RECORDING ENGINEER/PRODUCER

The Future of the
AUDIO CONTROL CONSOLE . . . page 14

CLUDING: CONCERT SOUND REINFORCEMENT

$2.50
October 1980
Volume 11 - Number 5

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Auto-Pak is Amek's computer editing and mass storage system. It is the only console computer that can vocalize all commands when entered and all information as it appears on-screen. The master keyboard selectively lights the correct keys for each function group. The computer also performs as a tape locate and motion control unit, employing the SMPTE time code.

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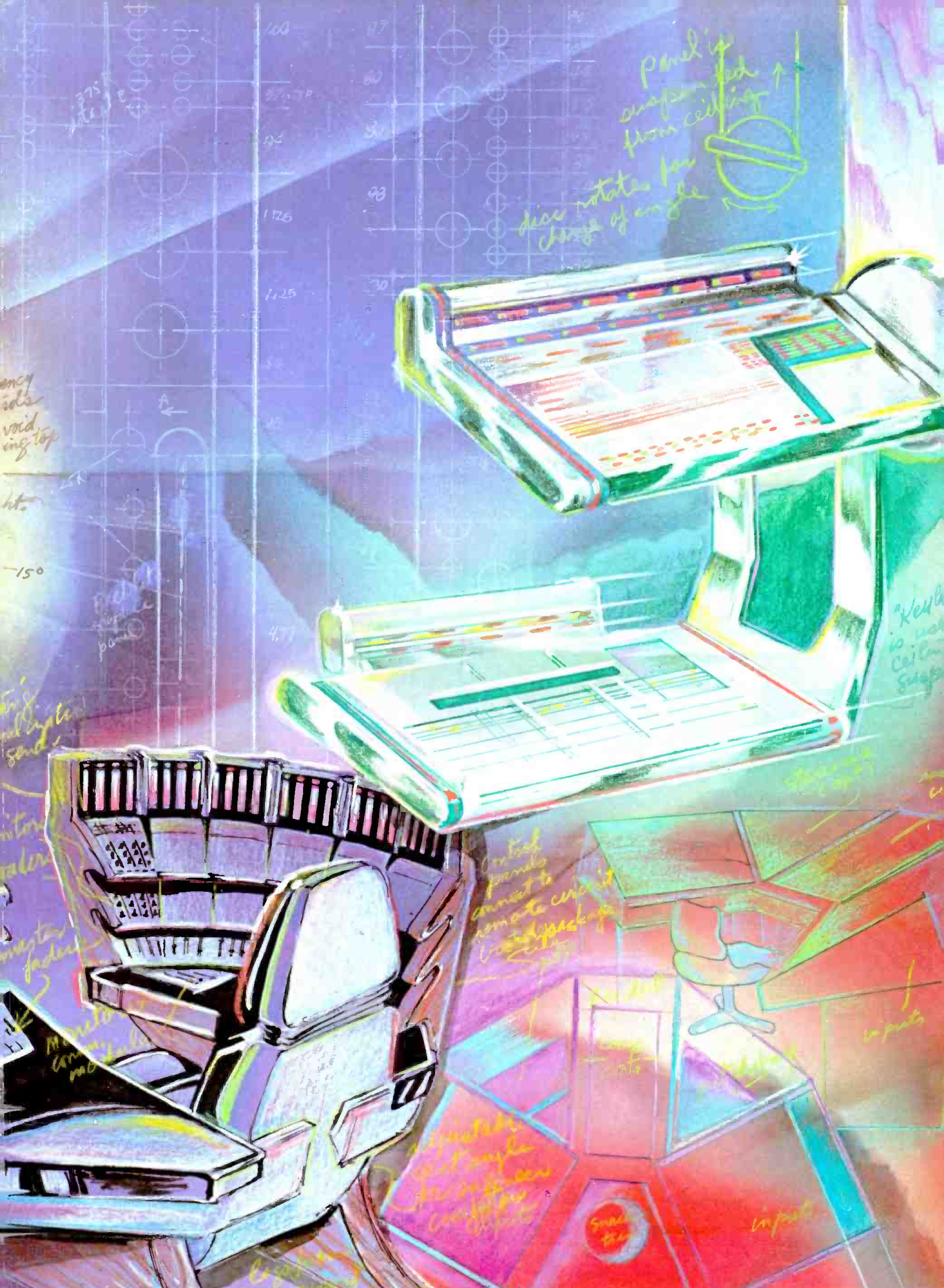
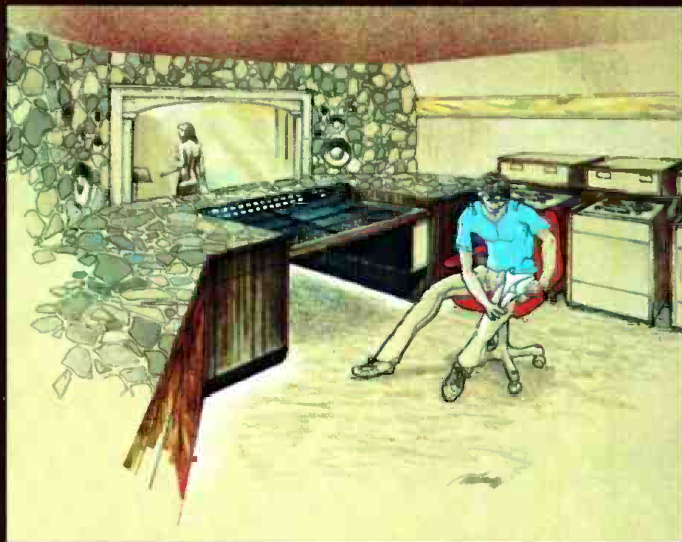
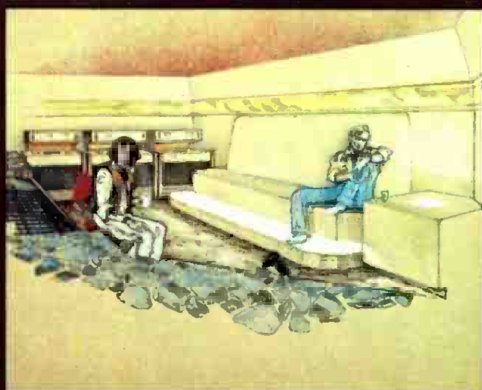

Panel is suspended from ceiling
disc rotates for change of angle
inputs

THE TOTAL ENVIRONMENT CONCEPT



A top-notch acoustic designer: One who is devoted to his craft, one of the best. A top-notch architect, a specialist in recording studio construction. A top-notch interior designer—a Hollywood set designer—one whose imagination is unbounded.



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United/Western Studios | a URC company

September 23, 1980

Quite frankly, studios that succeed in the 80's will have to be sensitive, more than in the past, to the desires and requirements of clients. Nothing new about this, of course, but as producers, engineers and artists become more aware of what can or can't be done with certain mixing consoles, the line of selectivity becomes vividly drawn.

As a studio manager, facing the decision of whether or not to update and purchase a new mixing desk presented me with a number of challenging questions.

"Which console will enhance the needs of present clients?"

"What manufacturer offers the field support we must have and what is the historical consistency of their back-up?"

"Will our volume increase be soon and sufficient to justify such a high impact expenditure?"

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Any further questions?

Jerry Barnes

Jerry Barnes
Vice-Pres. Recording
United Recording Corp.
Hollywood, California

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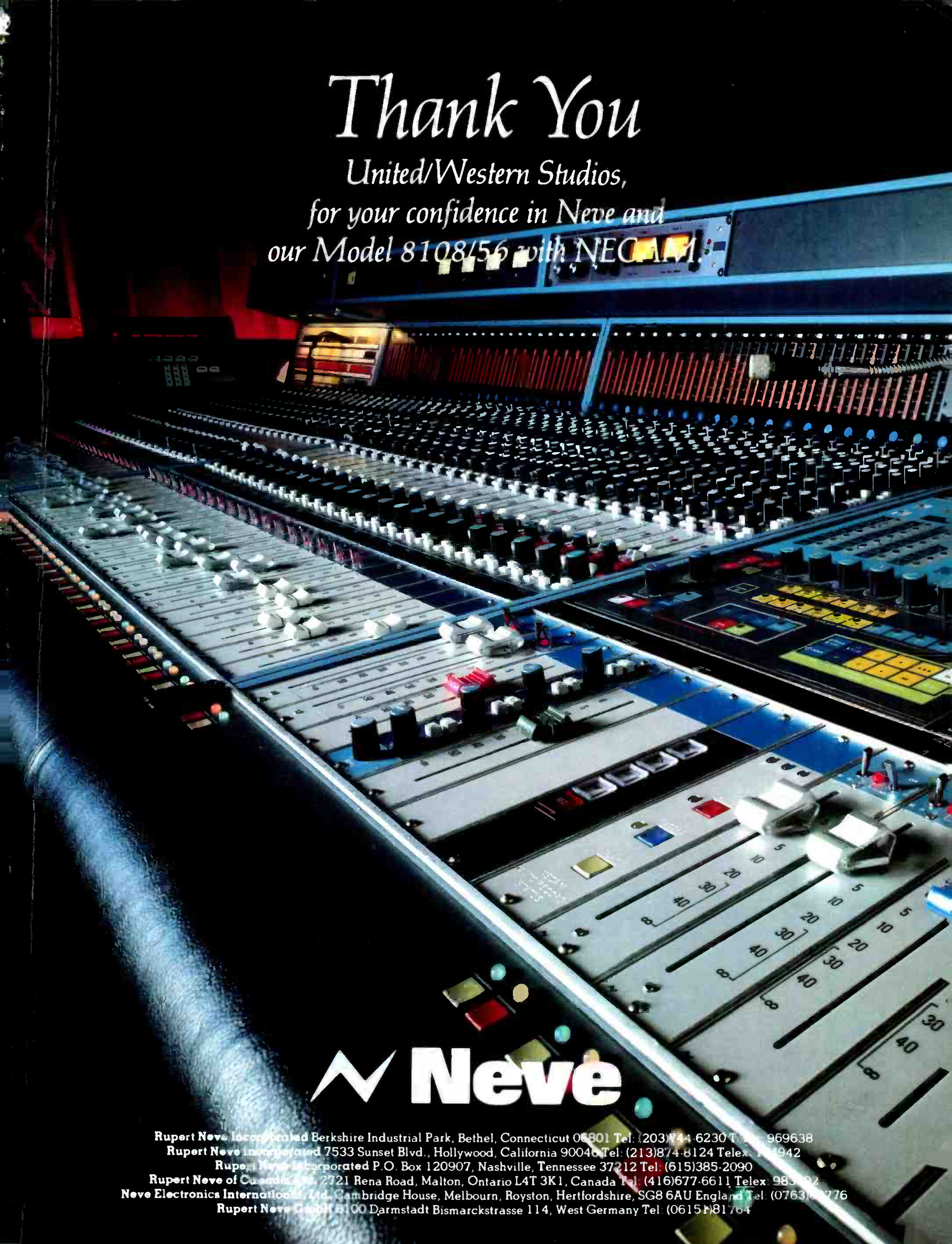
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for your confidence in Neve and
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5
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— the magazine to exclusively serve the **RECORDING STUDIO** and **CONCERT SOUND** markets . . . all those whose work involves the recording or producing of commercially marketable audio.

— the magazine produced to relate the audio **ART** . . . to the audio **SCIENCE** . . . to audio **EQUIPMENT.**

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— Contents —

— The Cover —
By **Trici Venola** — Her collage of concepts of graduate student designers at the Philadelphia College of Art, and Los Angeles' Art Center College of Design, of the study sponsored by **Quad-Eight.**

How To Recognize a Sierra Audio Studio:

It's busy
People like Sierra Audio rooms. On six continents of the world the leading studios in each market are of Sierra/Hidley design. Artists are keenly aware of the number of hits produced in Sierra rooms.

It's accurate
Producers and artists request the Sierra SM III Monitor System. Our new two-way system dramatically out-performs older three and four way concepts, and extends the range past 20 kHz.

