP) Tony Williams is very much a *live* drummer as opposed to a studio drummer, but unlike a lot of other drummers, he doesn't like his drums to be damped-down at all and that includes bass drum and snare. He likes to get the *rebound* you get from the skin when it's not damped. We did a number where he had to do a lot of snare rolls, so I removed the existing damping and it started to happen! It worked out really well and we went back and re-recorded the numbers me and you talked about before. Now so that he could have both skins on the bass drum and I could get the impact of the bass drum beater, we messed about awhile suspending mikes (RE-20) from the nuts that are inside the shell. It worked well – it was a different kind of drum sound than I was used to and I really liked it and it carried over to THE TUBES using the STC 4038.

HC: Could you describe the STC 4038's and the Vanguards?

KS: I think the STC's were BBC designed figure-eight. The Vanguards were condensor and very realistic. I found I had to use very little EQ. They were the way I wanted to hear the toms which is fairly full but with edge to them as well. The STC's are fairly smooth . . . but with "depth".

HC: What about this Albatronics parametric? Is that a U.K. brand?

KS: Just trying to think . . . I'm sure. Yes.

HC: Is this something A&M had in stock or something you carry around with you?

KS: Oh I don't carry around anything with me.

HC: Another thing that surprised me was your use of RCA 44's for cymbals.

KS: It worked!

HC: I hear it! A lot of EQ or any over-loading problems?

KS: I generally find I roll off the bottom in the lay-down, around 400 Hz. I usually add some at 12 kHz. No over-loading problems.

HC: Why did you use 44's on THE TUBES album?

KS: I used to use Beyer 160's, which are double-ribbon. I've always prefered ribbons for over-head because they're smoother than using a condensor. At A&M they only have one M160 so I had to use whatever ribbon they had left, and the RCA 44's were it. The whole thing of choosing mikes has built up for me over a long time. There have been times when I've been working in one studio, and there was a massive session going on in the studio next door. If they were using a lot of mikes, I would be left to experimenting with the remaining mike inventory. Ribbons I also love on brass. Once again they're much smoother than condensors.

HC: How do you feel about computerized mixing?

KS: Hesitant . . . Gus Dudgeon was doing some Elton John stuff experimenting with it and when he played the mix back, everything was at full volume for the whole title!

HC: Was it set up properly?

KS: Yes, but there was something obviously wrong. What I'm looking for is computers to take over from tape machines, which won't be too long. There will be fewer signal-to-noise problems.

HC: Lower distortion?

KS: Apparently, but by that time they will have found a *new* type of distortion . . .

HC: (laughter) Third-band, tri-tronic . . .

KS: Right! (laughter) I've heard some of the old Caruso recordings computer reprocessed and they were amazing. (See RE/P April'76 p. 57)

HC: How do you feel about quad?

KS: The closest for me is CD-4. *The* way for quad really is four-track tape.

HC: DBX?

KS: Once again I haven't tried it. I rely more on what I've heard from others. I think at AIR, London they tried some tests with DBX and at the edit points, if there were any level changes, it threw the DBX off completely.

HC: How do you feel about 24-track?

KS: I use it all the time.

HC: No problems with cross-talk and signal-to-noise?

KS: It's a bit worse. I tr[illegible] not to use the outside trac[illegible] 24.

HC: You're involved in [illegible] capacity. Have you taken it upon yourself to fight for promotion of your artists?

KS: Up until now, I've taken an interest but haven't fought to get it. I think the way it's going to get done is in the contract when I sign with an artist. For example, all over the world, Supertramp have done well – gold and platinum – but here, pfft.

HC: How have you accomplished your success?

KS: Patience . . . from what I see in the press or general talk, that seems to be the way THE TUBES see me – as having an excessive amount of patience. Luck, of course, would enter into it.

HC: Any comments on your grammy nominations for . . .

KS: Yes, I should have won!!! (laughter) Comments? I was flattered . . . I hadn't listened to too many of the competitors but I had heard Gino Vanelli – which was great.

HC: OK, thanks.

KS: Cheers.

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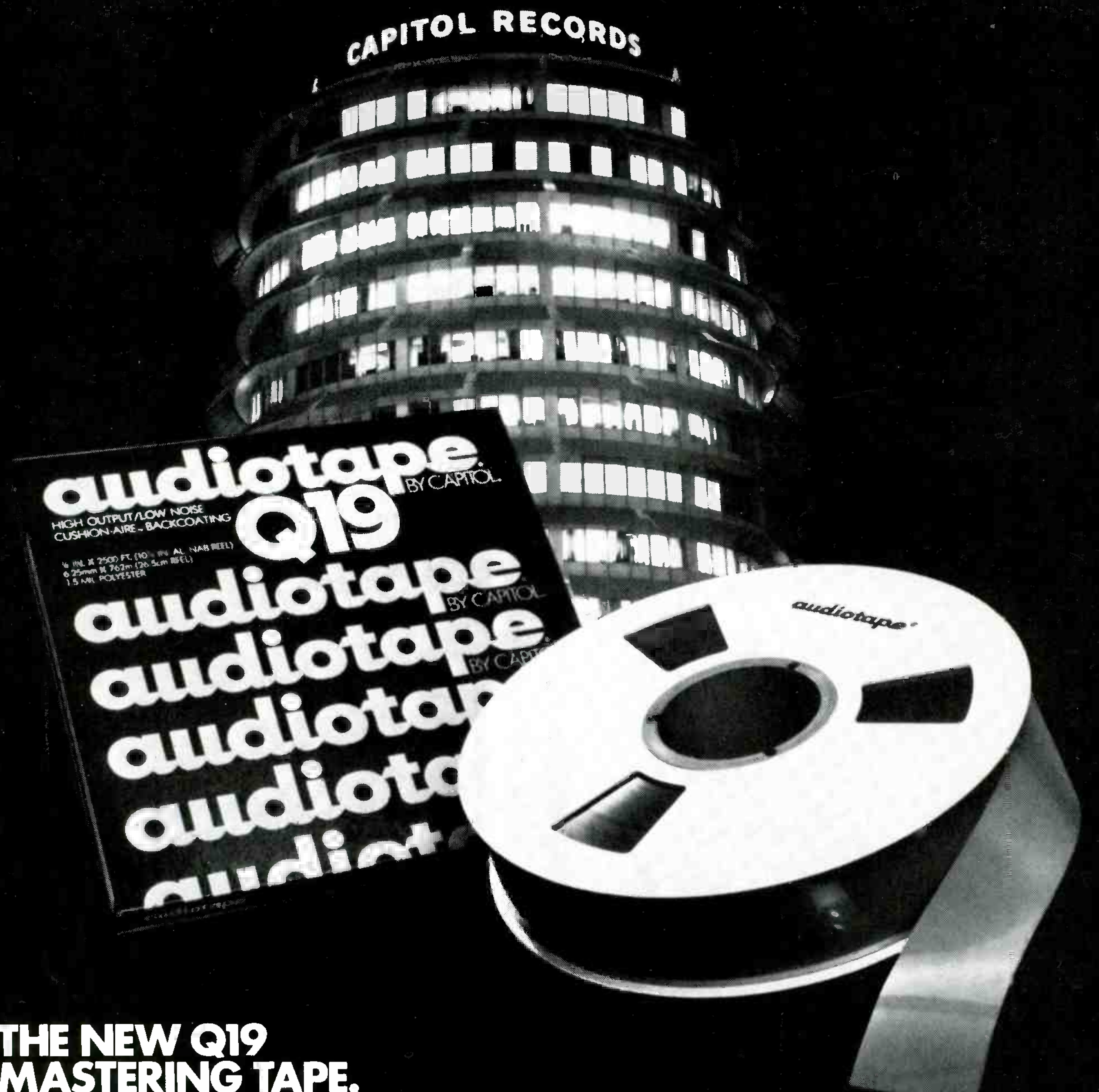

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Circle No. 131

The Smaller Studio

TAMPA St. PETERSBURG

a roundtable

. . . the story of the small recording studio is many stories . . . a 30 second fried chicken spot *("hey, can't you get the names of my store managers in at the end?")* hawking a $1.59 dinner bucket special on local radio *("say they're always smilen'!")*

. . . an order for 500 *("maybe we should get 600!")* album copies of the high school's Christmas concert you recorded *("when do we have to decide on which one of these 48 full color stock jackets . . . last year the band mothers sold chocolates!")*

. . . the thousands of demos for local groups at your most friendly price *("and, yeah, figure on about 15 tape copies we're gonna send/take around to these dudes in New York [Chicago, L.A., Nashville] . . . and you can sure bet we'll tell 'em where we recorded this at!")*

After Ken McArthur of Recnac Recording in Tampa originally suggested the subject, and then in R-e/p's contacts with many other *off-the-L.A.-Nasville-Chicago-New York-recording-axis* studios there was, indeed, found a distinct kinship of experience among smaller studios. What follows, is the distillation of conversations between four competitors in the Tampa/St. Petersburg area of Florida.

Does this discussion give you some ideas? Does your area want to be heard from?

Let us know how you feel about some of these subjects . . .

THE CAST

JOE BANANA – owner of Joe Banana Productions an eight track Tascam studio. Also, doing production work with Capital Records.

KEN McARTHUR – owner of Recnac Recording Studios a four track Tascam studio. Also involved in commercials, slide presentations, jingles, and music productions. In the process of setting up a publishing company.

MEL BURMAN – owner Professional Broadcast Productions, four track studio specializing in visual and broadcast production.

PAUL HAYES – owner Paul Hayes Productions, an eight track studio doing commercials, slide and film presentations, jingles and music production.

JIM COTTLE – Head Engineer for Paul Hayes Productions.

Photography: Murray Shank

ED: In discussing worst-case recording situations, it was the consensus of the group that difficult jobs almost always related to the equipment available, as well as equipment limitations. Generally, the greatest number of jobs did not present unusual demands. However. . .

MEL BURMAN: I had a job a couple of months ago . . . it was practically a charity job at that, a Temple out on Bayshore Boulevard. They had a music festival, and as I started to put the first mike up, I thought the orchestra leader was going to have a hernia. He got so upset because I was going to put mikes up in front of his violin players . . . he absolutely refused to let me put any microphones up.

I was determined to get a recording, so I found where the P.A. system was patched in and fortunately I patched in that way. It wasn't very good recording. Of course, I didn't have control of the levels, but I did get them some sound.

PAUL HAYES: In broadcasting, the saying goes, "when in doubt, plug into the P.A. system."

KEN McARTHUR: At least you have something there.

MEL BURMAN: Of course, you get all of the noise of the bad P.A. amp. You just hope that the amplifier that they have is clean enough so that it will give you a fairly decent recording.

JIM COTTLE: A lot of times what I have done, in a situation like that, for instance at an opera we did, where they have got to see the performance as well as hear it, I just took a piece of foam rubber, about an inch thick, and X/Y'd the mikes over the foam rubber on the floor. You can't see the mikes. It worked out really nicely, and you get good stereo separation.

BURMAN: Was it dense foam, or really light foam? Does it make any difference?

COTTLE: It was just something to keep the mikes off the floor. Of course, you don't have the head in the foam.

McARTHUR: There is a company that makes a special floor style mike holder.

COTTLE: The "Mike-Mouse," I think from Electro-Voice. It's the same principle. But, I'd like to say that I've found the experience of location recording was a tremendous asset for walking into a studio. Like, how in the world are you going to isolate a piano from a drum-set On location a change in mike position can make a *difference* in the sound. The difference is that in the studio a change in mike position can make a *better* sound.*

ED: The conversation moved around to the desire and need for more tracks.

COTTLE: Maybe 16 tracks would be a whole new ball game, but between two or four or eight . . . I think that you get a better location recording on two tracks than you do by trying to put it on four. I did location recordings for about three years. There were instances where I used more than three mikes but, they were pretty rare . . . and I came up with some fine recordings. Most times I would use two mikes and if I wanted to bring up the level of, say, a base guitar and I knew that I am going to have trouble with phasing because of it being so far away from the two stereo mikes then I would mike that with a third mike and put it in mono and bring that up. Just a little bit so that there is some monaural bottom. Then when you combine the two stereo signals they won't cancel out in the disc mastering process.

ED: On the subject of more tracks for studio operations:

HAYES: One of the things that amazed me was that at the time I bought the eight-track people had been saying, "Why don't you get an eight-track . . . we could do all sorts of things . . ." So, we purchased the eight-track and then the same people said, "Boy, that is beautiful. It does a great job; *now lets go back and talk about two tracks.*

McARTHUR: You mean they just didn't have the extra money?

HAYES: Either that or they felt that you had done such a great job on two tracks, why pay the extra expense. People would come into the studio because it *was* eight tracks, but would end up doing either mono or two track. And, the funny thing is we charge the same rate whether we do eight-track or mono. The only difference is in the amount of money for the tape.

**An excellent discussion of floor miking techniques may be found in Chapter 20, of MICROPHONES: Design and Application by Lou Burroughs. (This volume is available from R-e/p Books – see classified ad under "Books".*

COTTLE: I think the flexibility that you get with a higher track configuration is just fantastic. The ability to produce is increased more with the number of tracks you have.

JOE BANANA: Imagine what can be done with a track for each mike.

HAYES: It gets tricky. You also have to consider when the opportune time to expand is. When should you take that big jump. And, then you wonder, well if I buy 16 tracks, are they going to become obsolete too. Its like any business, there are problems. You have to be a pessimistic optimist . . . which is a realist.

And, then there are a lot of people who say, "Well if you go into a large recording studio what if somebody down the street puts up another studio." I've never been concerned about competition. I know that I have to sell whether I'm in the recording business or whatever. I have to sell our ability to do it, and do it well. So, it really doesn't concern me that there are five more studios around.

ED: Does a small studio have any advantages over its competitors; the large 16 track studios? Is there something that you can do to optimize your advantage?

JOE BANANA: Its not so much the level of equipment, but what you, the engineer, can do with "X" amount of equipment. Like one person operating 8 tracks might be able to produce more than another person operating 16 tracks.

McARTHUR: Sometimes the experience of having worked in a small studio, knowing how to cram everything onto the available tracks can be a great help when you get into a multi track situation.

HAYES: I guess lower rates is an advantage. But I don't see how you can say that something less than the best can be an advantage. In a small studio you have, perhaps, a better relationship, if we are talking in the music area.

BANANA: Recording music . . . as opposed to broadcasting production, jingles, audiovisual . . . that kind of thing!

HAYES: Yes, in the music area where you work closer with them (the group). I think there may be less pressure; a more relaxed feeling than in a larger studio.

ED: What about broadcasting production? Does the smaller number of tracks place a limitation on what you can do?

BURMAN: Not unless you're into music. For audiovisual, audio-radio spots and for that I think by all means four tracks are more than enough . . . unless you get somebody who wants six different sound effects going at the same time, and a music track, and a voice track.

HAYES: Those are the times it comes down to a razor blade (splicing) or mixing with the cart machines or auxilliary tape machines. I don't see in broadcasting production that you need a lot of tracks.

McARTHUR: For broadcasting production a small studio does have an advantage over a large studio because of cost.

BURMAN: I don't know. Some of the large studios do charge by the track, and chances are they would charge less for four tracks.

HAYES: Some of the larger studios don't like to mess around with commercial recording. I know of at least one in New York where every time the engineer was told he had a commercial it just put him into a panic. He just can't relate to the type of people who are in that type of business; the producers. There is a tremendous difference. The commercial producer doesn't have time. He wants it done, and he wants it done now . . . so you don't have time for the perfection. If you're going for a musical production or an album you're obviously going to take a lot more time. Generally you have a bigger budget, too.

ED: The idea of a truely professional co-op studio was vigorously discussed as a not-too-possible alternative to the partial expansion of each of the smaller studios in a particular area relatively remote from one of the large recording centers.

HAYES: It seems to me that if you've got three or four people who are going in together to have a sixteen or a twenty-four track studio that in order to make the thing work you've got to have tremendously compatible people. Their heads have got to all be in the same direction. At the same time, one should be proficient in one area and another in another area, and so forth. If you all have the same philosophies then I think that it would work, but otherwise I don't. If they are all doing the same thing then they would be competing with one another.

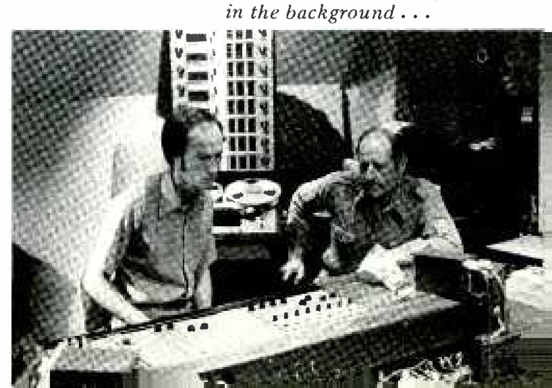
McARTHUR: The only way that I could see that it would work would be if one person were specifically interested in jingles, another just in audio-visual, and another in music production . . . or some combination like that. Like they divided it up like that. But, I have to admit it might be about the only way an area like this could ever afford a super-super studio.

HAYES: You'll also find that one thing leads to another and in a situation like that you are really restricted in how far you can go. You have to stay in your own area because someone else is handling another area. My feeling is that if I want to come into my studio at 9 at night, or 5 in the morning, I can do it and I don't have to ask anybody. Particularly with a small studio you can't really schedule things that well. If something comes up and you've got a chance to make a buck, or if you've got an idea you want to develop, you don't have to fight over a bunch of people who are in the studio. I don't think that that would be practical from my point of view.

ED: A discussion of how to finance expansion emphasized a general lack of any understanding among the local banking community about the business potential of recording studio operations:

HAYES: The way that I got into the recording business was through radio. I was working in radio and I couldn't make enough money to live. I was asked to do some free-lance spots, and then more and more of them, and there weren't any studios that I could run into and have the spots recorded. So, there was a small mono studio in St. Petersburg that had been locked-up by the Sheriff and the bank loaned me the money to buy the studio, which we moved out of the building and into my home. And then, I just expanded the studio as was necessary as I found more and more business. As it turned out I am eternally grateful to the radio industry for not paying me enough to live on. I was connected mainly with small stations with the exception of WSUN, which was at that time municipally owned. I wanted to stay in St. Petersburg, but I just couldn't make enough money so I went into the recording thing. I was forced into it because it was the only thing that I knew how to do. So far its worked out great. However, and this is the funny part of the story, in order to expand the studio from a one track-mono studio into four tracks I went to the bank. First, I talked to the vice-president of the commercial part of a large bank, and he thought that it was a good idea. The presentation that I made to him indicated to him that I'd thought the thing out pretty well, and he was in favor of making the loan. But, he wanted me to talk to the president of the bank before he stuck his neck out. So, we went down there and the president of the bank was rather conservative, and he felt that it might not succeed, and therefore he became real cautious. He asked me where he would get rid of the equipment if I went broke. I made some suggestions to him,

Jim Cottle (left) and Paul Hayes (right) at the Tascam, with the Scully 8-track in the background . . .



. . . additonal view of Hayes Productions' equipment racks

but they finally turned me down anyway. So I just gave up the whole idea. And again, they did me a favor because I wouldn't have been happy if I'd ended up with only a four track studio. Now here's what seems ludicrous, a janitorial service business came up for grabs, and I called the bank and talked to the banker again. He said that there wouldn't be any problem borrowing the money to buy the janitorial service. So, I just went down to the bank and signed the note and bought the janitorial business easy as that. So, my feeling has been that a number of bankers can relate to a mop and bucket they see when they go out at night, and when they come in in the morning, with someone pushing it. So, they know the function of the mop and they know that it goes into the bucket, it gets wrung out and eventually gets put on the floor and is supposed to clean it up . . . and that somebody will pay to have that done . . . but, recording?

I've never been able to get over that.

McARTHUR: I ran into the same thing. We financed a pet store with all kinds of fuzzy creatures. Now, what do you suppose a bank would do with five or ten dogs? But, when I went back to the same bank with a much smaller figure for a recording studio I was turned down.

HAYES: I think that the situation has changed now, to some degree. You have to establish good credit, like you do in any business. Today I've been told that I could get the money. However, I've got to limit myself and I've got to really think this thing about expansion through, because a loan is only as good as the ability to pay it back. My only concern at the moment is that a studio really has to establish itself, and if you're working the money pretty close, then you really have to have a running start. So, if you get a loan for X amount of dollars with the first payment due, with the interest added on, say eight or ten months down the road that's good, and you have a running chance. But, when you have to start paying it back on the first of the next month, that's tough, even if you have an existing studio. Even if you have eight tracks and you go to 16, I still think it would be a whole new ball game. You still need a little time to get rolling.

McARTHUR: You may need a completely different clientele. Your present customers may not be able to afford those additional tracks.

ED: Coupled with the difficulty of financing a local studio operation is the priority given to acquisition of equipment when money is available. With only X number of dollars where do you start. . . . where do you skimp? There was unanimous agreement that there was no place where skimping wouldn't show up.

COTTLE: I think it's important for the studio to progress, as I call it, unilaterally. I don't think it's worth your while to buy anything that you're going to throw away. Good pro equipment, besides going completely caput after a while, or becoming completely obsolete, just does not go out of style. I think it's important to buy good things from the start . . . at about the same level. I've seen a lot of small studios that have a tremendous monitor system and very cheap mikes. And, they try to justify the fact that they have cheap mikes by the fact they have the great monitor system. It just won't work, they aren't compatible.

The weakest points, it seems to me, are in the transducer areas; the mikes and the loudspeakers. That's why, I guess, you have to have so many mikes in a studio. .

. . because with one or another of a kind of mike you can be tasteful. They are so unscientific in their nature; they all have their little idiosyncracies, whereas, within certain broad limits a console is a console. I know there are people who will argue that the type of distortion from an API console is different from the type of a Neve, but I think that at this level that is a pretty infinite difference.

BANANA: The whole electronic system will be as weak as it's weakest link. It goes back to compatible equipment, and I can't see skimping in any areas.

HAYES: Being basically broadcast production oriented and now getting into music we just went through this when Jim (Cottle) came with us. Jim was telling us we had to know in the control room what the full spectrum of sound is. He kept pushing for a re-evaluation of our monitors. I'd been working with Sentry I and IA's since I got in broadcasting, and I was used to them. I'd lived with them for all these years. And being in the broadcast production end of recording I was not concerned as I should have been with all the little highs and lows necessary in music production . . . which I think points out a basic difference between the two kinds of work. We tried a lot of different speakers. Anyway, Jim finally talked me into getting these Sentry III's. The morning after they were put in Jim asked me what I thought about them. The more I listened, the greater the difference I began to hear, the more I was impressed. And I am sure that what we are able to hear now is reflected in the product we turn out. So, I think one of the most important things that you need is the best monitoring system you can afford . . . and a well balanced room. I think even before you get into equipment you have to start with a good location. A good building of ridgid construction. It makes a tremendous difference.

ED: The discussion of studio acoustics, dispirited as it seemed to be, indicated the phantom nature of the subject in the small operator's view:

COTTLE: I'd treat the acoustics with moderation. Because of the kind of trial and error approach you'd use until you were satisfied, the cheapest material you can find might be best. Generally, I guess, they say you should try to keep the reverberation down, and keep it constant over the frequency spectrum.

BANANA: I don't think there is one perfect set-up. It really has to be tailor made. If all else fails use egg cartons!

McARTHUR: Are you saying that if a studio isn't familiar with acoustics it's better to just work with a dead room.

COTTLE: I don't think so. Trying to use a dead room can get you into just as much trouble as a real live one. We have tried to approximate a living-room effect, at least in the control room. Most people are going to listen to music in their living room . . . so you might as well mix it that way. It's hard to do in a control room because you don't have all the reflectors and absorbers that you have in a normal living-room.

BURMAN: What influences the acoustics most in a small studio is the speakers. I've got a suggestion for anyone in a small studio. A small speaker with a baffle. It will reproduce the sound the way you will usually hear it. Those Sentry III's lure you. But, to really hear how the *rubber hits the road*, as they say, it's that little speaker. I just went out and got a speaker for a car radio and put it in a little box, because somebody is going to be listening to the sound on just something like that. . .

ED: And on to more substantive ground:

McARTHUR: What about noise reduction versus a better tape machine?

COTTLE: For what you spend for a Tascam, say, and a noise reduction unit, the combined amount just won't get you a tape machine that doesn't need noise reduction. If you got a DBX that you use with either then you certainly wouldn't be regressing in any manner.

HAYES: Mel do you use noise reduction?

BURMAN: No, I just put the signal into the Scully and it comes out nice and clean.

McARTHUR: Mel, you had that choice. Why did you choose the better machines over noise reduction?

BURMAN: Because, I wanted to get one of the best decks I could and the signal/noise on that one is unbelieveable.

McARTHUR: Does the linking up of your Scully with a Tascam board go back to the idea of keeping everything on an even keel, equipmentwise? Compatible?

BURMAN: I can see why a lot of people buy them (Tescam Consoles) Because they get you started, especially when you don't have much to spend . . . you can mix things with them, there are lots of channels . . . and for all they lack, which you find out about later . . . you can find ways to sort of work around the problem. So, while it's true that, at this stage, the board is my weakest link, it did get me started. If I had to do it over again, I'd try to get a better mixing console with more headroom, and built a little better.

COTTLE: I can see where having a better tape machine sort of shifts your sights from the noise produced by the machine so that you can worry more about the noise produced by the console. It isolates the problem in one less area.

BANANA: Use of noise reduction is a must, it takes care of it (the headroom problem) for you. I had Dolby when I had four tracks. Then I got DBX for my eight. I had liked the flexibility of the Dolby more than the DBX until I had used it some. DBX is just "on" and "off", and that's it. It's been trouble free, and I am super-satisfied with it. It's just like it's not even there.

COTTLE: I have a thing about noise reduction. A lot of people say they can hear it. Take the DBX on some tracks I heard in Maine, on eight track. The compression seemed to be off. It might have been a defective unit, but I'm not convinced it was.

BANANA: I am surprised to hear you say that, on mine there's instant recovery. I don't know how its so fast. Could the problem have been that it was a DBX recorded tape being played back without decoding the tape?

COTTLE: It *was* a previously recorded tape. The problem could have occurrred right there. As far as reducing the noise it did a fine job of that, but it seemed to me that the ratios of compression were off, not matching up.

Anyway another way we control noise is using compression. That is, you limit the dynamic range which enables you to put the signal at a more optimum level over a longer period of time. In other words, your average level increases and your apparent volume is louder. I'm talking about doing this at the input stage. If you compress something once it is on the tape then I don't see how you are going to do anything but add noise. If you're compressing something that is already on tape when the signal stops it is just going to bring up tape hiss.

McARTHUR: On this kind of equipment I've really noticed that there is a lot of added noise just from having an input switched on. It seems pretty important to go through and make sure all of the inputs that aren't being used are completely switched out of the system.

COTTLE: Manual noise gating. In other words, when a track is finished turn it off.

BURMAN: I'll tell you one noise reduction I have installed in my studio. I have switched over completely to Scotch 250. It is fantastic tape.

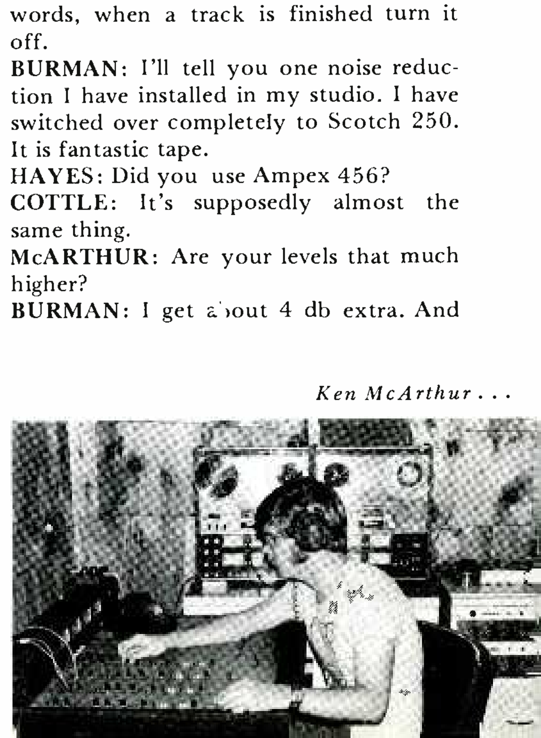
HAYES: Did you use Ampex 456?

COTTLE: It's supposedly almost the same thing.

McARTHUR: Are your levels that much higher?

BURMAN: I get about 4 db extra. And

Ken McArthur . . .



it completely eliminates any background noise. You would have to really turn it up high to hear anything. The signal to noise is unbelievable.
BANANA: It has a lot of headroom, And, there's another Ampex tape that's supposed to have more headroom than this.
HAYES: Grandmaster.
BANANA: The people at Criteria (Miami) tell me that it's got more headroom, about 6 dB, but the Scotch 250 will not print through.
BURMAN: I have no experience with the Ampex, but comparing the 250 to the 206, I had a lot of problems with sibilence, high-end distortion, and the 250 has completely eliminated it. I readjusted the bias on my own machines, full tilt, and the thing is just round as can be. I really like it. The sound I get, as well as the extra gain.
COTTLE: If your going to adjust for minus 6 dB then you are closer to the satuation point so your levels have got to be right on. There is no room for pegging, or any −2 or +2−1/2 without going to any distortion. You can keep a voice peaking at zero much easier than you can keep music. We have ours adjusted to a minus 3, which gives us 3 dB headroom. There's no substitute for keeping your eyes on the levels.
McARTHUR: Of Course, especially with semi-pro gear, you have to be very careful about overloading each stage.

What about patchbays for semi-pro gear like Tascam? Is the added convenience worth the added noise?
BURMAN: I highly recommend it.
McARTHUR: Do you pick up added noise doing innerstage patching?
COTTLE: No, but it is important to keep the lines short on innerstage patching, because it is all high impedence. Six or seven foot maximum, both ways combined; 3 feet one way, and 3 feet back. It's sheilded, but after you go beyond 6 feet in high impedance you begin to lose some of the highs, because of the capacity of the lines. The patchbay should be right next to the console.
BURMAN: You could do what I have done. I just went all low Z, and that helps a lot. We used transformers to switch the impedance, and ran all the lines that way. We also use low impedance mikes connected to the board through hi to low Z transformers.
COTTLE: I was speaking mainly about the innerstage patching. Would you convert that to low impedance?
BURMAN: Yes.
COTTLE: That's the way to go, then. That way you can run to St. Pete and back if you want to.

ED: The discussion of patching led naturally enough to the group's thoughts on the relative needs of outboard equipment:

Recnac Recording's peripherals rack . . .

BURMAN: Even though I have a board with equalizers I have an external equalizer, a Complimiter and, of course, a spring echo.
COTTLE: My favorite, I would way is compression, but not over a little reverb; a small spring unit . . . you've got to have that first.

The only problems I have found in spring units have come from overuse. You cannot push the use of a spring unit because it has its own characteristic sound. You have to be extra careful putting a drum through it. If you're not, all you'll get is underwater.
BANANA: I had absolutely no problem using a spring unit. I'm using a Clover unit. It's just about on the same level as the Tascam, which is the kind of board I'm using. They are just really compatible.

Circle No. 133

Once and a while I put a little touch on the snare drum, if somebody wants it, but its got to be very subtle or you'll get *sonar*, as Jim said. It's too easy to over use something just to make it obvious.

HAYES: We ordered a Mic-Mix reverb unit and they asked us what it was going to be used with, and then they wired it for use with a Tascam. Actually, the way I remember it, we called them and told them we would like to try it and if we liked it we would buy it. So, they said fine, send us a check, and we'll set it up on a trial. The way they handled it, they were very good to work with.

COTTLE: You can build around Hammond springs with electronics, the integrated circuits available now, that are fairly easy to work with. However, I want to *use* a studio; *not build* one. But, I guess, you do have to do a certain amount of that in a small studio.

McARTHUR: What after reverb?

COTTLE: Compression. There is nothing more frustrating than being into a really fine vocal, and to have someone knock your socks off with the next note. I use compression a lot. Changing the release time can give you some interesting effects. It can make the vocal sound more intimate. By using a moderate release time you are in effect bringing up the end of a phrase. I usually also use compression on bass guitar.

McARTHUR: It would also be good on something like an acoustic guitar used with a slide. It's very difficult to get something *twangy* sounding like that to sound natural. There's a very fine line between not being able to hear it loudly enough and having it sound too close. Compression can limit the dynamic range and make it easier to deal with.