It's beautiful
Sierra's designers and decorators know what creative people like, what makes them comfortable, and what makes them come back.

It's state of the art
Research in the art and science of studio design is continuous at Sierra. Our experience is greater. We've built more rooms. What we know today won't be learned by other designers for a couple of years yet.

It's successful
Of 245 studios the people at Sierra have designed over the last ten years, 245 are still in business. Every single one.

KITTY RECORDS
Tokyo (2 studios)
RECORD PLANT
Los Angeles (Stage Studio C)
SOUNDS INTERCHANGE
Toronto (Overdub and Mix)
STUDIO SIX
Montreal (Control Room)
ARMSTRONG AUDIO/VIDEO
Melborne, Australia (1 studio)
CINEMA AUDIO
Manila (2 studios)
PIERCE ARROW STUDIOS
Chicago (1 studio)
JEM RECORDS
Manila (2 studios)
MASTER SOUND
Atlanta (1 studio)
CBS/SONY
Roppongi, Tokyo (2 studios)
BIAS RECORDING
Virginia (3 studios)
KUANGCHI PROGRAM SERVICES
Taipei (1 studio)
PARADISE STUDIOS
Sydney (1 studio)
WING HANG RECORDS
Hong Kong (1 studio)
THUNDER ROAD
Calgary (2 studios)
PECAN STREET STUDIOS
Austin (2 studios, 1 cutting room)
ARTISAN SOUND RECORDERS
Hollywood (2 studios, 1 cutting room)
WAREHOUSE RECORDING
New York City (1 Control Room)
SMOOTH ROCK STUDIOS
Calgary (1 studio)
ALFA RECORDS
Tokyo (1 studio)
FANTASY RECORDS
Berkeley (Studio B)
SOUNDMIXERS
New York City (Studio B)
AUDIO INDUSTRIES CORP.
Hollywood (1 studio)
GROUP IV RECORDING
Hollywood (Control Room Monitors)
CBS/SONY
Shinanomachi, Tokyo (12 studios)
VIDTRONICS/TECHNICOLOR
Los Angeles (Video Sweetening Room)
FANTASY RECORDS
Berkeley (Studio D and new Disc room)
MR. "D's," KENDUN RECORDERS
Burbank (Super Studio)
SOUNDMIXERS
New York City (Redesign and Rebuilt Studio C)
SAUSILITO RECORD PLANT
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Los Angeles (Scoring Stage D)
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Austin (Recording Studio)
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Orem, Utah (Video Sweetening Studio)
SOUND ONE FILM CORP.
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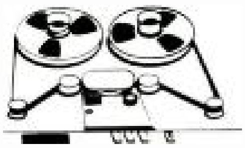

from: **David Starobin,**
President
Star Recording, Inc.
Philadelphia, PA

I read with interest F. Alton Everest's article *(R-e/p, June 1980)* on quiet air conditioning for the studio. Of equal importance to the low noise level of the system is its ability to maintain a stable temperature and humidity environment. A too hot control room increases the noise floor and eventually leads to component failure. High humidity affects tape handling and storage. A cold blast of A/C in the studio will knock the piano, strings, horns, etc., out of tune rather quickly.

I have found some useful information I'd like to pass along to other studio owners and designers in "NASA Tech Briefs," a government publication providing to the public dissemination of information resulting from the efforts of NASA. NASA research has resulted in products familiar in the studio: Ferrofluids, high-density circuit boards, a missle nose-cone material utilized as the body of Ovation guitars, etc.

A new system, developed by Henry D. Obler, of Goddard Space Flight Center, is described in the Fall 1979 NASA Tech Briefs. It is a design for an air conditioning system for environmentally controlled areas containing sensitive equipment and regulates temperature and humidity without wasteful and costly reheating. This invention is owned by NASA and a patent application has been filed. Information on this No-Reheat Air Conditioning system can be obtained by requesting Technical Support Package #25 from: Director, Technology Transfer Division, P. O. Box 8757, Baltimore/Washington International Airport, MD 21240.

Another useful bit of info: We're all supposed to be observing the President's Emergency Building Temperature Restrictions to conserve energy. Variances are obtainable and it can't hurt to put one up on the wall of the control room. A variance to the Certificate of Building Compliance can be obtained from your local congressman, the applicable category being "Manufacturer's warranty," meaning that most recording equipment components must be in a 70° free-flow environment to operate as specified.

from: **Steven Cavanaugh,**
WUOM Supporting Staff
Ann Arbor, MI

In his reply to my letter which appeared in this column in June, Mr. Hunt, of JVC Cutting Center, Inc., appears to have misunderstood my comments. He is under the impression that I was describing the "fixed pitch" method of disc cutting. Actually, I was describing the new method introduced on the CBS discomputer. With this system the cutting stylus does not advance *at all* for a large part of each revolution. Then at the appropriate time (which seems, from observation of playback, to be from once to three times per revolution) the stylus "kicks" forward by the distance determined by the computer. This is in contrast to ordinary variable pitch systems (such as is used in the Neumann VMS-70 lathe). With such systems the groove pitch is updated four times per revolution (or more), but the groove is *always* advancing, and the changes in pitch are implemented smoothly.

The jerk-like motion of the CBS system can be observed by playing any disc which has been cut at the CBS mastering room within the last two or three years. Other companies have also adopted the system, or similarly operating systems. The jerk causes admittedly minor problems, but I feel they are significant enough to bear mentioning, and no one else seems to have done so.

and, from:
Sequoyah Bear
(ed: Honest, folks!)
Helper, UT

I wish to speak out against the creeping-meatball tendency in the biz to use the incorrect spelling "buss" for the word "bus."

I rave from some sort of position of authority, being after all the Punctuation Editor of the Picture Paper (by moonlight).

Buss means (check yer dictionary) kiss.

Bus means mass transit, whether of people or electrons or anything else.

It's strange how a new piece of misinformation wriggles its way into the minds of these humans. Seems like (1) people realize that there is a word "buss," and most of the public don't know what it means, but they seem to have a place for it reserved in there mindspace. (2) There is a difference between the bus that is concerned with people and the one that is concerned with electrons. Therefore (3): "buss" appears to fit into the space which actually belongs to a very old word, much older than "bus" which is a contraction of "omnibus." The "us" part is the standard Latinoid ending, and the "b" can perhaps in this instance best be called an insulator, to keep the two vowels from shorting together. (Latin: dative plural of omnis, all.)

– continued overleaf . . .

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Select your multitrack recorder as carefully as the other facets of your studio. Select the Ampex MM-1200. Because it's the one multitrack recorder that can go through every change your studio goes through on its way to greatness. And still be as economical and easy to operate as the first great day you got it.

Growth without growing pains. With the MM-1200, you'll seldom be faced with a situation you can't solve. Because the MM-1200 comes prewired to make upgrading from 8 to 16, or 16 to 24-track operation simple and swift. And if adding channels won't solve your problem, the MM-1200's versatility will. Mastering, live sound reinforcement, double system sound, video sweetening or film and TV production/post production are all jobs that the MM-1200 can handle. Built-in single point search-to-cue, elevated record level capability, 16" reel capacity and fast start times also help you grow.

Performance you can depend on. The MM-1200 has proven itself under some of the most adverse conditions. The massive, stable top plate comes aligned and stays aligned . . . through repeated sessions in the comforts of the studio, or on remote locations.

Ampex keeps your options open. The list of optional accessories for the MM-1200 is the longest in the business. You can add multi-point search-to-cue and store 20 cue locations. This time-saving tape handling accessory provides tape time readout, cue point readout, "on-the-fly" cueing and more. Other accessories include the PURC™ Record Insert Controller, Search-To-Cue Remote Control, and MSQ-100 Synchronizer for jobs that require more than 24 tracks. Contact your Ampex sales representative for complete details.

AMPEX MAKES IT EXCITING

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for additional information circle no. 3

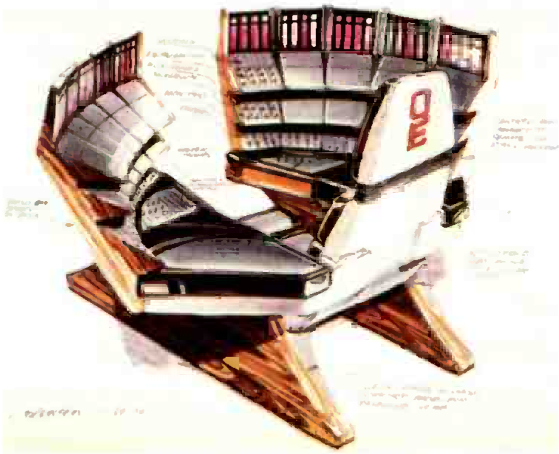
views

One view of where the Audio Control Console has been and where it might go in the '80s . . .

LOOKING AHEAD . . . BY LOOKING BEHIND!

By Dave Hadler & Ron Neilson

The audio mixing console in today's large multitrack production settings has rapidly evolved into a highly complex, microprocessor controlled instrumentation control panel. Behind one of today's big boards, it's easy to fantasize yourself at the top of the senority ladder at United Airlines . . . pulling down 120 grand a year. With all the readouts and light meters dancing their rhythms in the subdued control room ambience, you might just lean over to co-pilot Chewbaca and ask him to prepare the ship for hyperspace drive. Slide your consciousness back a few dB in a more down-to-earth fantasy, and you now at least have the power to avert (or create) a nuclear meltdown. At the very least, you're involved in a business that sometimes may seem as though it approaches these scenarios. The analogies can become fun, and this writer has seen more than one version of a jokingly placed modification to the console's nameplate which reads something like: *Flight Control - Emergency Oxygen!*



— the work of designer Lorensen, Art Center, L.A.

Although analogies to highly complex and technically sophisticated hardware abound, it's not in jest that many designers of audio control consoles haven't yet applied some of the simplification principles germain to their avionic or process control brethren. But, let's not jump too far ahead of ourselves. To place a little clearer perspective on today's audio consoles, let's go back just ten years. We'll assume that the reader knows that the audio board had its roots with rotary attenuators in the early days of broadcast.

It's 1970 and things for the audio engineer and his co-pilot, the producer or "A n' R man" were a lot simpler, and, perhaps a little more exciting, too. The audio board was essentially a handcrafted, painstakingly wired, one-of-a-kind creation. Many a studio still built their own. Many had to. There thrived a handful of a little larger than garage-sized businesses that strung together the components: card amps, summing amps, networks, equalizers, meters, etc. Once all the parts from the components' manufacturers were gathered, the systems engineer then began the black magic of stringing all of them together, and then trying to figure out how to make the "system" work. In England, where a producer named George Martin guided his charge through multiple generations of clever four-track "Lucy In The Skies," the pressure was now on every engineer to add a few more windings to the headstack; a wider piece of tape that could accommodate more flexibility in music production; and therefore, the real beginnings of today's super console — more control, more inputs, more, more, more.

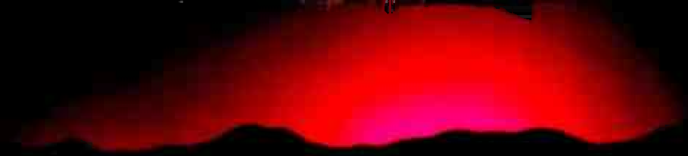
Meanwhile, back in the States, during this early decade, the record companies were seeking hordes of new talent, and the resultant stream of dollars from the baby boom's insatiable appetite for $3.98 LPs. It seemed to many a record company executive that there wasn't enough time or space in many of their own recording facilities to record the "music boom" product that would keep the pipelines flowing. So, almost unwittingly, as we will see later, the budget faucets were turned on and all it took besides beads, a guitar, and a song about people getting together, was a place to record it — and eleven others like it. Hence, came the primary business motivation for the renaissance of the independent recording studio. They had always existed before, but many were only specialized facilities that did just location work, demos, or mastering. As the "indy" studios popped-up everywhere and the record companies expanded their production operations, all the necessary goodies that went into creating a product that would elevate and proselytize the alternatives for a generation of music-"crazed" kids proliferated.