COTTLE: We have these Spectra Sonics compressors, and the reason we have them is not only that they are super good compressors but that they have the most variable parameters in a compressor that I could find at a low price; adjustable release time and adjustable slope. Most of the compressors on the market are not true compressors, they are limiters. This Spectra Sonics does a certain amount of limiting too. We can't control it, it is done internally on peaks, but it works more than just on the peaks. That's why I chose it. Now there are some smaller limiter models on the market. In fact, I think the DBX 161 is very good.

The crux of it is that in a small studio you are so much more concerned with getting more than you really can. The danger is that you can end up compressing or limiting everything; just squashing it which can totally ruin the product. There isn't much question . . . you can certainly get more apparent sound out of heavy limiting, and if you do it right you won't notice it as much. If somebody puts the thing on a large system and listens to it then you're probably going to hear the limiting. You have to be very tasteful with anything you do, you can't overuse anything. But, generally limiting will give you more sound. If everything is in the right place it's easier to control.

ED: Concerning the broadcast interface elicited these comments:

COTTLE: Test tapes are probably the first thing you have to have, relative to tones.

BURMAN: Then if only you could figure out what the TV stations are going to do with your audio after you send it over to them. I don't know if you guys do this, but we equalize for each station we send a tape to, because they do crazy things with your audio. Like Channel 13 will boost the hell out of the highs. And stuff we send over there sounds like it's coming out of a phone unless we equalize it. Haven't you had that problem?

HAYES: A lot of our stuff goes out of the area so, we don't hear it here. I know what you are talking about though, there never seemed to be any standard at any radio station I was around.

McARTHUR: You do a lot of audio visual work. What do you need as far as special equipment for that kind of work?

BURMAN: Essentially what any decent recording studio needs. We have quite a few records of background music and

Mel Burman's Professional Broadcast Productions set-up . . .

Mel Burman . . .

sound effects. That's about it plus, good microphones.

ED: Microphones, at last!

BURMAN: One thing that I've found important in recording voice tracks on location for an audio visual track is a really good location mike. We have used the Sennheiser MD-421-U-4 with a great deal of success. It is really important to use a good mike otherwise it just sounds like amatuer hour. Even if we use a cassette deck we'll use it with a studio mike. It really makes a difference.

COTTLE: On location a good mike with a rise in the presence region from 6 to 9 kiloHertz is really helpful. You really can't get close to the group, and it gives you that illusion of being closer to the mike. On the condensers they make a special capsule with that incorporated into them. The AKG C451 has several different capsules that you can screw onto the front. One of them is flat, one has a high frequency boost, etc.

Mikes are as variable as loudspeakers. They all have their own sounds.

BANANA: You have to suit your own ears and experience. Just by using different mikes you can change the sound of everything you're doing.

McARTHUR: Yes! A good case in point was that we recently recorded an acoustic guitar with an AKG D202E and then turned right around in the same studio, with the same guitar and guitarist, playing the same material and miked it with an AKG D160. The D160 gave us a much brighter sound than the D202.

COTTLE: Or take a Sennheiser MD-421. It's a completely different world. The 421 has a noticeable peak at 5 kHz, so its brighter. The Sennheiser has become a favorite for miking a snare because of that feature. You can't just look at frequency response either, because it's not just the frequency response that you're totally concerned with. It's the axis, the position of the mike, as well. In the small inexact studio you depend so much on your mikes for your sound.

BANANA: You can't generalize. But I wouldn't recommend that anyone buy a cheap mike unless you want one for a special effect.

ED: And then on track placement, and mixing . . .

McARTHUR: Joe, you've done a lot of work with demos and such. How do you go about deciding what goes on what track, when you only have four tracks.

BANANA: When I was doing 4 tracks every situation was unique. There were basic things to do just to get the rhythm tracks down. I would always sit down with the musicians and ask what they had in mind, and then lay it out the way we were going to do it. If we could do it the way they wanted it, then fine, and if not we modified it and got everybody ready with it. Before they actually started recording I would brief them on what to expect, and how things were going to be coming. Then it would fill out a lot easier. Rather, that is, than coming in and playing it by ear for the first time.

I usually leave the solo or lead tracks off the original tracks which would be drums, bass, maybe keyboards, and guitar.

McARTHUR: Did you put all of those on one track?

BANANA: No. I record that as a first take, those basic instruments together on separate tracks.

COTTLE: What we used to do sometimes was take the three tracks and mix it down to the fourth, or take four tracks and mix it down to another machine. The reason we could do that was because we had the Scully 280B and they were very low noise.

BANANA: If you have to pre-mix instruments on the same track, I would do your bass and drums on one track, and your keyboard-like instruments and rhythm guitar on a separate track, and maybe your lead guitar and harmonica on the third track.

HAYES: How many mikes do you put on your drums?

BANANA: Usually 5 or 6. I have a set of very old Gretch drums. Pretty funky but very good. If somebody comes in with a lot of toms I'd use six mikes. I'd like to use 9.

HAYES: And a track for each of them?

ED: Well that only leaves one question to be answered . . . How does a small studio make any money?

McARTHUR: How *does* a small studio make any money?

BANANA: How *does* a small studio make any money?

COTTLE: How *does* a small studio make any money?

BURMAN: How *does* a small studio make any money?

HAYES: How *does* a small studio make any money?

Why you can"t wait for:

DIGITAL DELAY

DIGITAL DELAY
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DIGITAL DELAY
DIGITAL DELAY
DIGITAL DELAY

by Ken Schaffer

Though it's only been around for four or five years, digital delay has become one of the staple tools of the modern full-deck recording studio. What's more, delay lines have become common to the signal chains of hundreds of theater, stadium and church PAs and to the road mixing consoles of over 100 major touring bands.

Wherever set up, delay lines have created production possibilities which couldn't even have been considered before. In live concert, digital delay, probably more than any other technique, has helped close the embarrassing gap between lush and patiently produced records and their mortally-thin *live* renditions.

Within the past few months and the introduction of technically-advanced delay lines in which stacks and stacks of shift registers have been replaced by more versitile Random Access Memories, (RAM) time delay has become the progenitor of new capabilities for which delay is only the *means* to generating signal modifications that allow radical new possibilities and time-saving practices, some of which will bring demonic smiles to the expressions of even the most jaded engineers and producers.

It is the intent of this article to discuss some of the new possibilities opened up by the *almost-continuously-variable* RAM based digital delay line and to explore the largely uncharted worlds of digital pitch-changing, automatic harmony generation and frequency restoration of sped-up and slowed-down tapes.

The units used as models for the ensuing discussion are two manufactured by Eventide Clockworks, the 1745M digital delay line and the H910 Harmonizer, both of which have been available for only the past few months. At this time, no comparable units are available elsewhere. But the promises implied by these revisited DDLs (digital delay lines) and their arsenal of effects as everyday mixdown tools make it inevitable that units of this type will soon be as standard to the studio as noise reduction, razor blades and gaffers' tape.

We'll also be breezing through the DDL in all its common uses. All in all, some of the technology is so new that even more uses should be arising regularly over the next few months, as units of this type are installed and put into the hands of more and more imaginitive engineers.

PLAYING WITH TIME

Before the advent of digital techniques, the only way to directly delay an audio signal by a second would be to put it through 186,000 miles of wire. Such a horrendous coil would involve losses, induction and troubleshooting nightmares beyond mortal conception.

Tape delay is, or course, more flexible than wire. Assuming a mere inch between the record and playback heads, a 30 i.p.s. base speed and a VSO, the delay created may be gotten down to as little as 30 milliseconds; a most useful figure near which some of the delay lines most exquisite effects occur . . . indeed, it's likely that such a technique was used for the ADT effects predominant on the Beatles' Rubber Soul LP . . . but the bulk, mechanical wear and ultimate lack of flexibility of such an arrangement became unforgiveable at the introduction of the first digital delay line.

Delay lines may, to the user, be considered transparent boxes, the actual workings of which are pretty ignorable except for those specific points which are actually interesting to know about.

You put audio in, and get unblemished audio out, albeit a short time (most often 10 to 100 ms later). First generation delay lines consumed a lot of power to make their shift registers register and suffered a consequent lack of reliability. Traditionally, units designed for recording applications allowed delay to be varied in steps ranging, depending upon the units' manufacture, from one to five milliseconds. Because of their design, varying delay while program material transited the delay line created gaps (also called artifacts, glitches and transients) which clicked their way onto the track; varying the delay time during a mix was, therefore inadvisable.

The latest breed of delay line, such as Eventide's 1745M, is *almost-continuously-variable* (delay steps in this particular unit are 20 microseconds and are effectively imperceptible) and allow for the delay time to be varied freely, even during soft passages.

In fact, this capability of changing a

THE AUTHOR

Ken Schaffer is president of the New York based Ken Schaffer Group, he being a one-man company and he being the seller of, probably, more digital delay systems than any other individual.

Ken's business is confined to digital delay lines and other particularly special "special effects." He's also kept busy consulting on projects requiring esoteric black boxes, for the studio and for touring bands.

Starting as a recording engineer (the Left Banke, of "Walk Away Renee" fame) Ken made a quick departure from that particular field when, as a novice captain of the console, he forgot to sel-sync a track of strings, excused himself on the premise that he had to go to the bathroom, and didn't reappear in a studio until a couple of years later.

His first long-running business was a specialized PR agency that handled such clients as Jimi Hendrix, Alice Cooper, Janis Ian, John McLaughlin, The Lenny Bruce Estate and, alas, The Comet Kohoutek. He remains active in this on a "one client a year" basis.

His special effects business keeps him most occupied. But he's also a partner in Douglas Records, a small record company with a good reputation, now distributed through Casablanca Records.

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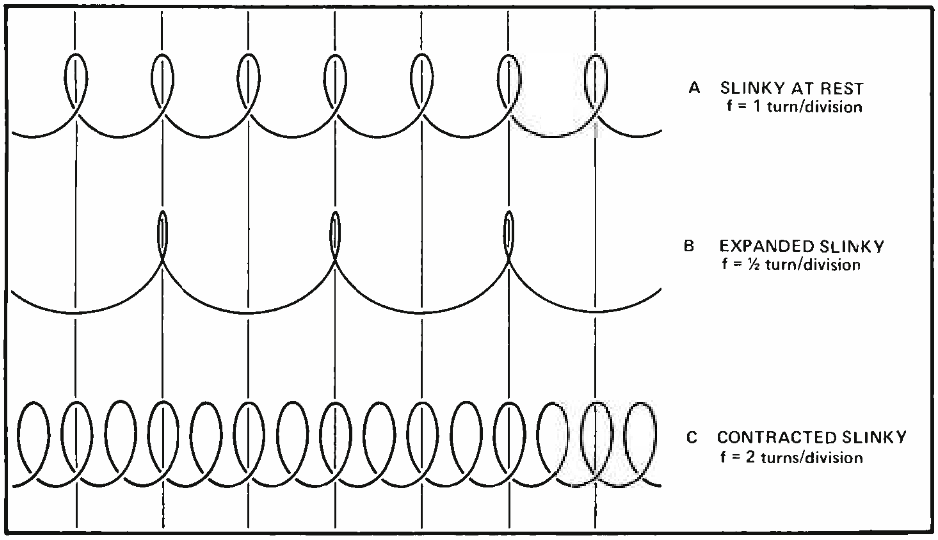


DOPPLER EFFECT ANALOGY

delay setting while a signal transits the unit sets the base for a whole platter-full of new effects; the most startling derive from an effect known as Doppler Shift.

Doppler shift causes pitch changes; through Doppler pitch change all harmonic ratios, and thus musical values, of the input signal are retained intact. Doppler pitch shifted signals are therefore perfectly at home with unshifted signals.

Anyway, delays of a second are way too long to create any of the more interesting and useable mixing effects time delay makes possible. But even one millisecond, by the wire route, would tie up 186 miles. As expensive as digital delay is sometimes asserted to be, it surely is, millisecond for millisecond, cheaper than this much wire. It's like Lenny Bruce once said: "There's only *one* telephone company:" in this town, it's digital delay.

Doppler explains the phenomena of the change in frequency of a cars' horn as it approaches you rapidly and then pulls away. As it approaches, the wavelength of its fundamental frequency is contracted in the space between the car and the stationary listener. Its frequency, whether merely listened to or actually measured, rises. Only at the moment the persistent horn is actually abeam the listener is its audible frequency the same as its stationary fundamental. As the car speeds away, it *pulls* its waveform along with it; graphically and audibly, this waveform must *stretch* to fill the increasing space, and the effect is a lowered pitch. Obviously, Doppler Shifts are velocity effects . . . they depend on changing time-space parameters. Have a friend speed head on toward your opposite thundering *short*; the freeways' music will make this physical theory most understandable. It also works on distant trains.

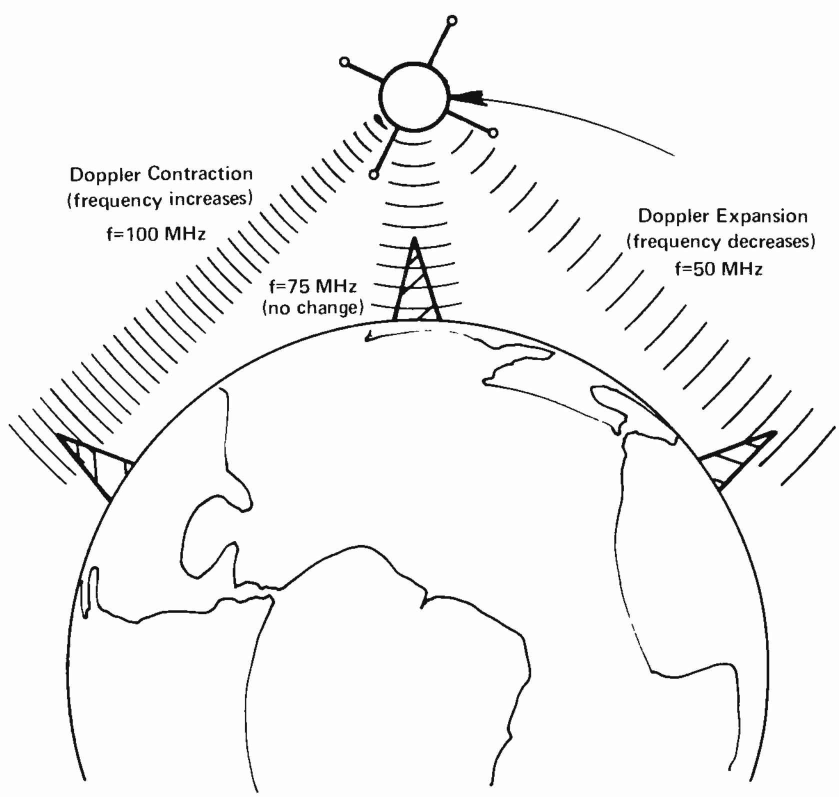
Because Doppler Shift is so much fun to explain, we'll point yet another entertaining picture of it. An even more graphic illustration of the Doppler Effect would be to take a Slinky toy of 100 turns. A 100 turns per foot Slinky could be transformed, for the sake of analogy, into a 100 cycle per second audio signal. With said Slinky up against your ear, pushing it, so as to contract the Slinky's 100 turns into half the space, creates a Slinky with a per-cycle wavelength of one-half, or a relative frequency of 200 turns/cycles per foot/second. Pulling the Slinky to a 2 foot length simulates a car pulling away so rapidly as to cut its horns' measurable frequency in half. This *is* rapid!

Increasing or decreasing any frequency by a factor of two is, of course, equivilent to a change of one octave up (twice the F) or down (half the F).

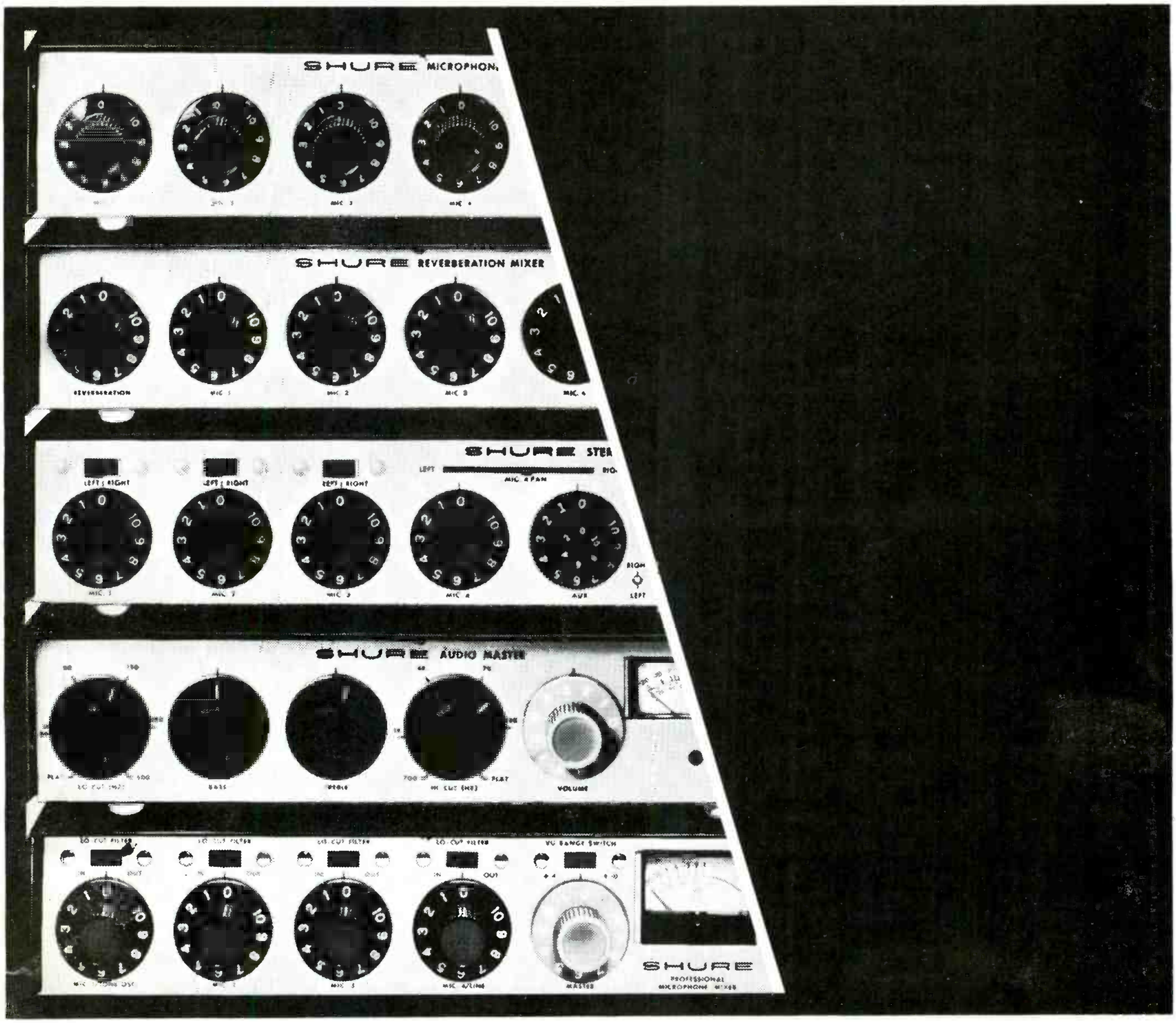
The continuously variable delay line allows for similar chicanary to be played with a speeding audio signal. The nuts and bolts method of generating such a pitch change would be to shock mount a high-grade speaker on a speeding vehicle on the salt flats. The delay line method is more tolerable.

To achieve Doppler pitch changes we must force the delay to increase or decrease *while an audio signal transits through the delay line.* If we cause the delay to *increase* while the signal passes, the same number of cycles must stretch to fill newly created time; this causes a downward pitch change. By setting the delay to decrease as the unsuspecting signal passes, we get a higher pitch at the output. The actual amount of pitch change is determined by the rate of delay change in milliseconds-change-per-second, and is entirely controllable over an indefinite range.

Before getting to the myraid applications of such a phenomena, we must establish that there are limits to the amount of pitch change we can unashamedly make use of. The Eventide Harmonizer allows for pitch changes of up to an octave either way (pitch ratio of .5 through unity through 2.00; pitch ratio of 1.00 is yaya in, yaya out, with no



DOPPLER SHIFT ALSO AFFECTS FREQUENCIES OF SATELLITE RECEPTION. (Greatly exaggerated here.)



Rack 'em up.

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pitch change).

The pitch change is created, we now understand, by a changing delay time. If we find that a rate-of-change of, say, 50 milliseconds per second gives us the desired pitch change, and we start the process at zero delay, then after one second our pitch changed signal will return 50 ms *later* than the signal we put in for processing. After four seconds the OC'd (pitch changed) return will be 200 milliseconds out of synch. Such deviations may be cute at first, but become musically unacceptable as the lag gets to be of such magnitude.

The solution, then, is to make a delay line that has a maximum delay which is still below the threshold at which the ear/brain combo yell "Out of Synch!" By designing a DDL so that the cycling varies from zero to, say 25 milliseconds and no more, then our pitch changed return will never be audibly out of synch with the input signal. The DDL is designed to roughly parallel a tape loop, digitally. When the delay cycling reaches its 25 ms terminus the delay drops back to zero and the cycling continues once again from there.

This drop-back-to-zero does create an instantaneous gap, known as a *data seam*, or glitch. Since the amount of pitch change is determined by the rapidity at which we cycle up or down, more glitches appear when more radical pitch changes are set. At relatively small, musically useful pitch changes (a third octave, typically) the glitches don't become boistrous enough to stretch our tolerance. Even at extremely radical pitch changes the PC'd signal can, depending upon program characteristics, still be usable, especially when the pitch changed signal is to be mixed in with other program material.

NOW THAT WE CONTROL TIME, WHAT DO WE DO WITH IT?

THE ADVENT OF AUTOMATIC HARMONY

The cycling digital loop of the almost-continuously-variable delay line thus allows for pitch changes of adjustable magnitude, up to about an octave up or down. In that musical values are held intact in the pitch changed signal, several applications of this new capability become apparent.

One of the most intriguing fulfills a mixing dream that probably began when the first generation of digital delay lines made possible ADT and automatic unison: how about automatic harmony?

Because this latest generation of delay lines is so new, automatic harmony is not an effect that's been employed in every studio, on every record. Contrariwise, only a few studios have the capability. But this is surely temporary, because one specific unit that allows for the generation of automatic harmony (the Eventide Harmonizer) can be put in place for well under $2000.

Between a pitch ratio of unity and a pitch ratio of .5 or 2.00 (an octave down or an octave up) a continuously variable pitch changer passes through every harmonic interval that might be selected as a harmony part by a harmonizing vocalist. A third up, a fifth down; a fifth up, etc. Each interval coresponds to a specific pitch change ratio. By setting the delay line/Harmonizer to that ratio (it can be tuned by ear or set using a chart revealing the relationships between harmony parts and pitch ratios) the output of the DDL, when mixed in with the unmodified original signal will establish two part harmony. The output of two Harmonizers will establish three part harmony.

If however, the pitch change ratios are not altered at the time musical chord changes occur on the accompanying track, we create a rather unnatural harmony part, in that the harmony interval stays the same, establishing "parallel harmony," and this harmony relationship would not likely be the choice of a harmonizing singer or player.

All that need be done to counter this problem is to change the pitch ratios when appropriate; this can be done by controlling the Harmonizer with an external keyboard or synthesizer audio output capable of supplying the Harmonizer a reasonably pure sine wave, as a control signal.

In either case, playing the appropriate note instantly establishes the appropirate interval, and this interval can be changed in exactly the fashion as would an accompanying player. (The Harmonizer can also be voltage controlled; like a synthesizer, the choice of a keyboard as a controlling interface is merely an expedient – the choice could as easily be *(have been)* a few toggle switches set at often used intervals, a potentiometer, etc.).

Thus it is possible for a solo vocalist to accompany himself on a keyboard and be heard in harmony. And, because of the cycling digital loop that allows for the fundamental pitch change, each *"work"* would come in at a slightly different delay time (somewhere between 0 and 25 ms) and suggest something even better than precise electronic modefication.

Bad Notes?

In the mixing process, such a pitch changer can be punched in for correction of single notes which are flat or sharp. It's merely necessary to run through once, establish the correcting pitch ratio, and punch through.

Frequency Restoration To Time Altered Tapes

What may one day become the most important use of this type of pitch change capability is the ability to restore the original pitch to a tape which has been sped up or slowed down.

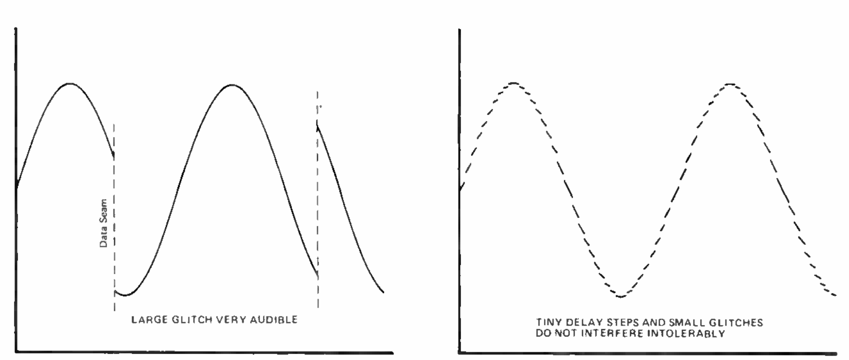
Assume a 60 second spot has been beautifully recorded, but clocks in at 63 seconds, *(as is always the case.)* A VSO controlling the tape transport can restore the time to the mandatory 60 seconds, but at the expense of the pitch which, in this case would take on a Mickey Mouse tonality.

No problem. The tape speed is now altered to the necessary time; a digital pitch changer is then put into the signal chain and is set to restore the program to precisely its original pitch.

An interesting option on the 1745M is one which synchronizes pitch restoration and tape transport drive. Eventide's pitch change module has three outputs designed to drive the most popular tape transports (60 Hz., 9.6 kHz, 19.2 kHz), thus doing away with the external VSO entirely, and making pitch restoration a one-handed operation. The tape is merely adjusted to run through in allotted time, and the pitch restoration is automatically synchronized!

Yet Another New Effect: TUNNELLING

By adding feedback to a pitch changing DDL, yet another new effect, called *tunnelling* is introduced. Tunnelling is derived from the fact that if we establish a pitch ratio of, say, 1.5, between the



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pitch changer's input and output, then 1000 Hz note entering the pitch changer will emerge as 1500 Hz. By feeding this output back to the input, it will reappear the second time as 2250 Hz. . . . and then 3375, 5062, 7593, etc., until it is inaudible because of either frequency or mud. The same tunnelling may be set up in a downward direction (1000 Hz in, 666, 444, etc.).

Depending on the program material, the tunnelling effect can be either downright dangerous or charmingly mystical. Upward tunnelling works well with guitars and, in moderation, bass guitars. Small increments of downward tunnelling (i.e. gentle pitch ratios, a la .95) can make drums sound backwards. Nearly any amount of tunnelling is prospectively the mark of the whole next generation of science fiction soundtracks.

Introducing feedback to a non-cycling, or stationary, DDL produces other effects, depending upon the delay time set in. At short delay times, flanging may be created, or percussive signals may be transformed into actual musical notes which sound something like effects achievable through synthesizers: Longer delay times create a distinct EMT--like reverb, while delay settings of 100 ms or more produce actual repeats and, with higher feedback levels, multiple repeats.

Traditional Delay

The new effects made possible by cycling delay certainly haven't had anything to do with the fact that the digital line is now an everyday studio tool. Because delay lines, in any form, are such rottenly expensive gadgets, they must — — even without pitch changing — — — do something quite dramatic, indeed, in order that such units be part of the mixdown apparatus of so many studios and the onstage apparatus of so many bands struggling to recreate, live, the vocal quality of records that took four nights till dawn to mix.

And so it is true. Effects derived from delay lines range from the *"invisible"* to the obvious and raucous. Whereas phasers and flangers strongly earn the right to be considered *"special effects"* boxes with *"sounds"* of their own, delay lines are most often used to create effects which are invisible to the listener.

The most common use, of course, is to create Automatic Double Tracking, or automatic unison. This application preys upon the fact that the brain cannot distinguish two identical signals unless there be an appreciable time differential between them. We'll call the point at which the brain *can* differentiate between the two signals "the critical point". Below that point is the region of ADT and signal *"thickening:"* Above that point, the region of discernable echo.

If we visualize that the delay lines' output, if even a millisecond removed from the input is, in reality, an *echo*, then ADT is an audible illusion suggested when the echo period is too short for the ear/brain combo to get inside and realize that what we've got here is the same signal separated by too short a time.

The amount of delay required to create ADT is dependent upon the attack of the instrument or voice being doubled. To make one lead vocalist sound like two in unison, delay settings between 25 and 40 milliseconds work best. Softer attacks, as with strings and vocal choruses require longer delay lengths — — up to 100 ms. — — and sharp attach insturments such as percussive guitars and drums beg for delays of 20 ms and under.

In no case is the delay period really critical. Below the most effective ADT setting, delays will tend to thicken the sound of the instrument, if not actually reinforce the instruments' sound level, without introducing the illusion of unison. At very short delay times, such as 10 ms or less, flanging effects will be introduced as the period of one cycle of the signals' frequency becomes identical with the delay period; i.e., a 1000 Hz signal has a per cycle period of 1 millisecond; this fact becomes the basis for both analog and digital flanging.

Flanging With Delay

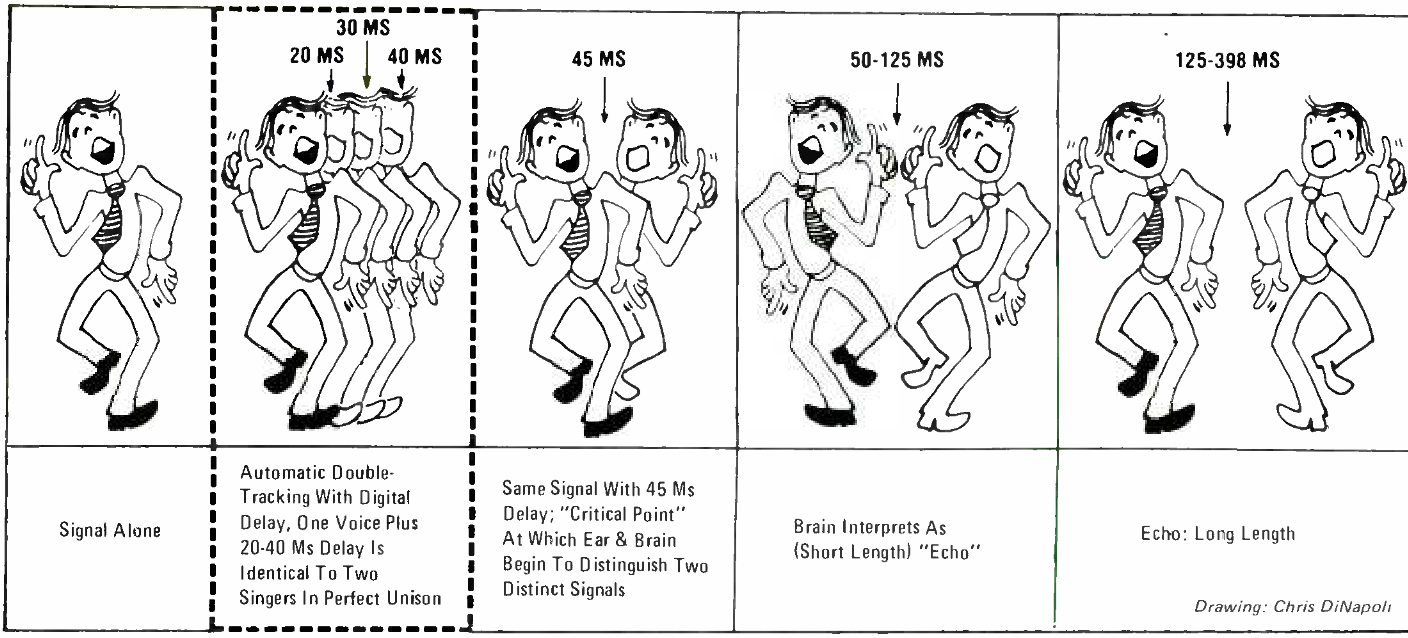
Real flanging — - an actual duplication of the cause and effect of the flanging effect created by two tape machines, a VFO and myriad patch cords — is possible because of continuous (or almost-continuous) delay.

With the original two-tape-machine-method we'd synchronize two tapes of the same material and then ruin the process by brushing up against the reel (flange) and knocking them ever-so-slightly out of synch.

The resulting cancellation creates the effect of an apparent continuous variation in tonality.