The new music entrepreneurs were given the green light and the green wallets to disappear for awhile (not too long) and then emerge from some new and obscure independent recording studio with the next hit record. And emerge they did. Many a staffer (read: record company engineer or network engineer) yearned for something better than the big company sinecure and floated away from the mother ship to set up shop to cater to the 24-hour-a-day music machine. And remember, most of those hours were billable to the production budgets of the sponsoring record company . . . even if the group was asleep, the lead singer couldn't get it together, or the one song that was ready and the others that weren't needed rehearsing.

What's all this got to do with consoles? A recap of the chronology: Lots of talent and money needs the engineer/producer and the facilities. And, of course, they both need the hardware.

Crash project comes up from producer/friend. Place the equipment orders now because it takes 3 to 6 months to build one of those big consoles that will handle the new requirements for . . . 16 tracks. Yes. The tape

. . . continued on page 18 —



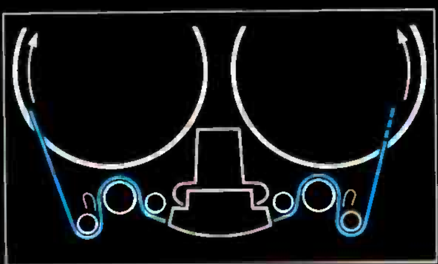
Why The MTR-90

When you're buying studio time, it can save you a bundle. If you're selling the time to your clients it can make you a bundle.
It's the MTR-90. The 16/24 channel professional recorder that is the state of the analog art. It's the new machine that outpaces the big names. And, you know who they are.

Here's why we're so confident:

Superior Tape Handling

Especially critical for wide-width tape, the Otari Optimal Tape Guidance system was the industry's first three motor, pinchrollerless two-inch tape transport; a system so superior to conventional pinchroller designs that it is also utilized on a competitive machine costing twice as much



money. The MTR-90 treats the important ferric oxide tape coating like a precious metal. Compared to conventional designs, the sonic "shine" and brilliance of a master recording stays on the tape. Smooth, even tape packs in all operating modes are the rule, not the exception.

Award-winning recording engineer Phil Seretti, owner of the one-hundredth MTR-90 and producer Janja Vujovich

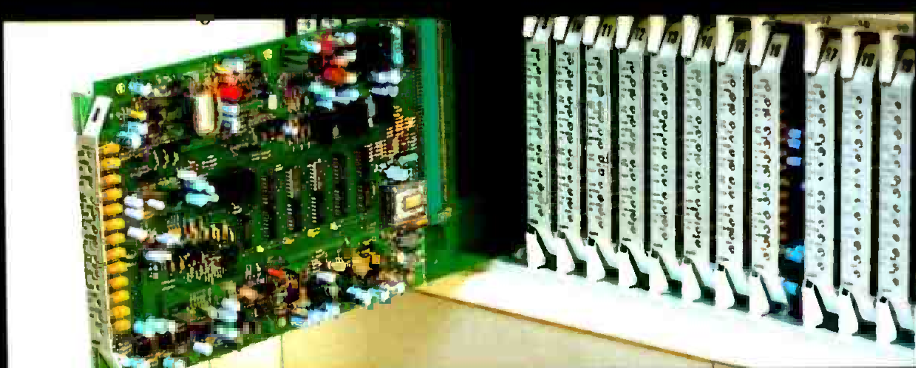
Advanced Audio and Control Circuitry

Easily accessible single card electronics not only save money, but reduce the complexity and problems of interconnection failures. Active mixing of audio and bias minimizes ringing and set-up difficulties. There's a transformerless playback amp for optimum transient response and dynamic range (greater than 71dB, 24-track: 30 ips, unweighted 30 Hz-18 kHz @ 1040 nWb/m). High slew-rate components in critical signal stages give you better aural results: Distortion-less than 0.5% at 1 kHz(250 nWb/m), Output: +28dBm.

Punch-ins and outs are totally transparent and effortless due to an integral digital timing section on each audio card that precisely ramps the erase and bias currents to yield "gapless" performance, Transport logic is digitally controlled for reliability and ease of servicing. There's a master crystal controlled reference clock for capstan, counter, record timing, bias and erase signals. Easy, rear-panel access to time-base functions facilitate SMPTE interface.

Easier To Maintain

The V.U. meter panel is hinged to give you wide open access. Remove the side panels, open the electronics bay doors and there is nothing you can't get to: power supply, master bias level, playback equalizations, motor drive amps, capstan and reel motors. The MTR-90 is designed for the real world of studios — where routine maintenance and care shouldn't have to be a headache.



Single card audio electronics contains record, reproduce, synchronous and timing circuits

Outruns The Herd.

The Extra Thought

The MTR-90 makes sessions go smoother because we also designed-in such features as: a VSO that offers ±20% speed variation with 0.1% resolution; a precision, continuously variable edit and cue control; and whenever you go into start/stop or fast wind, a circuit automatically mutes playback level —

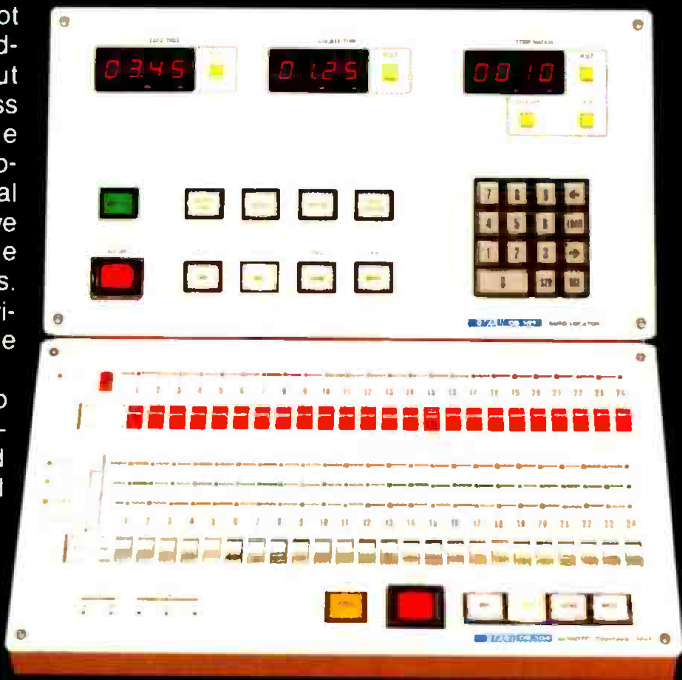
Easier maintenance with convenient access to all internal components

a feature that's designed to reduce ear fatigue and save your monitors. Every MTR-90 also comes with the industry's most advanced remote session controller. And, when you need the benefits of a sophisticated ten position memory locator, just plug-in to the back of the MTR-90 and place the optional locator atop the companion session controller's convenient roll-around pedestal.

If you've gotten the impression that it's advanced... good. If you also have any questions about how it will hold up to professional requirements, then let us assure you that we are prepared to stake our 16 year reputation for unparalleled reliability on the MTR-90. We've earned our place among audio professionals by competing with the best you've come to expect. Competing not only on features and advanced technology, but also with the ruggedness and essential service back-up so crucial to a professional product's total acceptance. After all, we do know that you make your living on our products. And we take that very seriously — because we make our living from you.

If you're moving up to a better, larger format machine, or moving the old one to the side, you need to get acquainted with the MTR-90. The New Workhorse. You'll find out for yourself why it outruns the herd. Just contact any of these fully committed dealers. Then arrange for a demonstration at your studios. The MTR-90 will give you every reason to consider that if you buy something else, you just might be buying something less... for more.

Watch for Otari's MTR-10 Series. They're the companion ¼" and ½" mastering machines that join the MTR-90 professional recorder.



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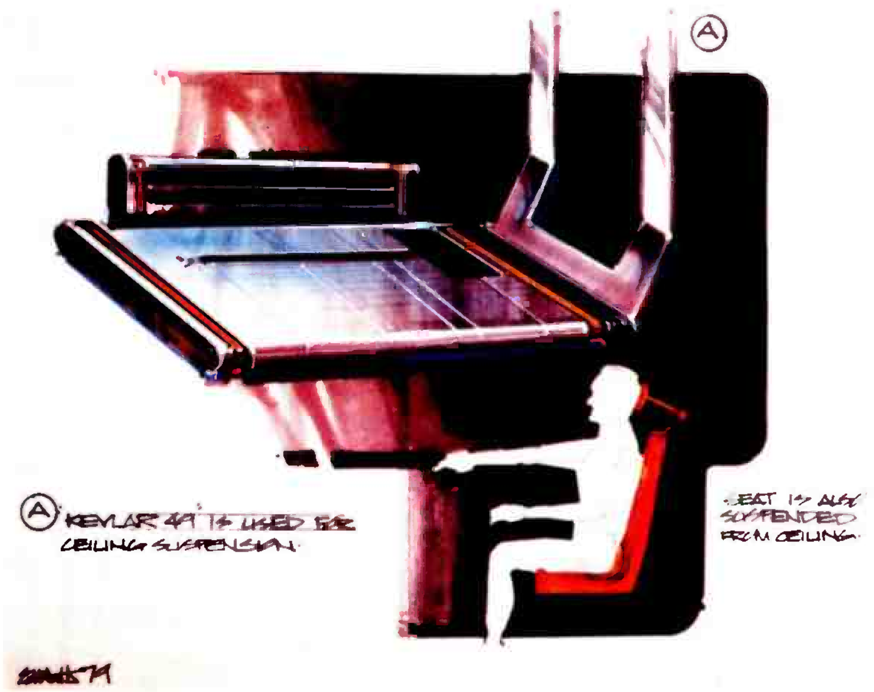
for additional information circle no. 145

ONE VIEW OF HOW THE AUDIO CONTROL CONSOLE MAY EVOLVE IN THE '80s



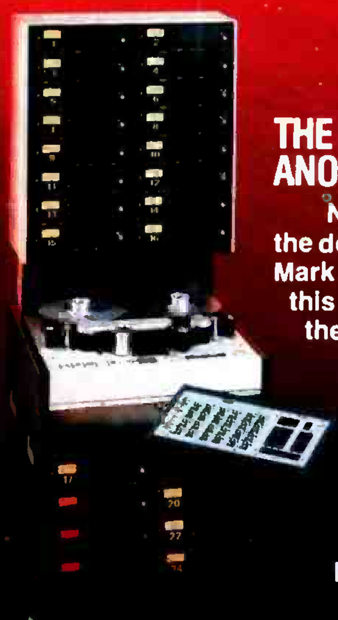
— the work of designer Ed Watts, Art Center, L.A.

boys figure they can move the slitters on their tape manufacturing machinery and repackage master tape for a new generation of mastering machines. In this case the tape boys are the same as the machine boys. We've moved a few years ahead now, and the royalty payments and concert gigs are getting pretty lucrative for the names we all know. Accountants agree that tax-sheltered artist/producer studios are a wonderful way to hide close to a million depreciating self-investment bucks. The studio designers and recording equipment manufacturers couldn't agree more. "Yes, I'll take one of those 32-input babys, a couple dozen mikes, three two-inchers (one spare), a jacuzzi, a couple pinball machines, and an investment tax credit."

By this time many of the "custom" console builders have either bitten the dust, gone into other businesses, or adjusted to the vast need for all the huge new boards that the studios had to have. Even some of the record companies bit the bullet and were forced to stay competitive and re-equip.

Enter the manufactured console. A bullish market created an economic climate necessary for companies that could manufacture and deliver a decent product — a feature-laden product that could cause them to flourish, assuming these new console manufacturers could stay competitive and up-to-date with the ever increasing spin-off technologies coming out of the computer business. The race was on for the company that could pack as much as possible into the boards. Boards that would now have to handle, (on separate inputs) such things as cowbells left (separate track), cowbells a semitone higher (separate track), and fourteen mikes on drums (separate tracks). Phil Spector donned his "Back To Mono" button, people laughed — but surely no one took that seriously! Everyone needed more tracks due to overdubbing technique.