With digital delay lines offering such tiny delay steps as to quality as almost-continuously-variable, flanging is produced as one delay is swept back and forth past another. This same effect is achiveable by analog delay lines utilizing recent "bucket brigade" chips. With digital systems the flanging may be made to appear after a given delay period following the direct signal. (It's even possible to "pre-flange," by flanging between two closely-set outputs and then having the "direct" signal appear some milliseconds later!)

The actual mechanizm of flanging with a digital delay line is to set two outputs within 10 to 15 milliseconds of each other and sweep them back and forth. Flanging can, of course, be done with a single DDL output against the original signal or, (sometime before or later) between two DDL outputs.



Comb Filters: Useful, Misunderstood

Yet another creation of the delay line is a uniquely applicable capability known as *Comb filtering.* The name by which it is known describes the characteristics of this filter quite well. The notch frequencies produced are regular and look much like the teeth of a comb. The notch frequencies are related and evenly spaced, like the harmonics of a non-sinusoidal signal.

To null a given frequency and its harmonics, a delay time equal to 1/F is established, and the input and output signals are added together out of phase. To null a given frequency and its odd harmonics, delay is set at 1/2F and input is added to output in phase.

Specifically, say we have to clean up a tape with heavy 60 cycle hum and its attendent harmonics at 120, 180, 240 Hz., ad infinitum. This is a common situation. To instantly establish a comb filter which sharply banishes such components a delay time of 1/60/sec, or 16.666 milliseconds, output added to input out of phase, is established. A nearly continuously variable delay line can be tuned precisely and be practicable well after individual notch filters become untenable. Several delay lines can be utilized to create complex filter characteristics which do their work while leaving broadband programs relatively unaltered.

It should be pointed out that there are remarkable similarities between flanging and comb filtering. In flanging, whether it is produced through two tape machines, a bucket brigade chip or a digital delay line, cancellations occur at harmonic intervals (which is what causes the flanging effect). With a flanger, the fundamental frequency null, along with its harmonics, changes owing to changing delay times. With a comb filter, delay time is stationary — a null is created at a fundamental frequency and its harmonics (if added back in phase) or odd harmonics (if out of phase).

A comb filter is nothing more than a stationary flanger. . . . a flanger nothing more than a moving comb filter!

The next time someone claims that his channel equalizers are so good that he "can flange with them," remember that many many filters may suggest — — — but do not a good comb make.

Good Bye Four Track

Obviously, a digital delay line of sufficient length (the Eventide 1745M has a max delay of 640 ms) can be used to create echos of any length. Delay lines usually have more than one output, and each output can be set to a different delay period — — — one for ADT, one for short length slapbacks, one for long term echos, etc. A delay line thus becomes a tremendous improvement over the old four-track machine for echo effects: the DDL offers better specs (with a typical dynamic range and S/N ratio of over 90 db), more flexibility in exactly setting an echo period that works perfectly with the music and, perhaps most importantly, indefinite unattended operation with no tape to rewind, remand or replace.

Final Plays

There are other interesting applications of delay lines which don't necessarily fit in with the explorations we've made so far.

*One of these is a vaguely mystical arrangement which allows for the "precognitive" or anticipatory keying of, say, a compressor. Given that compressors are wont to take their own good time to pump in and do their work, we can eliminate this key-in modulation by triggering the compressor with an instantanious program signal and setting the delay so as to hold back the program information until the compressor has had time to work. This arrangement will allow for heavier compression without time lags and other clues to our manipulations.

*Delay lines can be designed, as has been the 1745M, with a digital recirculation feature which allows the push-button trapping of a snip of signal and its repitition, without degradation, forever. Typically, the *loop* so formed can be as long as the delay lines' maximum delay.

Once trapped, the signal can be played with if the delay line is of the almost-continuouslv-variable type, and it's pitch changed. Multiple outputs allow for delay times to be changed *after* the signal is trapped and the creation of interesting syncopations between the trapped signals.

Used on stage, this feature allows an artist to intone "Awright!," hit a foot-switch connected to the recirculation bus, and back away from the microphone while his voice continues "Awright! Awright! Awright!" to the befuddlement of mystified fans.

*Once information is in the memory of a RAM–based DDL, it can be read out backwards or, in fact, in any programmed order. Interesting scramblers are suggested through this capability.

*One last note, and a suggestion directed at DDL manufacturers of all shapes and sizes: ADT through a DDL has one negative quality when compared with the unison produced by two singers: it has just a bit of an electronic sound. The automatic unison is just too perfect; delay one voice by 30 ms and it will stay delayed by 30 ms.

In real life, unison is less than perfect, and this makes it endearing. In addition to slight time differentiations between even the most attentive duos, there are slight pitch changes between their voices.

Converting a DDL to be less accurate is, technically a simple task. Every delay line should have an accessory module availabe which will randomize the delay slightly, so as to create a more natural sounding unison.

The Eventide Harmonizer, which in addition to being a versatile pitch changer is also a full-fledged DDL, has this capability to a limited extent.

Remembering the cycling delay time that causes Doppler Shift, let's imagine that we set the pitch changer for a unity pitch ratio, or a pitch ratio within a few points either side of 1.00. In this case, the cycling is slowed down to a point where little Doppler shift occurs; also, delay time will vary slowly between 0 and 25 ms. The result is a natural sounding unison signal — — — even superior to common ADT — — — which contains infinitesimally small changes in pitch and seemingly random delay times at the start of each of the vocalists' words.

The drawback here is that, because the delay begins at zero, flanging will be added to the return signal until it passes through about 10 ms.

By inserting a set delay of 10 ms to the cycling period (thus having the delay cycle from 10 through 35 ms.) the flanging can be eliminated and natural sounding unison results.

Adding this capability to any delay line involves only a randomized voltage control, white noise generator (for true randomness), or an extra hand on the delay-set control of a continuously-variable unit. Either way, it's within the grasp of anyone wishing to experiment, until the manufacturers make such a deseireable option available.

Ya all hear?

Digital Delay In Speaker Reinforcement

Though certainly not a recording effect, the utilization of one or more delay lines to synchronize speakers for sound reinforcement is (a) a major use of the delay line and (b) something that recording people probably ought to be familiar with, because digital delay speakers synchronization is becoming almost mandatory at large theater and festival sites. Indeed delay lines are now specified as standard equipment for all New York Broadway shows, have been part of the PA system at nearly every major outdoor music festival since 1973, and are being included in the systems of more and more church and university auditoriums as original equipment. What's interesting is that, while a handful of delay lines have so much to offer to any PA meant to reach people 100 or more feet distant, DDLs haven't yet penetrated venues where they might do the most good. . . . . . .ballparks and portable systems used at political gatherings. Perhaps the $12,000 DDL system that graces the newly opened Yankee Stadium will turn the tide.

If we assume that sound travels at a rate of one foot per millisecond through the air (which is close enough to its actual velocity) it's obvious that a listener 100 feet from the stage will not hear a speaker's voice from the stage speakers until a tenth of a second after the words are spoken. So we're out of synchronization.

If the entire venue is fed from a single set of stage speakers, this lack of synch is not a problem. . . mouths are too small to be compared critically at that distance.

But large sites require several stages of reinforcement speakers; typically, every 50 or 100 feet. A 150' room with stage speakers (stage A), and reinforcement speakers set back every 50 feet (B at 50 feet, C at 100) will produce a synchronization discrepency of 50 ms between each stage. A listener in the last row will hear first the signal from the stage C speakers, then, 50 ms. later, an echo from the later arriving stage B speakers and finally, 100 ms. later a clearly discernable though weaker and totally confusing signal from the stage A speakers. Hello, hello, hello. We've all heard this time and time and time again.

One alternative is to have a single set of speakers mounted at the stage, pumping a SPL of 130 db., so as to cover the entire hall. This works for *rock* owing to the biological peculiarities of the audiences, but for church and political speeches and commencement addresses, it will never do, and survival becomes a rough proposition for front-row listeners.

With multiple reinforcement speakers, delaying stage B and C signals so that they don't feed their respective speakers until stage A (and B, respectively) signals arrive through the air at their respective positions eliminates all echo. In addition, each stage can be run at a more moderate sound pressure level because it has less territory to go alone and because the distant speaker stages receive actual loudness enhancement from additive signals arriving in synch from the front speakers.

The whole process requires only that the distance between speakers be measured, that distance be converted into milliseconds, and that appropriately set delay lines be inserted into the feeds of the reinforcement stage amplifiers, Oula-la, Oula-la. Ladies and gentlemen good, clean sound.

Step II – Use The HAAS EFFECT. It's Magic And It's Free!

Magic is added to the above synchronization by adding a few milliseconds to each reinforcement stage's delay period. Stage B is now delayed not by 50 ms., but by 55. Stage C is delayed by 105. Why?

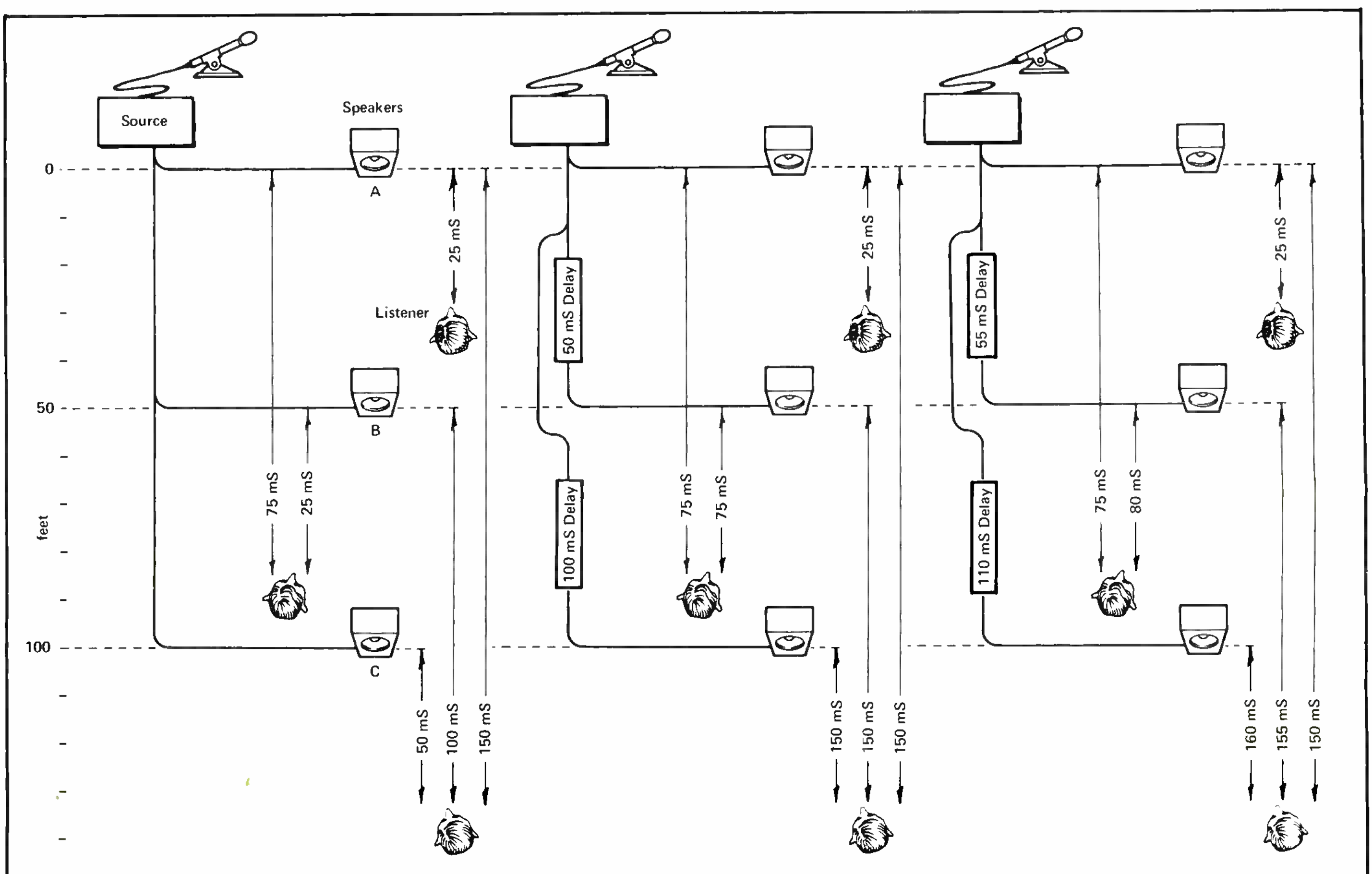
To take advantage of a psycho/acoustic dream come true: The Haas Effect.

Instead of delaying speaker stage B by 50 ms. to achieve perfect synchronization we delay it roughly 5 ms more, so as to ensure that the stage speaker signal (Stage A) reaches distant listeners *first*, likewise, reinforcement stages C, D, etc., are also delayed by an additional 5–10 ms more than the delay required for absolute synchronization.

Dr. Haas' magical effect is a strategem which couldn't be better. It's as if nature planned in advance for the possibilities opened through digital delay, and biology encouraged the only recent advent of good stadium/theater sound.

The Haas Effect? It says that a blindfolded listener, when asked to point to the place from which he believes the sound *originates*, will point not to the source of the *loudest* but to the source of the *first* arrived signal. Even with a 3 or a 6 db level difference, a properly Haased system will have distant listeners pointing to *the stage*, even if a reinforcement speaker twenty feet to their right puts out a louder relative signal!

The stage is where we'd like the signal



SIMULTANEOUSLY FED SPEAKERS AND THEIR MULTIPLE ECHOS

SYNCHRONIZED SPEAKERS FED THROUGH DIGITAL DELAY LINE

HAAS SYNCHRONIZED SPEAKERS; SOUND APPEARS TO COME FROM STAGE to seem to come from, right. Haas makes it possible to have it this way; and, once again, synchronized signals from several sets of reinforcement speakers have additive effects to the loudness anywhere in the arena. We have our cake and eat it too.

Future Haas:

The ramifications of the Haas Effect include applications (potentially) in recording, as well as with effects-mixes for theaters, concerts and motion pictures.

Multiple miking has not been uncommon in recording — microphones at different distances are routed in the mix to position the instrument somewhere in the stereo sphere. A possibility which will probably be getting a lot of work within the next few years is the use of precisely controlled delay times, rather than currently used standard volume panning, to assign position. Positioning by *source-of-first-arrived-signal* is a closer approximation of nature's way of positioning than is volume panning.

The human ear/brain is surprisingly. . . incredibly . . . sensitive to arrival time differentiation as the key element in positioning. The brain is, in fact, able to distinguish position variations by arrival time differences of as little as 1 millisecond. (Which still is a lot of time compared to microseconds and nanoseconds that are so important to the grand scheme of things physical).

Delay panning was employed experimentally on parts of the most recent Grateful Dead release (Steal Your Face, United Artists Records). A number of 1745M delay lines were stacked to assign left-right positions based on Haas arrival time.

With this method, and a lot of work needs to be done so I don't hold fast to this, left-right position would be assigned by delay. Loudness differences between the left and right channels would create illusions of depth.

In practice a computer is used to determine the actual delay times and loudness parameters for a given position, but it seems possible that such position assignment would create a more natural mix, as well as an increased depth perspective, than the panning we now commonly employ.

Yet another possibility would be to set up three-dimensional panning in theaters and live concerts; by varying delay settings to individual reinforcement stages (within the Haas Effect range) more weight could be assigned to rows AA or WW, establishing depth as well as left-right position.

Last Call

Delay lines — particularly the latest breed of almost-continuously-variable systems — are fun to play with. Feedback excursions and cycling delay loops create tricks with time, pitch and tonality that can easily be the jewelike center for a multitude of hit records.