Producers, artists, and ENGINEERS were now confronted with an unprecedented degree of "artistic control" over three chord



STUDER A80VU Mk III
Shown with 20-address memory autolocator and channel remote

THE A80VU MARK III... ANOTHER UNAPPROACHABLE STUDER!

No other multitrack recording system has ever neared the degree of perfection attained by STUDER, and the new A80VU Mark III represents one more milestone in the decade-long production of this classic machine. As always, STUDER sets and maintains the standard for the industry.

This newest A80 from STUDER has been outfitted with a completely redesigned, "close proximity" headblock for virtually undetectable drop-in/drop-out operations and totally reliable performance in the most demanding situations. Actually, the repositioning of the erase/record/playback heads is one of the few modifications made since the A80's inception, the original design being so advanced that a revision would be meaningless.

The new A80VU Mark III system includes a 20-address memory, microprocessor controlled, programmable autolocator with channel remote, which enables all tape deck functions to be operated within easy reach of the engineer at console position. The autolocator remote control is packaged as a fully-adjustable, freestanding unit.

As with all A80 models, the Mark III functions flawlessly and with satin smoothness—a machine worthy of its name and the absolute confidence professionals place in it.

for additional information circle no. 6

STUDER REVOX

Studer Revox America, Inc./1425 Elm Hill Pike, Nashville, TN 37210/(615) 254-5651
Offices: Los Angeles (213) 780-4234/New York (212) 255-4462/Canada: Studer Revox Canada, Ltd.

songs, and variations thereof. Because every player involved in a session now wanted to hear his riffs on his own headphones, complex monitoring facilities became de riguer. In mixdown, two echo sends would no longer do, so console engineers re-routed, re-amplified, and re-sharpened their pencils on the bills of materials for four and then eight. In the short span of four years (we're now about mid-seventies) the stock console all but replaced its custom predecessor. It also grew several feet. Independent monitoring sections became laced between the basic recording function inputs in most consoles in order to consolidate the increasing number of input modules. Meters now ran the entire length of the board. Soon, these super boards and their droning slave tape machines became the individual characterization of the studio. *The reputation of the studio and its staff nearly rested upon the reputation of its hardware.* Consoles and machines surely were not the only gauge of a studio's hipness. All of the creativity would certainly be met with the console engineer's endless creativity on schematic drawings.

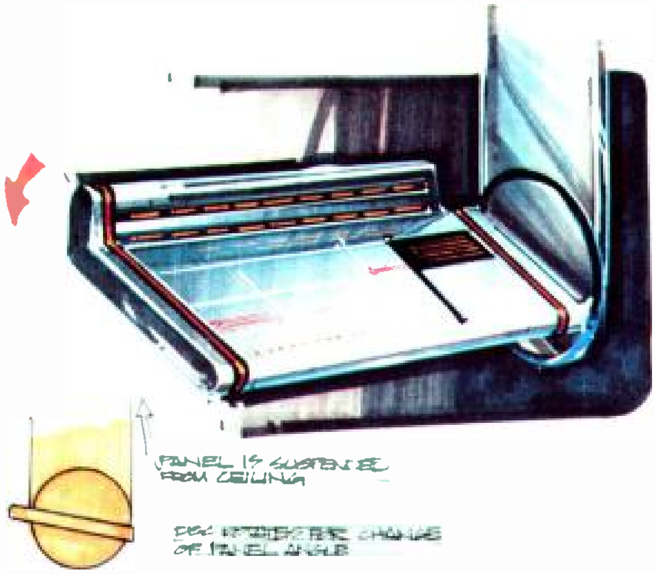
By the mid-seventies, the recording engineer now had accomplices to every creative act. For it took several hands (and ears) to twiddle the EQ, punch-in the delay line, fade the vocals, and watch all the peak overload indicators at the same time. Never mind the group in the studio. They had long since laid down the basics and were off arguing with their manager and the record company why they needed an additional 30K to finish the album. And they usually got it.

Because the intensive creative concentration, now typical of many record productions, mandated control over literally hundreds of elements, there had to be a better solution for enterprising engineering talent — enter the automated console and its instrument of feasability — the Voltage Controlled Attenuator (VCA).

Right around 16 track conversion time, and back in Newton, Massachusetts, (near M.I.T.) and also right in the heart of Hollywood, came simultaneous developments that would eventually cross paths. The newly discovered VCA could now be put into the signal chain of the audio console, and a DC voltage could now control an AC signal. Magic!

It wasn't long before consoles appeared with this magical device, patented and subsequently sold to practically every console manufacturer. A VCA allowed masterful control of many input sources via attendant sub-master faders that didn't degrade the signal. There came a firm from Canada, whose name is the same as the fruit that's placed in a martini, which caused a revolutionary "shock wave" throughout the recording industry. This company was first to incorporate an early variation of the VCA in their console (among many other advanced features). Unfortunately, due to undercapitalization and an inabillity to control their new-found panacea, their products found their way into such obscurity as sitting in one prestigious studio's hallway in the mountains of Colorado. One must look back though and acknowledge a bold and visionary attempt to change the ways of conventional audio console thinking.

While still in its infancy, the VCA found its



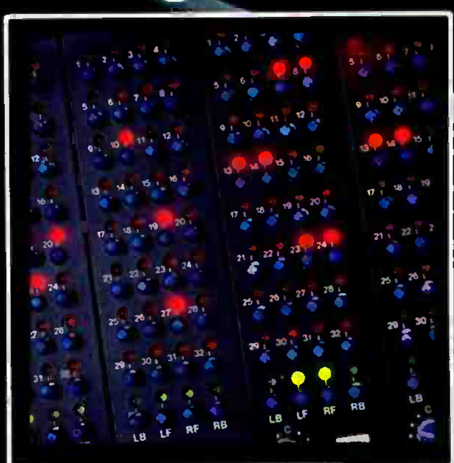
— the work of designer Ed Watts, Art Center, L.A.

. . . continued on page 25

How do you define a "fully programmable" mixing console?

(Hint: There is only one correct answer)

TOTAL RECALL™
from Solid State Logic



Total Recall™

from Solid State Logic

ONE VIEW OF HOW THE AUDIO CONTROL CONSOLE MAY EVOLVE IN THE '80s

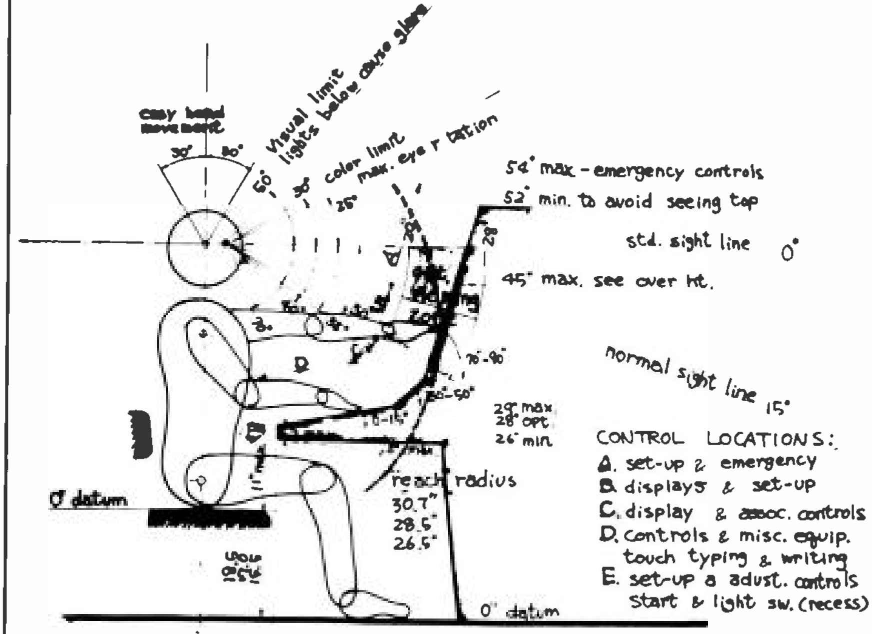
way in the first add-on application of automation: COMPUMIX, by Quad-Eight Electronics. By routing the audio through a portable controller and associated rack of electronics — the recording industry's *first* application of digital audio control came into being. By this time, engineers had discovered that it was a *theoretically* (emphasized) simple task to apply digital computer theory to converting the DC control signal of the VCA to a digital code. Just throw a compatible timing code with it (like SMPTE) and you have a repeatable, "hands-off" and perfectly synchronized elastic memory on a given mix. Without redundant passes to the audio master, and the laborious repeat mixes with all their robotic attendant moves, a producer could re-appear joyously in the studio the next night and drop the abrasive guitar solo 3 dB in the first two bars of the bridge without re-doing the entire mix. Wow! This would save studio time!

What happened was this new facility would increase the numbers of possible mixes so that all the saved time got used up anyway. Automation could allow the most perfected product; it could do almost impossible mixing tasks.

As the seventies rolled on, so did the suppliers to the hardware requirements of the studios. Two-inch tape could now hold 24 separate channels, improved tape formulations and noise reduction gear would remove further limitations on quality of reproduction through the media. And sure enough, an obscure (at least to the record industry) company figured out a way to positively synchronize two tape machines. So 8 became 16, 16 begat 32, and you have it — 24 begets 48 — less the sync tracks. There's a fortune waiting for the man who can synchronize without taking two of those precious 48 tracks. Now, in overview, which is what this article is really about, this all might seem a little ridiculous. The story may sound a little reminiscent to the Detroit automobile executive who curiously picks up his son's copy of this magazine. The tune will sound the same . . . Maybe if we add a little more chrome, our sales will increase? How about a 650 HP Turbo-ram fueled, 3 MPG convertible. I'll have to present this to my directors next meeting . . .

Albeit, the high cost of money aside, and the spigots of the record companies turned down to a trickle relative to the mid-late seventies, the cost for the latest recording console completely outfitted with all the goodies now expected has gotten drastically expensive, and drastically complex. Which raises the question to subsequent statements I've heard of late regarding the recording engineer's function in today's studios: Does one need a degree to run this board? What with log sheets that cue the *programmer* (read: engineer) as to what was programmed via keyboard, and the concern for passive data incidental to everyone but the bookkeeper — somewhere in the programming session of the new consoles, sometime between playing blackjack on the console's CRT, will come room to actually hear and try to work with the music that's still

ADULT MALE SEATED AT CONSOLE (Vertical)



Anatomical Facility Study by Koichi Tanaka

going down on the compatible old magnetic tape format. Roll-in the digital machines! Disregarding the early reports of lessened arm muscle response and the curious unwritten chapter to this investigation, the recording console still looms as getting bigger (and better) in the years to come. Experts think that soon there may be no audio at all in an "audio" console. It's just ones and zeros, from microphone in to line out.

Without further discourse regarding the optimisims and justification of video production in the recording studio, let us say that the future holds a commentary for the engineering of consoles in two words: more controls.

'Way back when, the first consoles for multitrack recording were almost always designed by the engineer who would operate them. As previously stated, in the "custom" days, he at least had a cursory knowledge of where to kick it when things went wrong. Today, the number of design decisions that go into the complete engineering of a console system are multi-disciplinary. There's audio, mechanical, graphic, digital, and last but not least, ergonomic decisions to be made. The engineer today is faced with a multiplicity of layouts to the configurations of the console's controls. An engineer now has to consider his arm span, number of fingers, speed of the bearings in his chair, and more importantly his capacity to scan high-density knob configurations that often require split-second decisions. At least that hasn't changed. Yes, he often shares these decisions with others. But, the increasingly complex monster that sits in front of the producer and artist becomes increasingly more intimidating for interaction. Thus, the engineer is now being relied upon to a higher degree for his specific knowledge of the in's n' outs of the new technological processes. A real "pilot" as opposed to being a driver.

Late last year, engineering personnel at Quad-Eight embarked upon a program to re-examine the basics of the console design. They decided to sponsor a project with the country's leading schools in industrial design. The subject: a recording console. The schools: Philadelphia College of Art, and The Art Center College of Design, in Los Angeles. The students: senior graduates who also were involved in interesting design projects for such firms as Texas Instruments, General Motors, Olivetti, etc. The two design teams were given slightly different project parameters so as to broaden the perspective of possible unique visions that a man/machine interface might take. In simple language, the console was termed an industrial controller — much like a telephone switchboard. No serious discussions regarding the internal functions of the console were given to the design teams, other than, of course, the descriptive functions of the processes necessary to accomplish a recording session, mixdown, and control of various external facilities. So as to avoid manufacturing myopia, further input was given to the students by acoustical designers — Messrs. Jeff Cooper and studio owners Kent Duncan and John Storyk. Recording engineers Brian Ingoldsby, Danny Starobin, Steve Fouce, and Jeff Sykes rounded out the practical stuff. After an indoctrination in basics, the design students were to consider the following factors in attempting to design a new console from scratch:

a. Multiple operations
b. Multiple function controls
c. Static vs. dynamic controls
d. Status display
e. Back-up capability (memory)
f. Expandable function capability
g. Acoustical and environmental factors

What was achieved at the project's end was an exciting glimpse of where the future recording console might be. New materials emerged. New colors emerged. Brand new shapes emerged. Engineers could operate controls with their feet, too — just like an organ console.

In review of the project, many striking and significant problems came to light. Whereas

. . . continued on page 28 —

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the XL-305
by

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- Totally new design approach
- The sound of a live acoustic chamber
- Natural sound, even on percussion
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The Master Room XL-305 is a totally new design approach in reverberation technology. For the first time, the qualities and properties of a live acoustic chamber are available in a rack mount unit at an affordable price. There is a natural sound on percussion, as well as voices and all other musical instruments. This quality has not been obtainable from other compact reverberation devices. The XL-305 exhibits no unwanted side effects; it's as natural as a live chamber itself.

To hear this new advancement in reverberation, see your professional audio dealer and ask for a demonstration of this exciting new unit. Hear the XL-305 "Acoustic Chamber Synthesizer" for yourself, and you too will agree . . . It's INCREDIBLE.

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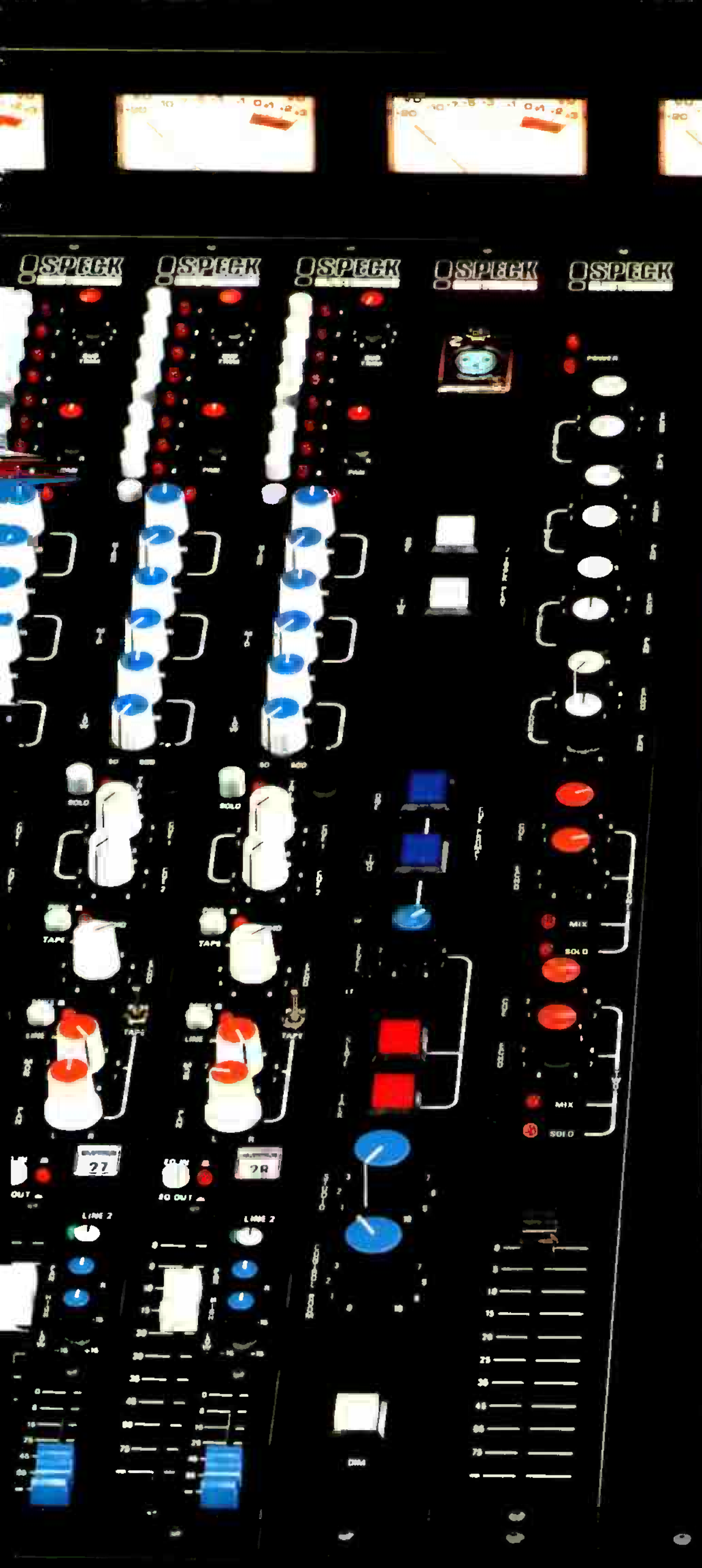

JUST FOR THE RECORD

One of the basic reasons you'll ever require more input modules for your console than tracks on your multitrack recorder is for effects and external processing equipment.

Each input/output module in our "D" series console provides the engineer with two complete line inputs. One for the tape track and one for the effect. This is made possible with a second line input that gives you slide fader level control, equalization, pan, and channel mute. So during a mix, when an effect is returned to the I/O module, it's returned to the second input section on the same module it was sent from. This keeps you and all that audio very well organized.

THE "D" SERIES RECORDING CONSOLE...

...is available with 16, 24, or 28 I/O modules. That's 32, 48, or 56 line inputs respectively, but the mainframe and patchbay for all input configurations are wired standard for 56 line inputs, 32 track tape operation (or two 16 track recorders), and all console connections are accessed via 8 high density gold pin connector blocks.

Please call or write for further details and brochure about the "D" series recording console.

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ONE VIEW OF HOW THE AUDIO CONTROL CONSOLE MAY EVOLVE IN THE '80s

the acoustic environment limitations produced only some minor changes, human engineering aspects were dramatic and totally unique. For example, the density of knob functions that could be scanned and subsequently addressed and changed were a monumental limitation and would positively be a prime candidate for Quad-Eight engineers to address new engineering principles. Color, graphics, and status display could play a much more significant role than the accepted black or brown-faced products. The straight-line module design was folded about in several dimensions to produce different human factor responses. The photographs throughout this article clearly illuminate the scope of response to the above items.

It became increasingly evident throughout the Quad-Eight design project that it is no longer accurate to presume that a recording engineer must design his console. Getting back to our opening analogies, does an astronaut design his space capsule? As the number of technologies that are utilized increases, the designers responsible for this ultimately "artistic product" must keep their heads clear. Although the industrial design student's team viewpoint was not wholly valid for various reasons, many of the product development innovations that came to light are immediately adaptable. Depending upon the growth and competitiveness of the industry, the Quad-Eight engineering staff surely has an insight into that which seems missing in other areas of the industry. No doubt, they'll hold their trump cards on these interesting innovations until the climate of acceptance is ready for a dramatic change.

The future of the audio console lies in expanding interdisciplinary synergism. It will expand the product's usefullness beyond expectations. If you don't know how that tune goes, I'll hum a few bars and you can fake it. □□□

news

SOUND WORKSHOP ACQUIRES LICENSE TO BUILD TRANS-AMP

In an effort to offer the Trans-Amp LZ Low Noise Amplifying Device on a more cost-effective basis, Sound Workshop Professional Audio Products, Inc., the Hauppauge, New York-based manufacturer of professional recording mixing consoles, has been issued a license by Valley People, Inc., Nashville, Tennessee, to be the first and only company allowed to build this transformerless mike-pre. The license restricts Sound Workshop to the manufacture of the Trans-Amp LZ only for use in its own proprietory products; Sound Workshop will not market the Trans-Amp LZ as a stand-alone product.

Since 1978, Sound Workshop has been offering the Trans-Amp LZ as part of an optional microphone preamplifier in its top-of-the-line Series 1600 Recording Console. The increased manufacturing efficiency afford by its license with Valley People will enable Sound Workshop to now offer the Trans-Amp LZ as standard equipment on its new Series 40, which supercedes the Series 1600.

"We are convinced the Trans-Amp LZ mike-pre offers the aboslute state-of-the-art for our industry," stresses Michael Tapes, president of Sound Workshop. "We chose to implement the Trans-Amp LZ into our designs years ago, based on the wide acceptance Valley People had gained: the increased market value of a studio offering the Trans-Amp LZ to its clients increases the appeal of our own consoles. And we have gone to extreme lengths to extract total performance from the Trans-Amp LZ by implementing the design to its fullest. Valley People and, in particular, the Trans-Amp LZ's designer, Paul Buff, who deserves full credit for its optimization, market acceptance, and penetration."

SPARS AUDIO RECORDING CONFERENCE III THURSDAY, OCTOBER 30

The Society of Professional Audio Recording Studios (SPARS) will host Spars Audio Recording Conference III on Thursday, October 30, it has been announced by Joseph D. Tarsia, president.

Spars Audio Recording Conference III will be held at the Doral Inn (New York) and will consist of three seminars offering

news

perspectives into business, technical, and engineering aspects of recording studios:

Studio Marketing Techniques (10:00 a.m. - 1:00 p.m.): An innovative panel, led by Murray Allen (Universal Recording Corporation), will present marketing techniques that have been successfully employed to keep their studios productive — in spite of today's economy.

Technical Downtime — The Invisible Thief (2:00 p.m. - 5:00 p.m.): Robert Liftin (Regent Sound Studios) will chair a panel of respected technical experts who will delve into one of the most profit draining aspects of any recording studio — technical downtime.

Good Engineering Practice (7:00 p.m. - 10:00 p.m.): This topic will range from A to Z — Azimuth to Zenith — and will present, *for the first time*, the conclusions of a comprehensive SPARS study of recommended audio recording practices. Guy Costa (Motown/Hitsville, USA) will head a panel synonymous with good engineering practices.

Ample time will be allocated for questions and answers.

Advance registration for non-SPARS members is $75.00 per session or $150.00 for all three seminars. Students and faculty may gain admission at half price. An additional $10.00 will be charged for door admission; but SPARS cannot guarantee that door admission will be available, as registration for Spars Audio Recording Conference III is limited.

Further information and/or registration should be directed to Malcolm Pierce Rosenberg, Esq., SPARS Administrator, 215 South Broad Street (7th Floor), Philadelphia, PA 19107. (215) 735-9666.

FLANNER'S PRO AUDIO NEW MID-WEST OTARI MTR-90 REPRESENTATIVES

Flanner's Pro Audio, Milwaukee, has recently been appointed Midwest representative for the new Otari MTR-90. According to John Loeper, sales manager of Flanner's, "It is our feeling that the combination of the Otari MTR-90's and the Neotek consoles provides the demanding studio owners with a cost-effective 24- or 16-track operation without sacrificing the high demands for sound quality and versatility. The Otari MTR-90 reflects current technology with its fully symmetrical tape path and pinch roller-free direct drive capstan. The transformerless design of the Neotek Series III Recording Console offers in-line monitoring and logic controlled FET switching of console status functions."

According to Loeper, "The Otari/Neotek super system increases our ability to offer professional sales, consultation, design, and fast comprehensive service."