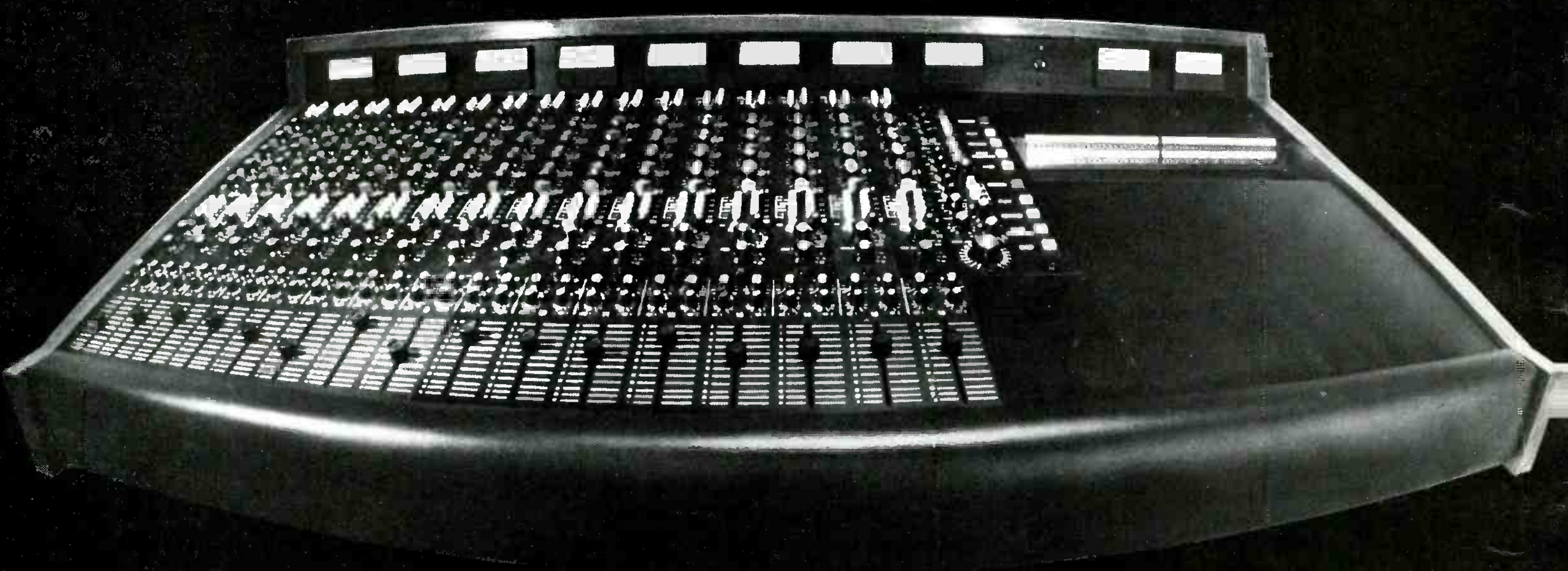
Now that units are available which are, to all intents, continuously variable and fully remote controllable, only cost is the factor preventing stacks of multiple delay channels from being incorporated into mixing consoles as almost-standard equipment.

If nothing else, throwing a few milliseconds of delay, or a tiny increment of pitch change into one side of a stereo mix is fun and distinctive. The delay line is one of the few black boxes versatile enough to be used both as a special effect and as a neutrally colored tool.

The most impressive signature to the usefullness of such units is that, after using them on record mixes, over 100 bands have laid out the few kilobucks necessary to bring one along on the road. Among them are the Who, Beach Boys, ELO, ELP, Peter Frampton, Led Zepplin, Grateful Dead, Lynyrd Skynyrd, Eagles, Kool & The Gang, Yes, Aerosmith, Billy Cobham, Bad Company, AWB and dozens of top PA companies.

Happily, recent technology had made DDLs and pitch changers such as Eventide's Harmonizer affordable — — — in the under $2000 range, and opened up the possibilities suggested in this article to more and more studios and groups.

SWEET SIXTEEN



When you stepped up from four tracks to eight, you knew sixteen wasn't far off. And if you've been looking for someone to put it all together for you, consider it done.

The el-tech 1616-25 recording console gives you all of this:

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New Products

CROWN INTRODUCES THE EQ–2 STEREO EQUALIZER

The CROWN EQ–2 is an eleven band per channel stereo equalizer, providing full equalization from 20Hz to 20KHz. Each band features a ± 15dB boost or cut capability. The filters are of half octave constant bandwidth design, set nominally on octave centers. Each filter has an associated control allowing ±.5 octave adjustment of the center frequency. The two channels can be cascaded to produce a full range half-octave equalizer.

Flexible shelving-type tone controls allow the user to adjust the bass or trebel frequency response before detailed equalization. The combination of tone control and minimum phase filters permits extremely smooth equalizing. Transformerless balanced inputs and outputs providing either unity or switched 10dB gain are featured for the professional user, along with unbalanced inputs and outputs for other audio applications (the unbalanced inputs include an attenuator control to maintain overall system balance when boost controls are in use).

Clip level indicators monitor four critical internal points in the circuitry to signal overloading of the EQ–2. An automatic five-to-seven second muting at turn-on prevents the passing of system turn-on transients to the speakers.

The EQ–2 comes with a full warranty covering 3 years of parts, labor and round trip shipping. Each unit is accompanied by a proof of performance sheet showing actual specifications obtained in final factory checkout.

The EQ–2 may be rack mounted, or mounted in an optional cabinet which is available.

Price: $899.

CROWN INTERNATIONAL, 1718 W. MISHAWAKA RD., ELKHART, IND. 46514 (219) 294-5571

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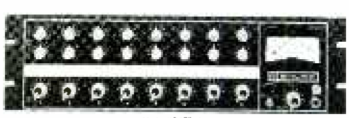
SHURE EXPANDS LINE OF SOUND REINFORCEMENT COMPONENTS

The new products are the SR101 - Series 2 audio console, the SR110 professional monitor mixer, and the SR109 professional mixer.

The SR101 - Series 2 is a rack-mountable, eight-channel mixer/preamplifier that affords exceptional system versatility, complete, accurate controls, and highest quality performance. It has an accessory monitor output for connection of one or more new SR110 professional monitor mixers for use in stage monitoring (foldback), multi-channel tape recording, or stereo broadcasting.



Model SR109 Mixer

Model SR110 Monitor Mixer

Model SR101 Series 2 Audio Console

The new SR110 professional monitor mixer is a rack-mountable, eight-input, single-output line level mixer that can be used when a separate stage monitor (foldback) mix is needed in a sound reinforcement system. Several SR110's can also be used in making multi-track recordings, or stereo and quadriphonic broadcasts (two for stereo, four for four-channel, etc.) or as a mixdown panel associated with a sound reinforcement system. Up to eight SR110's may be stacked and connected with a single SR101 - Series 2 audio console or SR109 professional mixer. A mix bus allows two SR110 mixers to be interconnected, providing 16 inputs when connected to two SR101 - Series 2 consoles or two SR109 mixers.

The SR110 may also be used as an accessory to the new Shure SR109 Professional Mixer. The SR109 is a solid-state, eight-channel microphone mixer/preamplifier that enables the operator to mix as many as eight microphones with individual control over volume, and high - and low-frequency equalization. The SR109 also features an adjustable peak limiter with LED (light emitting diode) indicator that prevents output overloading and a peak-responding LED that indicates output clipping level.

User net price of the new Shure SR101 Audio Console is $1,180; the SR110 monitor mixer is $150; and the SR109 professional mixer is $660.

SHURE BROTHERS INC., 222 HARTREY AVE., EVANSTON, IL 60204.

Circle No. 149

NEW YAMAHA S4115H SPEAKER SYSTEM DESIGNED FOR MANY APPLICATIONS

The new Yamaha S4115H is "made to order" for a wide variety of applications. Combined with Yamaha's EF-series inte-

grated mixer/amplifiers or Yamaha's PM-series mixers and separate power amps, it is ideal for club PA systems, perfect as a wide-range uncolored keyboard sound system, or suited as a powerful and clean vocal sound system. Vocalists, are said to especially appreciate the S4115H's even dispersion.

The Yamaha S4115H two-way speaker system is conservatively rated at 100 watts continuous program power, handling peaks of many times that value. It has a sensitivity rating of 101 dB at one meter with just 1 watt input giving this system the efficiency needed to deliver high sound pressure levels when driven by even modest amplifiers. At its full rated power, the S4115H produces sound above 120 dB SPL.

The low frequency section of the new S4115H combines the benefits of a front-loaded horn with a ducted-port bass reflex enclosure. It projects harmonics directly forward with a "shaped" dispersion pattern. The woofer is a 15" model with an edgewound voice coil for high efficiency, and it features a cast aluminum frame for rigidity and reliability. The high-frequency section consists of a Yamaha horn/driver, a combination radial horn and compression driver featuring a simple to replace driver diaphragm. The sound is open and accurate, free of "sharpness" or "edge" that characterizes some horn/driver systems. An RLC (resistance, inductance, capacitance) dual-section crossover network affords smooth transition between the woofer and the compression driver.

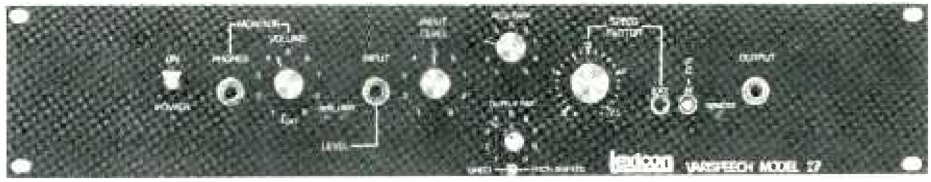
YAMAHA INTERNATIONAL, CORP., MUSICAL INSTRUMENT DIV., P.O. BOX 6600, BUENA PARK, CA 90620.

Circle No. 150

LEXICON ANNOUNCES VOICE RANGE PITCH SHIFTER

Lexicon, Inc. has announced a new real time pitch shifter with special signal mixing and recirculation facilities useful for creation of unusual studio sound effects, as well as compressing and expanding speech in film and video editing.

The new stand-alone system called Varispeech Model 27 is designed for use with variable speed sound reproducers such as reel-to-reel tape units, etc. All units have provisions for external pitch control through remote resistance or voltage programming. In addition, the Model 27 has an interface for speed control of those reel-to-reel transports that have provisions for external electronic speed control. For example, a simple patch card connection allows Varispeech's microcomputer to control a Revox Capstan speed over the entire expansion and compression range of 0.5 to 2.5 times normal speed allowing perfect restoration of voice pitch at any speed setting.



All units feature Lexicon's exclusive patented "intelligent splicing" which eliminates splice noise or pop common in other pitch shifting techniques. Pitch and speed are controlled by a single calibrated speed factor dial. Other features include light emitting diode level indicator, built-in headphone amplifier and both XLR-3 and quarter-inch input and output connections.

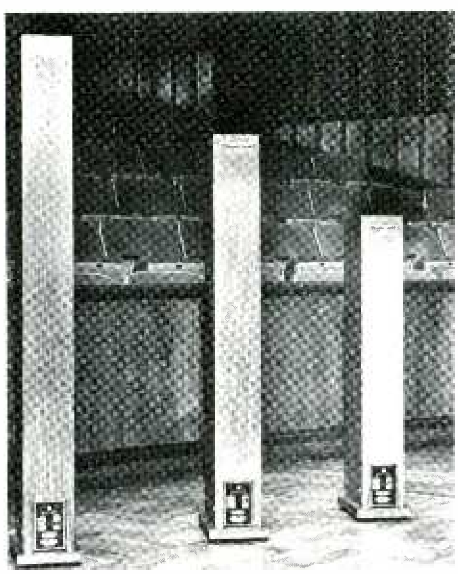
Model 27 Varispeech is priced at $750 with delivery in 30 days.

LEXICON, INC., 60 TURNER STREET, WALTHAM, MA 02154, (617) 891-6790.

Circle No. 151

EXTENDED BASS RESPONSE IN MASTER–ROOM REVERBERATION CHAMBERS

MICMIX Audio Products, Inc. announces the availability of extended bass response characteristics in its line of Master-Room tm Reverberation Chambers. Normally rolled off at 6 dB per octave below 200 Hz, extended bass



response units carry the flat response to 50 Hz and are less than 8 dB down at 20 Hz. Units with extended bass characteristics are available on new orders at no additional charge and parts kits are available from the factory at nominal cost for conversion of existing units in the field. The extended response on the Master-Room chambers is extremely clean, according to the manufacturer, with no evidence of 'tubbiness' or similar distortion which is so often associated with low end characteristics of most reverberation chambers.

MICMIX AUDIO PRODUCTS, INC., 9990 MONROE DRIVE, SUITE 222, DALLAS, TX 75220, (214) 352-3811.

Circle No. 152

See our ad on page 63

LOW IMPEDANCE "MIC SPLITTER" INTRODUCED BY SESCOM

A new device termed a "Mic-Splitter", that's designed to split the output of a low impedance microphone two ways, is now available.

The lightweight and durable Model MS-1 can feed two mixers simultaneously with isolation between the two units at 30 dB. Either output can be shortened with only up to .1 dB effect on the other output.

The new unit features a special Sescom transformer with isolation resistors and ground lift switch for the right hand output connector.

A passive device that does not require batteries or other power source, the Mic-Splitter comes in a rugged die cast aluminum housing and measures 4-11/16" x 2-11/16".

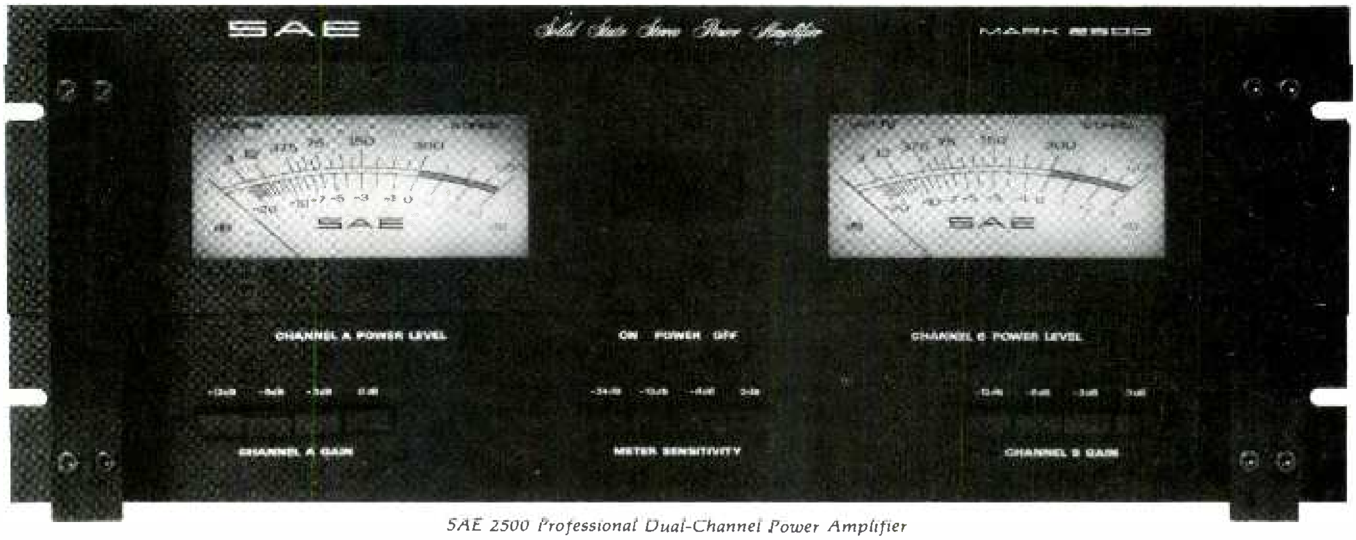
Important specifications include: primary impedance at 150–250 ohms; secondary impedances at 150–250 ohms (two); input level at -10 dBm @ 30 Hz .2% THD.

The Model MS-1 is available for immediate delivery and is priced at $36.30 per unit. For additional information on this product or any of the Sescom broadcast accessory equipment:

SESCOM INC., P.O. BOX 590, GARDENA, CA 90247, (213) 770-3510.

Circle No. 153

Powerful alternative.



SAE 2500 Professional Dual-Channel Power Amplifier

When you compare power amplifiers, you have to look at the hard facts. The SAE 2500 Professional Dual-Channel Power Amplifier has them—top power, specifications, reliability and features that make it the most "powerful alternative."

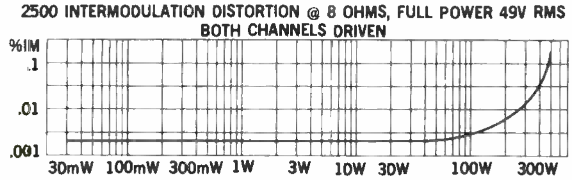
Power. 450 Watts RMS per channel, both channels driven into 4 Ohms from 20Hz to 20kHz at no more than 0.1% total harmonic distortion. Or, 300 Watts RMS per channel, both channels driven into 8 Ohms from 20Hz to 20kHz at no more than 0.05% total harmonic distortion.* Plus, a new, smaller wide-channel power transformer coupled to 4 computer-grade capacitors for a power supply that varies no more than 10% from no load to full load. (For extra protection, there are relay and thermal cut-out devices.)

Other Specifications:

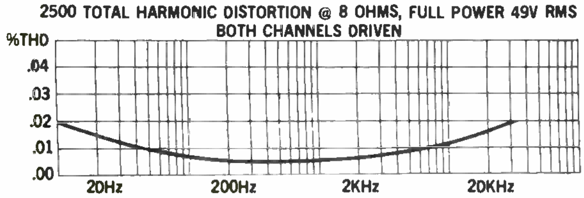
IM Distortion from 250mW to rated power at 4 or 8 Ohms with any 2 mixed frequencies between 20Hz and 20kHz at 4/1 voltage ratio..........0.05 Max.
Frequency Response at rated power.................... ±0.25dB, 20Hz to 20kHz
Noise..Greater than −100dB below rated power
Transient Response of any Square Wave .... 2.5μsec. rise and fall time
Slew Rate ....40V/μsec.
Dimensions .... Front panel: 19"Wx7"H, Chassis: 15¾"D (excluding handles, controls and connections)

*These specifications comply with FTC requirements for power amplifiers.

2500 INTERMODULATION DISTORTION @ 8 OHMS, FULL POWER 49V RMS BOTH CHANNELS DRIVEN



2500 TOTAL HARMONIC DISTORTION @ 8 OHMS, FULL POWER 49V RMS BOTH CHANNELS DRIVEN



Reliability. The SAE 2500 gives you high current capability with Parallel-Series-Output Circuitry (PSO)—without loss of wide power bandwidth, low leakage current or super-high slew rate. Sixteen triple-diffused output transistors have an electrical and thermal SOA 50% higher than maximum design requirements for reliable high demand capability. This configuration can handle anything from continuous full signals to highly reactive surge loads—all day long without failure or overheating. Dual relay disconnect circuits and plug-in board design further assure reliable performance.

Features. Feedback level controls assure a constant input impedance of 50k Ohms and reduce the noise figure to more than 100dB below rated output in all positions. Loudspeaker protection relay-activated circuit automatically disconnects speakers in case of ±DC outputs. Plus, direct power reading VU meters and forced air cooling.

The SAE 2500 Professional Power Amplifier weighs only 58 lbs. making it practical for portable sound reinforcement, public address, communications and recording applications.

The professional alternative.

SAE

Scientific Audio Electronics, Inc.
P.O. Box 60271
Terminal Annex
Los Angeles, California 90060

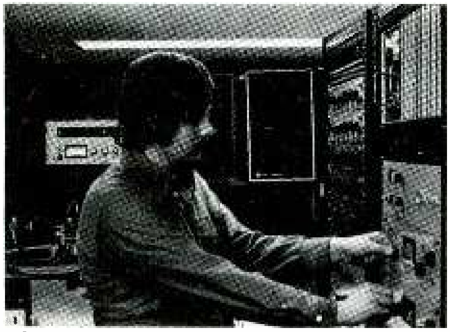
Please send me the reasons (including available literature) why the SAE 2500 Professional Dual-Channel Power Amplifier is the "Powerful Alternative."

Name ____________________

Address ____________________

City ____________ State______ Zip______

Circle No. 154



"Carl Rowatti, Chief Engineer, adjusting the Program limiters prior to cutting a master lacquer".

SOUND WORKSHOP 220 DOUBLER/ LIMITER

The new SOUND WORKSHOP 220 Doubler/Limiter incorporates an electronic delay system and a studio quality peak limiter to provide 2 versatile signal processors in 1 device.

The delay system uses analog means to provide from 5 to 40 milliseconds of delay. An output mix control is provided so that the direct signal, the delayed, or a mix of both can appear at the output.

The front end of the 220 incorporates a studio quality peak limiter with a slope of 20:1. The attack and release times have been set to provide accurate control of peaks without any breathing or pumping sounds. The limiter may be used independently or in conjunction with the delay mode.

The 220 has both mic and line inputs and outputs to allow interface into simple or complex stage, studio, or home systems. The 220 can operate directly into a tape deck or PA system without the need for consoles or external mic preamps.

The 220 can rack mount or sit on a table top. It sells for $500 and is warranteed for 2 years parts and labor. **SOUND WORKSHOP, 1038 NORTHERN BLVD., ROSLYN, NY 11576, (516) 621-6710.**

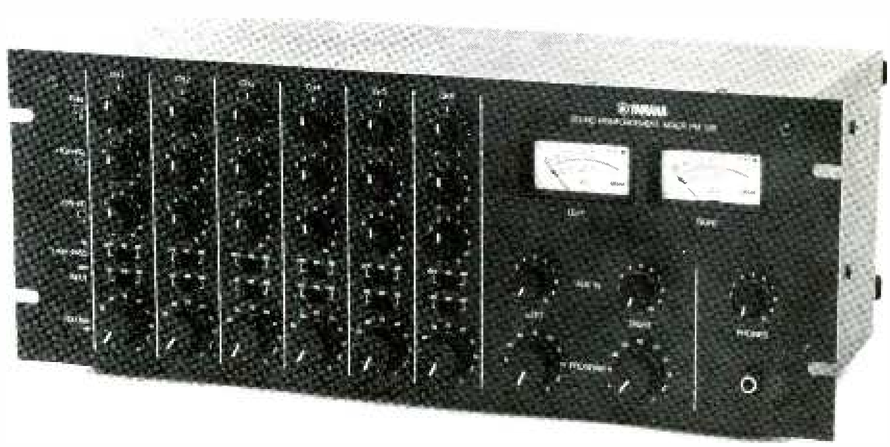
Circle No. 156

TWO NEW COMPANION SUBMIXERS FROM YAMAHA

The new Yamaha PM-170 and PM-180 professional sound mixers bridge the gap between basic mike mixers and larger sophisticated mixers. The new units are said to offer unparalleled features, flexibility, performance and reliability in compact, handsome and reasonably priced packages. They feature illuminated precision VU meters, EQ and filters on each of the 6 inputs, switchable input and output levels, and true +24 dBm into 600-ohm output capability.

The new Yamaha PM-180 offers plug-in compatability with low-impedance electronic instruments, tape machines, microphones and a variety of accessory equipment with all input channels transformer isolated XLR connectors. The PM-170 unbalanced phone jacks for plug-in compatability with electronic instruments, low and high impedance microphones, tape machines and accessories. Both units feature INPUT level switches on all channels, built-in 2-frequency HIGH PASS filters, LOW and HIGH frequency equalization ±15 dB maximum boost and cuts on all inputs, channel assignments via stereo PAN pots, master PROGRAM controls, a headphone jack, and much more. Two Program Mix Buses can be used to create stereo programs, or two discrete monaural programs, and if desired one bus can be used for the program and the other as an echo or monitor channel.

The prime differences between the new PM-170 and PM-180 are distinct. Volume and equalization controls on the 170 are arbitrary (scale of 1-10), the 180 are dB calibrated. Channel inputs and main outputs on the 170 are 6 phone jacks (unbalanced) and 2 unbalanced phone jacks, and on the 180 are 6 XLR's (balanced) and 2 balanced XLR's. The PM-170 is specially designed for submixer



use with Yamaha's EM-series integrated mixer/amplifiers, while the PM-180 is built for use with other Yamaha PM-series mixers and consoles.

Both units feature totally modular circuitry; metal chassis completely enclosing the circuitry, shielding it from sources of hum, static and noise, including RFI (radio Frequency interferency); and rugged rack mounting making them just right for permanent installation as well as ideal and tough for the road.

YAMAHA INTERNATIONAL CORP., MUSICAL INSTRUMENT DIVISION, P.O. BOX 6600, BUENA PARK, CA 90620.

Circle No. 157

NEW PRECISION MONITOR LOUDSPEAKER SYSTEMS ANNOUNCED

This new line of monitor systems utilizes components chosen from several major manufacturers. Incorporating the latest developments in compression drivers, horn design and woofer technology with custom cabinetry, the PROFESSIONAL MONITOR STANDARD Loudspeaker Systems offer many important performance improvements.

Test measurements shows mid and high frequency distortion has been drastically reduced thus eliminating the "honk" common to most horns. The use of these new horns maintains better than 90^0 dispersion to beyond 20 kHz. New woofers offer efficiency improvements as much as 10 dB over conventional monitor systems.

Systems are available in standard monitor configuration, an extended bass model, and a high power model suitable for rock performance use. Cabinets are available in studio utility black, Formica or fine wood veneers, and a heavy-duty Finnland-Birch road cabinet. Available with crossovers, or wired for bi-amp use, systems prices start at $649. Engineering data test results and comparison tests available upon request.

PROFESSIONAL AUDIO SYSTEMS ENGINEERING, 7330 LAUREL CANYON BLVD., NORTH HOLLYWOOD, CA. 91605, (213) 982-1141.

Circle No. 158

NEW SHURE ADD-ON MIXER ADDS VERSATILITY TO MULTI-MICROPHONE SOUND SYSTEMS

Named the Model M677, the new unit is designed as a "slave" mixer for Shure products such as the M67 and M68 series microphone mixers, the SE30 gated compressor/mixer, the M610 feedback controller, and the M63 audio master.

When used with the Shure M67 and M68, the new M677 provides a more convenient method of stacking mixers, plus gives two additional microphone or line level inputs over those available when stacking two four-channel mixers.

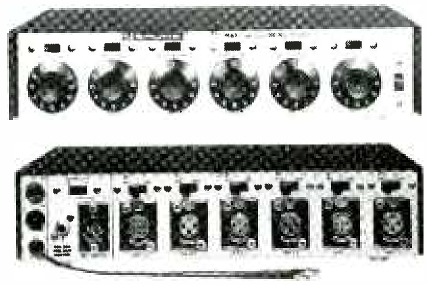
By using an M677 with a Shure SE30 Gated Compressor/Mixer, an operator can convert the three-input mixer system of the SE30 to a nine-input mixer for applications such as high quality sound reinforcement, radio or television broadcasting, and sound recording.

By combining an M677 with a Shure M63 Audio Master, a user can have a six-channel microphone mixer (microphone or line level signals), with very flexible equalization, a 600-ohm line output, a VU meter, and a headphone monitor. The M677 in combination with the M610 provides 6 inputs plus an octave graphic

Circle No. 159

equalizer.

The new Shure M677 Mixer can be powered either from the nominal 28 to 30 Vdc output of the attached master mixer or from a Shure A67B Battery Power Supply.



User Net price of the Shure Model M677 Add-On Accessory Mixer is $181.20. **SHURE BROTHERS INC., 222 HARTREY AVE., EVANSTON, IL 60204**

Circle No. 162

NEW RUSSOUND DIRECT BOX

Russound announces its new IMP-1 Impedaverter, a general purpose "direct box" which allows the recordist or performer to connect any microphone or instrument pickup to any mixer, recorder or reinforcement system input.

The Impedaverter accepts input signals from lowest mike levels up to 1.5 volts and has an output impedance under 100 ohms, permitting it to drive up to 500 feet of line without high frequency loss or hum pickup.

Its dual inputs permit bridging off other lines without loading the source and its dual outputs will drive a sound reinforcement system and a recorder, or multiple recorders simultaneously. Both phone and RCA type phono jacks at inputs and outputs maximize convenience in connecting users other equipment.

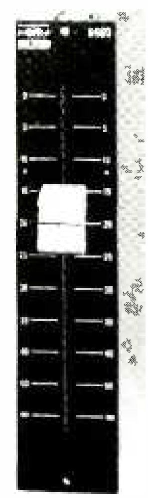
Gain is switchable to -20 dB (for high output microphones), 0, +20, or +35 dB. Phase inversion is also available in the 0 dB position to correct for phase anomalies in associated equipment.

The Russound Impedaverter is portable and can be mounted to a mike stand using hardware supplied. It is powered by easily available 9 volt transistor radio batteries. The IMP-1 costs $59.95 from audio and music dealers or the factory. For complete product information and list of dealers, contact:
RUSSOUND/FMP, INC., BOX 204, STRATHAM, NEW HAMPSHIRE 03885.

Circle No. 163

MODULAR AUDIO PRODUCTS TO MARKET NEW LINE OF PROFESSIONAL AUDIO ATTENUATORS

Designated as Models: 8160–Monaural, 8260–Stereo (2 gang), and 8460–Quad (4 gang), the new units are designed for applications where reliable performance and long life are a must.



The new models utilize a precision conductive plastic resistance element in a truly stepless, 600 ohm constant impedance ladder network configuration in conjunction with multiple finger precious metal wiping contacts. The mirror-finish elements are rated in the millions of cycles of operation, with extremely low noise characteristics.

Resolution is infinite, and tracking accuracy is within ±0.5dB. Maximum attentuation is better than 95dB, with interchannel isolation in multi-gang units greater than 80dB.

A specially developed, rugged slide mechanism, featuring low friction synthetic bearings, assures smooth, uniform operation, and a pressure adjustment is provided to set the "feel" to individual operator preferences. Both the elements and the slide mechanism are carefully protected against dust, dirt and liquid spills by a special gasket seal around the slider arm.

All three models are housed in the same standard size case featuring a black anodized aluminum faceplate with a permanent white epoxy scale, accurately calibrated in dB of attenuation. An attractive slide type indicator knob is provided. Dimensions are 1-1/2" Wide x 7" High x 3-1/2" Deep. External connections are made via a P.C. connector furnished with each unit.

Available as an option, is an internal SPDT microswitch which provides both normally open and normally closed contacts, and operates at the bottom of the slider travel (infinity position) for broadcast CUE, channel ON/OFF, or other desired functions.

MODULAR AUDIO PRODUCTS' INC., 1385 LAKELAND AVENUE, BOHEMIA, L.I., NEW YORK 11716

Circle No. 164

ORBAN/PARASOUND MODEL 111B SPRING REVERB

Orban/Parasound announces the availability of its new dual-channel Model 111B Spring Reverb. Featuring the same basic electrical design as its popular single-channel predecessor, the new 111B offers a new bass control and "quasi-parametric" midrange control which permits stepless adjustment of its ± 12 dB equalization range, as well as continuously variable control over center-frequency and bandwidth.

Included in the new 111B is the unique "floating threshold limiter" which minimizes "spring twang" and provides absolute protection from overload. Also retained from the previous model are line-level balanced outputs and smooth four-spring (per channel) sound.

The Model 111B comes in a standard 19" rack mount and is 3 1/2" high. Price for the dual-channel 111B is the same as the old single-channel 106CX: $695.00 (including power supply).

PARASOUND, INC., 680 BEACH ST., SAN FRANCISCO, CA 94109, (415) 673-4544.

Circle No. 165

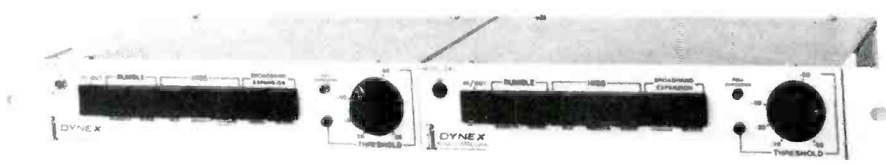
"DYNEX" NOISE SUPPRESSION FROM INOVONICS

The Model 240 is a program-controlled filter/expander which offers a simple, yet effective means of suppressing a certain amount of residual background noise in audio reproduction systems, TV film chains, "phone patches," etc.

Among DYNEX' features are: selective suppression of rumble, hiss or wide-band noise, and restoration of program dynamics by linear broadband expansion. Variable threshold adjustment and visual indication of expander operation are provided.

INOVONICS, INC., 1630 DELL AVE., CAMPBELL, CA 95008, (408) 374-8300

Circle No. 166



Circle No. 167

PEAVEY CS 800 STEREO POWER

Pictured is the CS 800 Stereo Power Amplifier by Peavey, a new ruggedly constructed commercial sound unit.

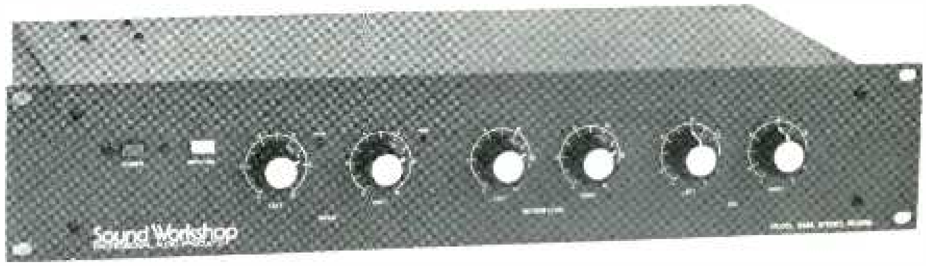
The CS 800 is rated at 400 watts per channel into four ohms and is powered by 24 15 amp – 200 volt power devices. The THD is a low, 0.1% from .01 watts to 400 watts. Frequency response is plus or minus 1 dB from 5 Hz to 30 kHz. The unit is fancooled and boasts a massive 7.6 pound aluminum heatsink.