STUDER REVOX OPENS NEW FACILITIES IN NASHVILLE

Studer ReVox, a company world renowned for its pioneering development and manufacture of high quality audio equipment, formally opened the new facility of

Dr. Willi Studer (left) and president of Studer ReVox of America, Bruno Hochstrasser during ribbon cutting ceremonies.

its American subsidiary at 1425 Elm Hill Pike, in Nashville, on Friday, September 26. Dr. Willi Studer, the company's founder and board chairman, visited from Switzerland to emphasize the importance of his company's efforts in the U.S. market.

Dr. Studer was presented an Honorary Tennessean certificate on behalf of Governor Alexander.

dbx® DISCS ADDS TO ROSTER

A&M Records and dbx, Inc., have announced that two A&M hit albums will be released in dbx Encoded Disc format under the A&M label. The two albums are *Rise* by Herb Alpert and *Close To You*, by the Carpenters.

According to Marv Bornstein, International Vice President of Quality Control for A&M Records, "The dbx Encoded Disc format is an outstanding development for recording technology. The dynamic range of these records is almost indistinguishable from that of the master tape. The music is heard against a background of virtual silence. We are proud to release these popular albums as dbx Encoded Discs."

Performer Richard Carpenter is enthusiastic about the release of the Carpenters' album as a dbx Disc. "I heard the dbx Encoded Discs played and I was very impressed," he said. "I am delighted that *Close To You* was chosen as one of the first A&M albums to be released as a dbx Disc.

These A&M albums mark the entry of hit pop titles into the dbx library of Encoded Discs.

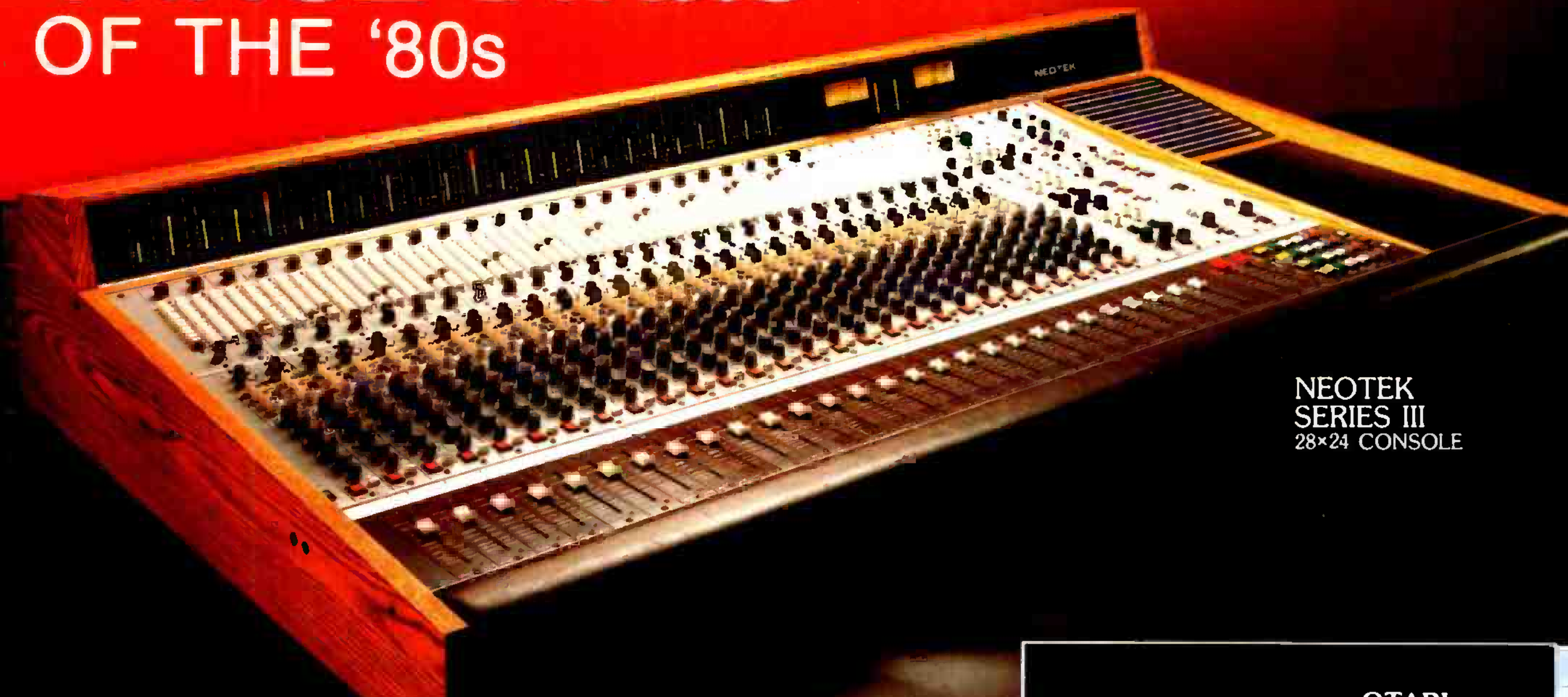
Four hit albums from Direct-Disk Labs are being released as dbx Encoded Discs, according to Jerome E. Ruzicka, vice president of dbx and director of the dbx Encoded Disc Program. The four hits being released under the Direct-Disk Labs label are: *Blood, Sweat & Tears*, *Neil Diamond: His Greatest Hits*, The Who's *Who Are You*, and Loggins & Messina's *Full Sail*.

"These releases are part of our aggressive program to expand the repertoire of the dbx Encoded Disc Library to include a wide selection of pop and rock recordings. By the end of the year, we expect to have released approximately ten albums in this popular category," said Mr. Ruzicka.

Chalfont Records' sonic spectacular, *The Empire Strikes Back*, John Williams'

News continues on page 168 —

MEET THE NEW TRACK STARS OF THE '80s



ANNOUNCING THE OTARI/NEOTEK 24-TRACK SUPER SYSTEM

With the introduction of the MTR-90 Masterecorder, Otari has made its entry into the world of professional 24-track recording. The MTR-90 reflects the leading edge of current technology with its fully symmetrical tape path and pinch roller-free direct drive capstan. The transformerless design of the Neotek Series III recording console offers in-line monitoring and logic controlled FET switching of console status functions. The Neotek/Otari Super System delivers simple and efficient operation yet affording the experienced engineer an unprecedented degree of flexibility at a surprisingly affordable price. With "hands-on" experience in the recording industry, the consultants at Flanner's Pro Audio know how to put the Neotek/Otari Super System to work for you. No matter what the size of your purchase, you'll appreciate the personalized service received before, during and after the sale. Flanner's Pro Audio has been selected to represent most of the leading names in professional audio products which means they probably have the items you want in stock for immediate delivery! Call today for your next purchase!

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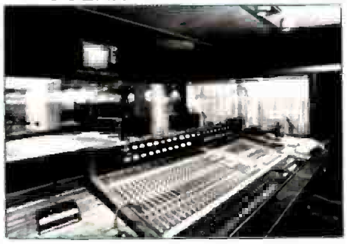
STUDIO UPDATE

Northeast:

■ **OMNI RECORDING STUDIOS** (Morristown, New Jersey) is celebrating its second anniversary with the completion of a combination vocal/drum booth. The new addition fills 80 square feet of the 1,200 square foot 8-track studio and contains eight mike inputs. Omni also features a Quantum console, Crown amps, Big Red, Little Red, Auratone monitors, and dbx noise reduction. Instruments include a Steinway grand piano and a Hammond organ with Leslie. *44 Abbett Avenue, Morristown, NJ 07960. (201) 539-8804.*

■ **OCEAN RECORDING STUDIO** (Forked River, New Jersey) has recently installed dbx compressor/limiters, Orban parametric EQ, and an Otari MX-5050 ½-track mixdown deck. Other electronics at Ocean include Orban reverb, MXR DDL, and mikes by Shure, Sennheiser, AKG, and Audio-Technica. A Hammond B-3 organ with Leslie has also been added to the studio's instrument collection. **DENNIS BOURKE** is the owner/operator of the eight-track studio. *1809 Ravine Drive, Forked River, New Jersey 08731 (609) 693-6439.*



Aura Recording, New York

■ **AURA RECORDING** (New York City) has taken delivery of a new Amek M-3000 36-input mixing console in its all new Studio D. Installation of the board was handled by Martin Audio/Video Corporation of New York, Amek's east coast distributor. The new facility also features a 24-track Ampex MM-1200 recorder with Audio Kinetics Intelocator and a 2-track Ampex ATR-102 for mastering. **ALLAN MURCHIN** is the owner of Aura.

Southeast:

■ **THE LOWER LEVEL RECORDING STUDIO** (Nashville, Tennessee) has now been in operation for a little over three months and happy to report that they are off to a great start. Lower Level is a 24-track studio with an atmosphere that is reported to be relaxed and extremely comfortable. Outboard equipment includes UREI LA-2As, LA-4As, dbx 160s, Audioarts 160 Parametrics, Lexicon Prime Time, Eventide Harmonizer II, Loft Flanger/Delay, Orange County Vocal Stressor, and Lexicon 224 Digital Reverb. Also included are MCI and Studer tape machines, Dolby, UREI Time Aligned™ monitors, and a Neotek Series 3 transformerless console. *P. O. Box 110728, Nashville, TN 37211. (615) 331-9635.*

■ **BIG MAMA RECORDING STUDIO** (Knoxville, Tennessee), located in the foothills of the Great Smokey Mountains, **NORBERT STOVALL,** owner, is proud to announce the addition of Studio B plus new gear such as an Eventide H-949 Harmonizer, DeltaLab DL-2 Acousti-Computer, limiters by UREI and Inovonics, Ashly parametrics, Prophet 5 synthesizer, Calibration Standard MDM-4, mixdown monitors, mikes by AKG, Neumann, Sennheiser, E-V, and Shure, and a completely covered glass cube which offers all the comforts of inside and all the atmosphere of outside summer and winter. Console and recorder is supplied by MCI. All outboard gear is supplied by Audio Architects, Nashville. In addition to Stovall, who also is house producer and engineer, the staff includes **ALLEN WRIGHT,** studio manager and engineer, and new addition **RON E. DOBBS,** chief engineer, who comes from Louisiana by way of Studio In The Country, and River City Recorders. Studio A is 24-track and Studio B 16-track (still under construction), both with full mixdown capabilities. Another added feature is a wood-burning fireplace which heats both studios in winter months. *400 Ensley Drive, Knoxville, Tennesses 37920. (615) 577-5597.*

■ **PYRAMID ENTERPRISES** (Lookout Mountain, Tennessee) has upgraded its studio by adding four new inputs to their Sphere Eclipse A console, bringing it to 32 in x 24 out, and by installing an ADR Scamp Pack — 2 noise gate expanders. A Lexicon 224 digital delay unit has also been added along with a new Studer computeritzed autolocator and remote control for their A-80 24-track recorder. New sideboards include an Eventide Instant Flanger. *P. O. Box 331, Lookout Mountain, TN 37350. (404) 820-2356.*

■ **dgp STUDIOS** (North Miami, Florida) announces that **GEORGE BLACKWELL** will locate his commercial production studio within the dgp facilities. The industrial and commercial production studio will have two-, four-, and eight-track capabilities and will feature dbx and Dolby noise reduction, video-sync recording, and a full complement of noise gates, compressor/limiters, harmonizers, and EQ. Time correction devices will also be offered. *1975 N.E. 149th Street, North Miami, FL 33181. (305) 940-6999.*

■ **AIRWAVES STUDIO** (Atlanta, Georgia) is expanding into a 3,000 square foot, 24-track studio in conjunction with a theater for live recording. The new studio is owned by Diversified Communications Corporation, a multi-media advertising and marketing firm. **BILL MOHR** is the resident producer/engineer. *88 Druid Circle N.E., Atlanta, GA 30307. (404) 581-0589.*

■ **TWELVE OAKS RECORDING STUDIO AND SOUNDSTAGE** (Smyrna, Georgia) has opened its new facility to serve the Atlanta market. The 27'x45' sound stage can accommodate video work, tour rehearsals, and audio/video oriented presentations. The 16-track MCI equipped recording studio was designed by **CHRIS JONES,** of Wilson Audio, utilizing UREI 813 Time-Aligned™ monitors, Crest amps, and White passive equalizers. Twelve Oaks is owned and operated by **SONNY LALLERSTEDT** and **RANDY BUGG.** *3830 South Cobb Drive, Suite 100, Smyrna, GA 30080. (404) 435-2220.*

■ **RANDY'S ROOST** (Nashville, Tennessee) has recently installed new Lockwood Major Universal monitors with a Quad 405 current dumping amplifier and Neve 20-87 custom built equalizers. The studio is also expanding its services to include record pressing and tape duplication. *RCA Building, 30 Music Square West, Nashville, TN 37203. (615) 254-8825.*

Midwest:

■ **SOUNDTREK STUDIOS** (Kansas City, Missouri) has purchased and installed an Otari MTR-90 24-track and UREI 813 monitors. The equipment was sold and installed by Flanner's Pro Audio, Inc. *Soundtrek Studios, 3727 Broadway, Kansas City, MO 64111. (816) 931-TREK.*

have you?
• increased track capacity — gone 24, 16, 8 •
• added key people • won awards •
• moved or expanded • added important equipment •
these are some of the interesting news items that can be announced in the next available issue. Write:
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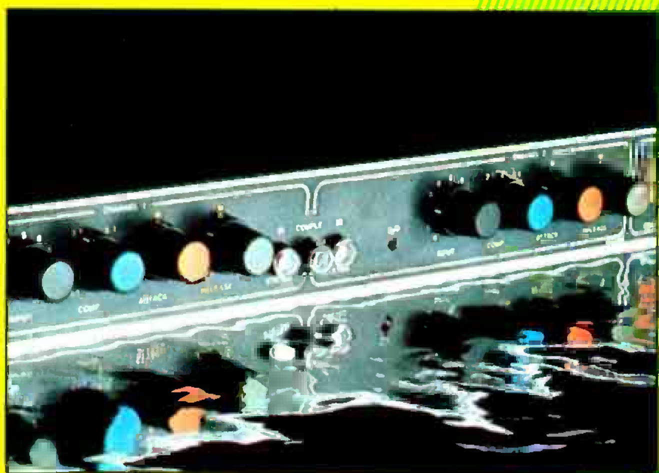
Gemini Easyrider



Controlled Signal Processing

GEMI
EASY

CONTROLLED SIGNAL PROCE

Why you need a Compressor/Limiter

First, let's consider a particular property of 'live' sound and see how it compares with the technical limitations of equipment and materials normally used in its recording and reproduction.

DYNAMIC RANGE

Each medium that can pass (or transfer) audio signals has a difference in amplitude (or loudness) ranging from where the signal is obscured by noise (is too quiet) to where it saturates the medium (is too loud). This difference is termed the medium's dynamic range and is measured in decibels (dB's).

A typical, high level, magnetic tape, for example, has a dynamic range of approx.. 67dB. (Source: 'Studio Sound', Vol 21 No 4, Apr '79). 'Live' sounds are not so restricted; emitted into free air, sound can easily have a dynamic range in excess of 120dB.

Fig 1. (Right) Illustrates some (optimistic) examples of dynamic range including those typically available to the listener.

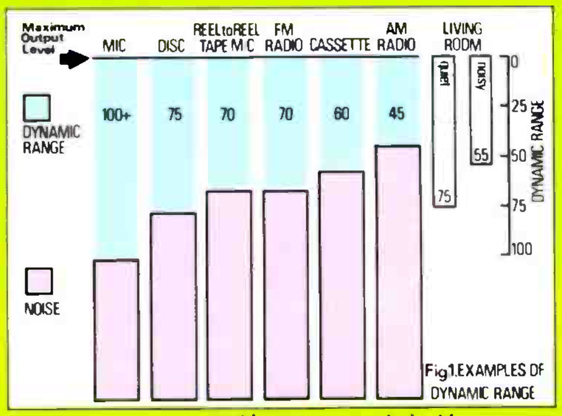


Fig 1. EXAMPLES OF DYNAMIC RANGE

MULTITRACK

Modern multitrack techniques, requiring acoustically 'dead' rooms and high input levels, can give rise to very wide dynamic range signals. On the road to presenting those signals as a final product there are two important considerations:

1) you are likely to encounter a medium whose dynamic range is more restricted
2) you may find it difficult to record sound peaks without overloading your equipment.

To help you overcome these restrictions you need a compressor/limiter which can 1) compress the dynamic range of a signal down to fit that of a more restricted medium and 2) maintain an overall maximum level (or limit).

The Gemini Easy Rider has been designed by Audio & Design (Recording) Ltd to perform these functions.

NI RIDER

ING AT YOUR FINGERTIPS...

What the easy-rider does

IN BRIEF

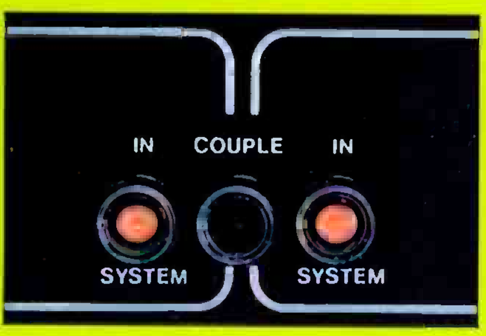
The Gemini Easyrider processes any 'line' level signal (i.e. output of pre-amp). It has two channels, which can be stereo coupled for mastering (Fig 3), each with independent input control, output preset. The compression ratio (COMP) can be varied from 1:1 to 20:1 (LIM) with semi- automatic attack and release controls (Fig 2). Gain Reduction is indicated in twin, 20 element, calibrated l.e.d. displays (Fig 4).

The threshold of compression is automatically varied to average 10dB compression under dynamic conditions, with constant output level at any ratio.

THE CONTROLS

INPUT — over 30dBs of gain for matching internal levels to your signal level with ample 'make-up' gain.

OUTPUT— presetable for 'set it and leave' drive up to +12dBm.



COMPRESSION RATIO (COMP) (Fig 2) — the ratio (or slope) of input to output after level rises above the threshold e.g. 2:1 ratio means "for every 2dBs input above threshold raise output 1dB". Continuously variable 1:1 to 20:1 (LIM).

ATTACK (Fig 2) — The reaction speed for instituting gain reduction at the set ratio after signal level exceeds threshold. The attack time is variable from 500μs (Fast) – 5ms (Slow) plus programme controlled handling of fast transients.

RELEASE (Fig 2) — controls recovery speed when signal falls below threshold. Set 4ms (Fast) to 4secs (Slow) with 'automatic' position.

STEREO MATCH (COUPLE) (Fig 3) — links the two channels to prevent 'image shift' which would otherwise occur if gain reduction is uneven (channel to channel), essential when preparing stereo masters.

The Gemini Easyrider is the latest in our range of stereo compressor limiters. It will add 'punch' to your recordings by improving signal density and 'tighten' the sound by allowing higher average modulation level on tape. In addition, you don't need any sound engineering experience to operate it.

Local Distributor

SPEC. FOR ONE CHANNEL

Frequency Response:	+0dB−1dB, 20Hz to 25kHz at threshold ref 1kHz
Noise:	Better than −82dB Ref to limit level
Condition:	Measured band limited 25Hz and 25kHz
Distortion:	@ 1kHz 0.15% Ref to +12dBm (max limit level)
Condition:	3 sec release time @ 1kHz, 10dB Gain Reduction
Clip Level:	Output stage +18dBm into 600 Ohms Input stage +18dBm
Crosstalk:	@ 10kHz −77dB. ref to +12dBu on opposite channel
Make-up Gain:	33dB
Output (Pre-set):	Calibrated −3dBm to+12dBu ,ref limit threshold
Stereo matching (worst case):	±1dB Channel to channel over 20dB gain reduction
Input impedance:	Greater than 10k @ 1kHz
Output Impedance:	Less than 1 Ohm @ 1kHz
Thresholds/Ratios:	switched 1:1 variable 1.5:1 to 20:1 thresholds automatically adjusted
Limit Attack:	Fast: 500uS for 10dB over limit threshold Slow: 5mS for 10dB over limit threshold Dynamic attack changes in relation to level
Release:	Fast: 15mS on 10dB over limit threshold Slow: 4 secs on 10dB over limit threshold Auto: 15mS on 5 Secs
Input/Output/Earthing:	via 12 way Barrier strip
Side Chain Access:	via 3 pole jack socket
Metering:	Calibrated 20 segment LED Bar graph
Power requirements:	230 VAC±7%. 115 VAC±10%, 50/60Hz selectable
Power Consumption:	15 watts
Size:	STD Rack, 1¾" x 19" x 7½" (44.45mm x 482.6mm x 190.5mm)
Weight:	5.5lb (2.5kg) Shipped in purpose-built export packing

U.S.A. Audio & Design Recording, Inc.
P.O. Box 786, Bremerton, WA 98310 U.S.A.
Tel: (206) 275 5009. Telex: 230 152426 ADRUSA

Worldwide Audio & Design (Recording) Ltd,
North Street, Reading, Berks RG1 4DA.
Tel: (0734) 53411. Telex: 848722 ADR UK

STUDIO UPDATE

Midwest (continued)

■ **TALDEK SOUND INDUSTRIES** (Newton Falls, Ohio) has announced their opening. The studio is equipped with a TEAC 80-8 with dbx, TEAC Model 5 boards, digital delay, flanger, and other outboard effects. Instruments available include an in-studio drum kit, and drum room, various guitar amps, baby grand piano, Mini-Moog, and Fender Rhodes. Mikes include Sennheiser, AKG, Shure, Electro-Voice, and Neumann. *530 Arlington Road, Newton, Falls, OH 44444. (216) 872-5719.*

■ **TECHNISONIC STUDIOS** (St. Louis, Missouri) has recently added to their 24-track Studio A complex the DDL-2 Acousti-Computer with add-on memory module just in time to put the finishing touches on **HEAD EAST'S** new album for A&M Records. Studio C, their audio/visual production room, has put a new ATR-104 in use. Technisonic custom duplicating services have been expanded with the addition of a new SMT automatic record press, bringing the total to three presses. The purchase was essential because of the rapidly expanding syndicated radio market. Records are pressed for such weekly shows as **LIVE FROM THE LONE STAR CAFE,** and **ROLLING STONE MAGAZINE ROCK REVUE.**. *1201 S. Brentwood Boulevard, St. Louis, MO 63117. (314) 727-1055.*

■ **NEW DAWN PRODUCTIONS** (Cedarburg, Wisconsin) has added a Neotek Series I 16x8 recording console for production use. The unit was sold and installed by Flanner's Pro Audio, Milwaukee, Wisconsin. *New Dawn Productions, P. O. Box 186, Cedarburg, WI 53012. (414) 377-8830.*

■ **WGUC-FM** (Cincinnati, Ohio) has moved to its new combination broadcast and recording studios in the Crosley Telecommunications Center. The two studios and two announce booths will serve both the public radio station's broadcast needs as well as offering 24-track recording facilities to the public. The installation was designed by **JOHN STORYK. JAMES STITT** is the station's director of engineering.

■ **KELLY BRYANT, INCORPORATED** (Milwaukee, Wisconsin) has taken delivery of a used Harrison console from Studio Supply/Chicago.

Bleifeld Productions

■ **MARTY BLEIFELD PRODUCTIONS** (Fort Wayne, Indiana) has completed its new 24-track studios with equipment supplied by Studio Supply/Chicago. Among the gear installed is a Sound Workshop 1600 console with trans-amp mike pre's and VCA grouping, UREI monitors and limiters, Orban parametric EQ, Ecoplate reverb, and a Studer tape machine.