The CS 800 also has external plug-in capability, providing for balanced inputs or crossover network for each channel. The front panel features L.E.D. peak overload indicators, high temperature indicator light and lighted push button on and off switch.

The CS 800 is one of three new commercial sound power amps offered by Peavey, the others being the CS 200 Mono and the CS 400 Stereo models, both with specifications comparable to the CS 800.

PEAVEY ELECTRONICS, CORP., 711 A ST., MERIDIAN, MS 39301.

Circle No. 168



SOUND WORKSHOP 242A STEREO REVERBERATION SYSTEM

The new Model 242A Stereo reverberation system is a revised version of the popular 242. The 242A now provides dry/reverb output mixing, balanced line inputs, and line level drive into 600 ohms for studio, disco, broadcast, or home reverb applications.

Other features include peak reading LEDS, input mixing for stereo return from a mono send, dual input level controls, reverb output controls, independent channel EQ, and dual mic preamps. Unlike other reverberation systems, the SOUND WORKSHOP 242A can be used directly with any tape deck without the need for a mixing console, or external mic preamps.

Clean natural reverberation is provided in a compact self-powered chasis that may be rack mounted, or used on a table top. The 242A sells for $450 and is warranteed for 2 years, including parts and labor.

SOUND WORKSHOP, 1038 NORTHERN BLVD., ROSLYN, NY 11576, (516) 621-6710.

Circle No. 170

L.J. SCULLY INTRODUCES "THE LATHE"

The L.J. Scully Manufacturing Corp., Bridgeport, Conn., has introduced a new mastering unit called "THE LATHE." Designed to meet the most pricise requirements of the record mastering industry, "THE LATHE" displays the simplicity, ruggedness and maintenance-free operation traditionally inherent in Scully equipment.

The feed and depth systems of the new lathe utilize a proven sample-and-hold approach to generate multiple bit, digital information as to the frequency and level content of the program material. The use of a high bit sampling rate allows minute changes in pitch and depth to occur many times a revolution which the manufacturer claims makes this new unit the most versatile on the market.

Other quality features of "THE LATHE" include: a digital L.P.I. readout for precise and repeatable settings; a 150X Nikon microscope with verticle illuminator; a quick change cutter head mount and saddle designed to accommodate both Westrex and Ortofon cutter heard; plus 16 2/3 r.p.m. turntable speed for CD-4 cutting. Also included is a complete disc playback/calibration system and a computer designed isolation that eliminated external rumble. All functions are fully automated for easier operation. "THE LATHE" is mounted on a heavy duty, welded steel frame for stability.

The development and engineering of "THE LATHE" is the result of direct requests and suggestions from disc mastering engineers throughout the country. The finished product combines the best qualities of conventional lathes with new, state-of-the-art, computer technology.

Circle No. 169

In addition to the "THE LATHE," L.J. Scully, also manufactures the Preview Master, a tape to disc transfer machine which, together with their many other disc cutting accessories, enables Scully to put together a disc cutting system to meet any requirements.
L.J. SCULLY MANUFACTURING CORP., 138 HURD AVE., BRIDGEPORT, CT. (203) 368-2332

Circle No. 171

INOVONICS' TAPE REPRODUCE AMPLIFIER

Inovonics has introduced its Model 376 Tape Reproduce Amplifier, a self-contained, dual channel, reproduce-only electronics package for professional applications.

Among the features of the Model 376 are an optimum combination of IC and discrete circuitry for low residual noise, and 3-speed equalization with wide adjustment range for any combination of NAB and IEC characteristics, 3-3/4 to 30 ips.

Inovonics' Model 376 accommodates virtually any tape or film reproduce head, and with the 01 option can be strapped for either Hi-Z or Lo-Z head windings.

Additional features include remotable solid state EQ switching, and phase compensation adjustment to correct recording errors which degrade multi-generation tape copies.
INOVONICS, INC., 1630 DELL AVE., CAMPBELL, CA 95008, (408) 374-8300.

Circle No. 172

BGW MODEL 100 STEREO/MONO AMP

The BGW Model 100 stereo/mono power amplifier is designed for use as a precision monitor amplifier, for use in bi- and tri-amplified systems as a tweeter or horn driver amplifier. It is also ideally suited for driving electrostatic or conventional headphones.

Unique features include accurate clipping indicators, simple mono/stereo switching capabilities, professional broadcast version available with Cannon style input connectors and 8-pin octal type connectors for input matching transformers.

The circuitry used in the Model 100 is identical electronically to that found in our Model 250B, the differences being only in the power level. Each LED clipping light is driven by a 3-transistor 1-shot circuit. This circuit lights the LED for a quarter of a second whenever the amplifier is clipped.

Specifications: Stereo power output: 40-watts minimum sine wave continuous average power output per channel with both channels driving 4-ohm loads over a power band from 20 Hz to 20 kHz. The maximum total harmonic distortion at any power level from 250-milliwatts to 40-watts shall be no more than 0.1%.

30-watts minimum sine wave continuous average power output per channel with both channels driving 8-ohm loads over a power band from 20 Hz to 20 kHz. The maximum total harmonic distortion at any power level from 250-milliwatts to 30-watts shall be no more than 0.1%.

Mono power output: 80-watts min-

Circle No. 173

imum sine wave continuous average power output monaural driving an 8-ohm load over a power band from 20 Hz to 20 kHz. The maximum total harmonic distortion at any power level from 250-milliwatts to 80-watts shall be no more than 0.1%.

Frequency response: +0, -0.25 dB, 20 Hz to 20 kHz.

Noise & Hum Level: Better than 106 dB below rated output into 8-ohms.

Output Impedance: Designed for any load impedance equal to or greater than 4-ohms.

Size & Weight: 1-3/4" x 19" standard rack front panel x 11" D. 18 lbs. net.

Price: $319.00: with standard ¼" phone jack input connectors. $339.00: with cannon type input connectors and octal input transformer sockets.

Delivery: Beginning August 1976.

BGW SYSTEMS, 13130 SO. YUKON AVE., HAWTHORNE, CA. 90250, (213) 973-8090.

Circle No. 174

NEW ACTIVE EQUALIZER FROM ALTEC

Designed to provide accurate equalization of the entire audio spectrum for Professional and Industrial applications, Altec's new model 1650 Active Equalizer recently introduced contains 28 active band rejection filters at 1/3 octave center frequencies.

According to R.J. Forbes, National Sales Manager, Commercial Products, the 1650 equalizer is designed for use in a variety of Industrial or Professional Sound Systems requiring accurate control of environmental-acoustical conditions.

Each of the 1650's 28 filters provides up to 15 dB attenuation at its center frequency and is skirted to crossover with adjacent sections at -7 dB combining to give ripple-free summation over 85% of its 31.5 to 16,000 Hz range.

"The high- and low-pass filters roll off at 18 dB octave with continuously variable 3 dB down points," Forbes said. "And the control panel is calibrated in a number of high and low-pass frequencies."

Additional features include balanced operation with 150-ohm or 600-ohm output impedance and dual level gain, offering compatible use in high level or low level systems.

A pioneer in the industry, Altec has been designing and manufacturing a variety of industrial professional and consumer sound products for over 40 years.

ALTEC, 1515 SO. MANCHESTER AVE., ANAHEIM, CA. 92803.

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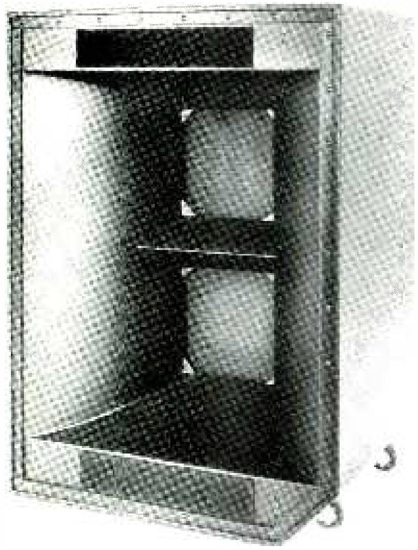
The Model 391 High Frequency Horn Package, a 90^0 horizontal dispersion by 40^0 vertical dispersion radial horn which accommodates a 2" compression driver is one of two new products recently introduced. The 391 is designed primarily for PORTABLE low-distortion high level sound-reinforcement applications and incorporates several unique design features.

The horn is molded of balsa-rigidized fiberglass-reinforced plastic, sandwiched between shells of 1/2" plywood. Support and protection for the driver are provided by sturdy steel brackets. The result is a FLAT PACKAGE which allows several horns to be stacked and aimed without additional hardware.

The Model 391 is furnished complete with professional 3-pin audio connectors and finished in utility black.

The Model 545A Low Frequency Horn Enclosure is a front-loaded horn built for two 15" loudspeakers. The horn-flair is molded of balsa-rigidized fiberglass-reinforced plastic. The cabinet is of 5/8" plywood. At 115 pounds, the resulting assembly is approximately 40 POUNDS LIGHTER than a similar all-wood enclosure.



Further, the cabinet is fitted with recessed casters and handle which allow it to be tipped on edge and wheeled from place to place. Moving a similar wheel-less cabinet would be, at least, a two-man operation. The Model 545A is furnished complete with professional 3-pin audio connectors and finished in utility black.

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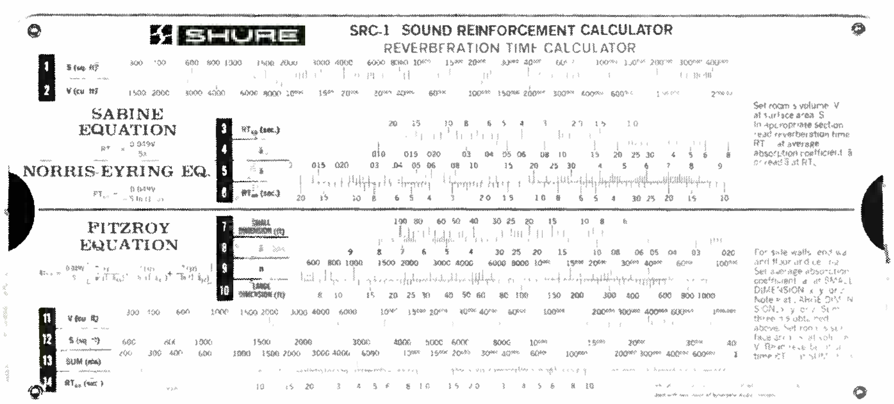

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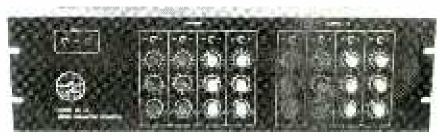

Suggested applications for the SC-66 include feedback control, acoustical correction, improvement of microphone and speaker response, and generation of special effects. Suggested list price for the SC-66 is $599.00, F.O.B. Rochester.

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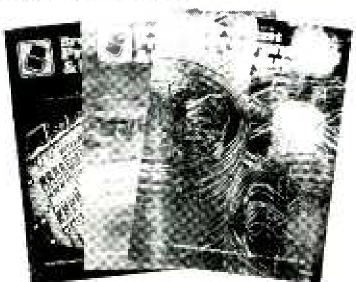

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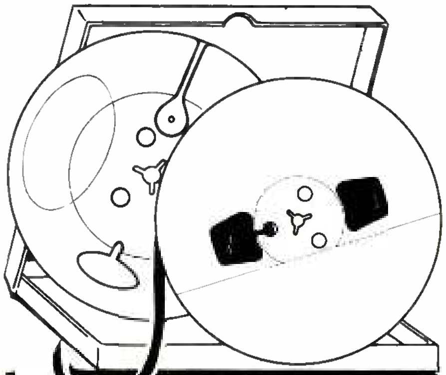

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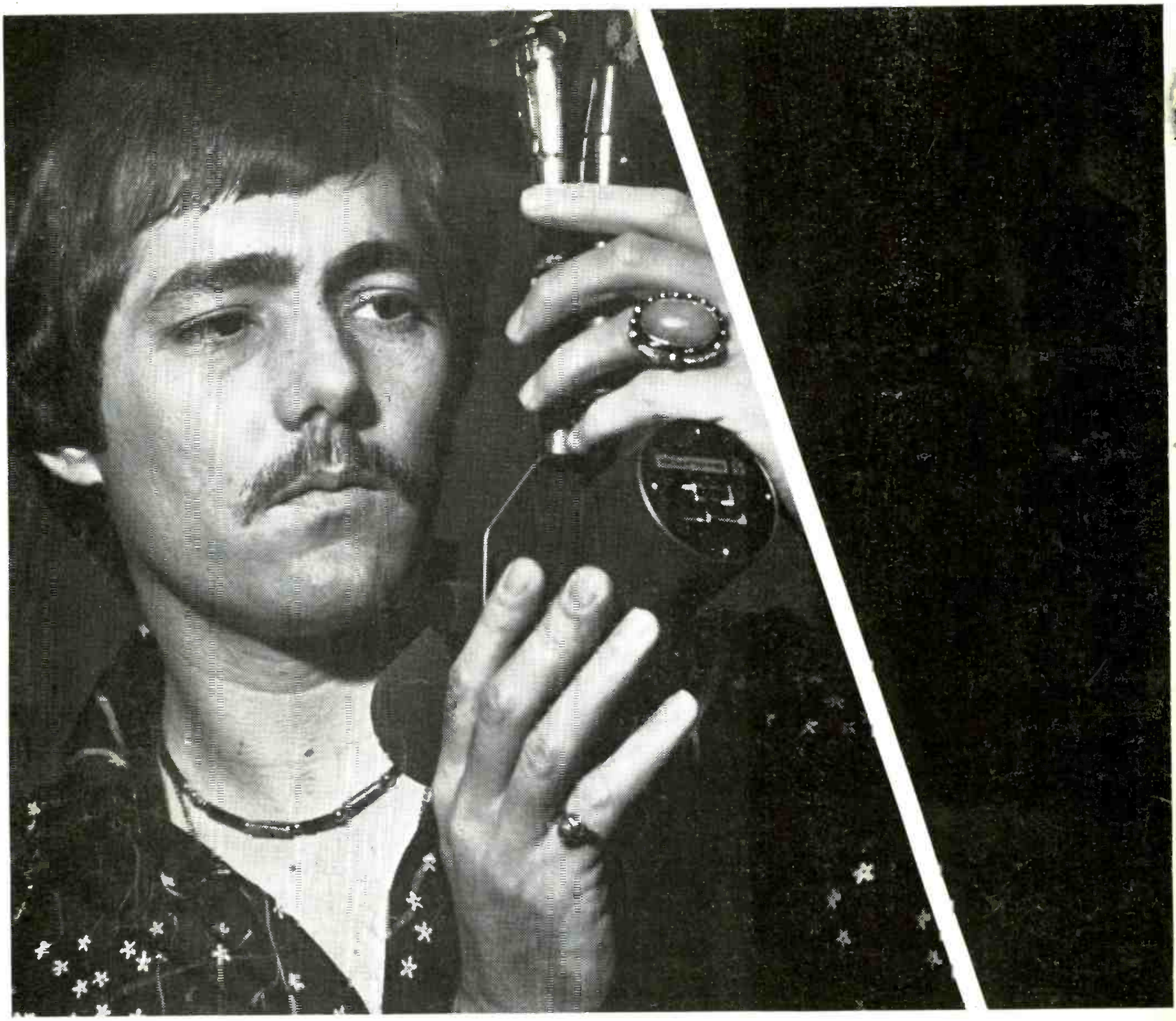

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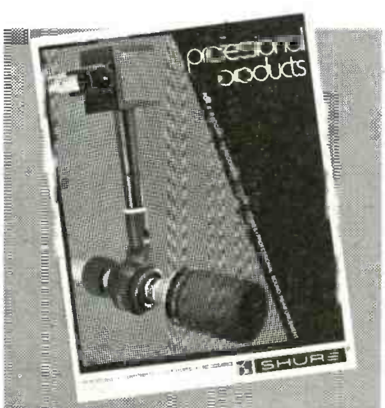

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