■ **AUDIO SERVICES COMPANY** (Mishawaka, Indiana) has upgraded their equipment list with the addition of an MCI 24-channel console, two Otari Mark II mastering machines, dbx noise reduction, E-V PI-15-3 Sentry monitors, and TAPCO power amps. The studio is a 16-track facility. *3016 Home Street, Mishawaka, IN 46544 (219) 255-5198.*

Southwest:

■ **REELSOUND RECORDING COMPANY** (Manchaca/Austin, Texas) has just received a JH-24 MCI 24-track transformerless machine for their remote recording bus. The remote unit has recently completed dates with **TED NUGENT** for his next live album on Epic Records. Concerts were recorded in Detroit, Buffalo, Cleveland, Boston, New Haven, Pittsburgh, Rochester, and Providence. **RIC BROWDE** and **CLIFF DAVIES** were producing with **MALCOLM H. HARPER, JR.,** and **GREG KLINGINSMITH** at the console with **MASON HARLOW** assisting. *P. O. Box 280, Manchaca, TX 78652. (512) 472-3325 or 282-0713.*

■ **THIRD COAST SOUND** (Austin, Texas) has recently expanded to include an Ampex MM-1100 machine with Dolby "A." The console is a new Series 3 Neotek. Monitors are 813 UREIs and Auratones. Microphones include a full cabinet of AKG, Sennheiser, Neumann (including some interesting tube varieties), and the rest of the gamut. Third Coast has three studio rooms, and the shared availability of a 200,000 cubic foot sound stage for larger setups. Studio musical instruments are numerous, including a Yamaha Grand, Martin D-28, Fender Jazz, Hammond CV, and many more. In video thre are three RCA one-inch recorders, Grass Valley switching, and a CMX 340X editing console. The video suite is interlocked to the audio studio using BTX for sweetening. All audio in video is Dolby "A" encoded, and the video suite has JBL and Auratone monitors. A smaller Series 2 Neotek console and 4-track Ampex is also included in the video package. *501 North Inter-regional, Austin, Texas 78702. (512) 478-0019.*

Westwood, Tucson

■ **WESTWOOD RECORDING STUDIOS** (Tucson, Arizona) has expanded to a fully automated 24-track studio with the installation of an MCI JH-114 24-track recorder with AutoLocator fed by an MCI JH-636 28x28 automated console. The monitors are bi-amped JBL 4333s, and Auratones. Signal processing gear is by Allison (Valley People), UREI, Orban, dbx, and Eventide. Mikes are by Shure, Electro-Voice, Beyer, Sennheiser, and AKG. The 32' x 22' studio also offers a Baldwin grand piano, various Arp synthesizers, a Hammond B-3 organ with Leslie, and Rogers and Slingerland drum kits in and adjoining 10' x 10' drum room. Westwood also has a smaller, fully equipped production studio. The studios are owned and operated by **BILL CASHMAN, ROGER KING,** and **FRED PORTER.** *964 West Grant, Tucson, Arizona 85705 (602) 622-8012.*

Mountain Sates

■ **BOETTCHER CONCERT HALL** (Denver, Colorado) has purchased and installed a Neotek Series 1 24 x 8 console for recording and sound reinforcement. Flanner's Pro Audio, Milwaukee, Wisconsin, supplied the equipment. *1245 Champa Street, Denver, Colorado 80204. (303) 629-1534.*

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STUDIO UPDATE

Southern California

Soundcastle

■ **CAN AM RECORDING STUDIOS** (Tarzana, California) has recently commissioned Everything Audio to install a new Tannoy Buckingham monitor system in the studio. The system will be powered by Crown PSA-2 amplifiers with White 1/3-Octave Equalizers. *18730 Oxnard Street, Tarzana, California. (213) 342-2626.*

■ **SOUNDCASTLE** (Los Angeles) has just announced the installation of a new 48 x 32 Neve 8108 console, and the addition of a new Steinway concert grand piano, as well as extensively updating their outboard rack. Construction of an isolation booth with an 18-foot ceiling has also been completed. *2840 Rowena Ave., L.A. CA 90038. (213) 665-5201.*

■ **DIGITAL MAGNETICS** (Los Angeles), **ARMIN STEINER's** new facility is fully equipped with Sony digital equipment including two complete PCM systems, a BVE-500 editor, and a pair of APM-8 loudspeakers with Accurate Piston Motion drivers. A Sony DAE-1100 editor is on order for delivery this fall. Steiner is the founder of **SOUND LABS STUDIOS.** *(1800 Argyle Ave., L.A. 90028. (213) 466-3463.*

Paramount

■ **PARAMOUNT SOUND STUDIOS** (Hollywood) (Not affiliated with Paramount Pictures), has recently completed construction of its $1.2 million Studio C recording and control room facility. The control room playback system features a pair of Cerwin-Vega 189SC 18-inch Stroker™ dual spider woofers. The complete monitor system includes, in addition to Cerwin-Vega woofers, a pair of Time-Aligned™ UREI monitors that utilize a single coaxial 15-inch woofer as well as two other 15-inch woofers. Paramount's consultant, **DAN GWYNN** (Electro-Media Systems) tuned the installation with the assistance of Cerwin-Vega's **MARSHALL BUCK.** The control room features a custom 40-channel Harrison 4032C control console, triple layered tempered glass on all doors and windows for tatal acoustic isolation, and adjoining air-conditioned isolation rooms for tape decks, noise reduction, and other support equipment. *(6245 Santa Monica Boulevard, Hollywood, CA 90038. (213) 461-3717.*

■ **GOODNIGHT** (Van Nuys, California) owner **KEITH OLSEN** has opened a new 24-track studio. Built by Keith Olsen and **GORDON PERRY,** of Goodnight, Dallas, Texas, the room contains a Neve 8108 desk, Studer machines, and a full complement of effects gear. Equipment and room voicing was done through Everything Audio. *15456 Cabrito Road, Van Nuys, CA 91406. (213) 787-3722 or 787-0563.*

■ **GOPHER BAROQUE PRODUCTIONS** (Garden Grove, California) has just celebrated its one year anniversary in their new location. Reportedly one of the busiest 8-track (soon to be 16) studios in Orange County, Gopher Baroque caters to the songwriter, offering production assistance as well as free use of all studio instruments, (including Kawai grand piano, Hammond B-3, Fender Rhodes, Arp Omni, complete drum kit, etc.) The equipment, according to owners **MICHAEL MIKULKA** and **STEVE MC CLINTOCK** includes a highly modified Tascam Model 15 mixer, 80-8 recorder with dbx, MICMIX 305 Reverb, JBL speakers, Marshall Time Modulator, MXR doubler, dbx limiters, and mikes by Neumann, AKG, Electro-Voice, Sennheiser, Sony, and Shure. Gopher Baroque is actively involved in jingle production, as well as publishing, and is closely involved with several major publishers. *12202 Garnet Circle, Garden Grove, CA 92645. Studio: (714) 893-3457. Message: (714) 975-1107.*

■ **MAGIC ALEX ENTERPRISES** (Riverside, California) has upgraded to ½-inch 8-track operation and added dbx noise reduction and compressor/limiters, an Advanced Audio D-250 digital delay, Symetric SG-200 noise gates, and a Control Audio C-101 spectrum analyzer. New condenser mikes by Sennheiser and Shure have also been delivered, and the studio itself has undergone a re-design to emulate an anechoic chamber which can be tuned for specific recording applications. **CRAIG SANDELL** is the owner/engineer of Magic Alex. *5299 Noble Street, Riverside, CA 92503. (714) 689-5316.*

■ **INTERNATIONAL RECORDING** (North Hollywood, California) is putting the finishing touches on its new ultra high speed 6-channel 70 mm re-recording (dubbing) studio. Stage A features a custom designed 6-channel console providing capabilities reportedly never before available. Among its provisions are 96 inputs and 39 outputs, full 6-channel stereo grouping, 6-channel graphic equalizers, 8-channel pan pots, and the ability to generate any film format including conventional mono optical, Dolby stereo optical, 4-channel stereo, 6-channel stereo, 70 mm Dolby stereo with stereo surrounds. Backup equipment for the stage includes twenty 35 mm high speed reproducers, two 6-track high speed recorders and two 16-track Ampex recorders locked to the film chain. The studio is capable of presenting 35 mm and 70 mm release prints in any format, and is available for use as a review theater for private screenings. The projection booth is fully equipped for single and double system operation. New equipment also includes a fully operational transfer department, two cutting rooms with more to be added, and a printing department with the capability of sounding 70 mm and 35 mm release prints. A second stage will available early next year with 4-track stereo dubbing facilities. This will also be high speed with additional capabilities for automatic dialogue replacement and complete screen facilities. **BILL SCHLEGEL** or **DAN MITCHELL** should be contacted for bookings and additional information. *11128 McCormick Street, North Hollywood, CA 91601. (213) 769-6644.*

■ **SERI SYSTEMS** (Canoga Park, California) owner **PHIL SERETTI** is awaiting delivery of a new Amek Model M-2000A automated desk and an Otari MTR-90 for his audio/video facility.Everything Audio will supply the equipment and acoustical consulting. *20649 Londelius, Canoga Park, CA 91306. (213) 656-8825.*

■ **BUZZY'S RECORDING SERVICES** (Los Angeles) is installing 45 channels of dbx Model 208 noise reduction to supplement its multitrack capacity. The units were delivered by Coast Recording Equipment Supply, of Hollywood. **ANDY MORRIS** is chief engineer of the studio.

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STUDIO UPDATE

Southern California (continued)

■ **SALTY DOG RECORDING** (Van Nuys, California) has begun operations of the new Fairlight CMI (Computer Musical Instrument), at the Van Nuys location. Fairlight CMI is a "state-of-the-art" audio-digital storage unit. Salty Dog Recording's research and development team has also finished custom modifications on their 3M 2-track master mixdown machines, which include the removal of transformers and an upgrading of circuits. *14511 Delano, Van Nuys, CA 91411. (213) 994-9973.*


Salty Dog Recording

■ **STUDIO SOUND RECORDERS** (North Hollywood, California) announces the completion of their new two room facility which includes Harrison automated consoles, UREI Time Aligned™ monitors, EMT gold foil echo, Lexicon 224, and a full array of microphones and outboards.

■ **THE CONCORDE RECORDING CENTER** (Los Angeles) has opened its doors in the old ABC Recording Studios which were operated briefly during the past year by Scott/Sunstorm. The facility features three 24-track recording studios, one with full video sweetening and post production facilities. Two of the studios have undergone complete acoustical redesigning and restructuring as well as complete electronics upgrading. Plans to re-do the third room will be implemented by the end of the year. The new managing director of Concorde is **WARREN ENTNER,** formerly a member of "The Grass Roots," and later with the Gem/Toby Organization, a London based management firm. **REGGIE DOZIER,** who has worked at the studios for many years, is the chief engineer of the facility. *8255 Beverly Boulevard, Los Angeles, California 90048. (213) 658-5990.*

■ **THE SOUND CHAMBER RECORDERS** (Pasadena, California) have recently upgraded to 16-tracks with the addition of a 3M M-79 tape recorder. A pair of Super Red monitors have also been installed to supplement the JBL-4311s and Auratones. Mixing is done through a Tangent 3216 console with UREI limiters and BGW supplying the power. **RICHARD McILVERY** is the studio manager. *27 South El Molino, Pasadena, California 91101. (213) 449-8133.*

Northern California:

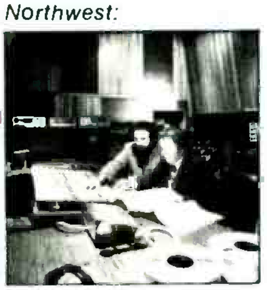
■ **MUSIC ANNEX RECORDING STUDIOS** (Menlo Park, California) announces the installation of their new 32-input Neve console in Studio A, plus a complete acoustic reconstruction of the control room. The console installation was handled by Bay Area Studio Engineering. New additions to the operation's Studio C include an MCI 416, an Ampex 16-track, and a Steinway grand piano. *970 O'Brien Drive, Menlo Park, CA 94025. (415) 328-8338.*

■ **BAYSHORE STUDIOS** (San Carlos, California) is celebrating its first anniversary of operation. The 8-track facility features Tascam boards feeding recorders by Otari, ReVox, and Tascam, and JBL 4311s and Auratone are employed with amps by BGW, Sansui, and Marantz. Outboards include Delta-graph and TAPCO EQ, dbx RM-155, and Sound Workshop stereo reverb, and mikes are by Sennheiser, Shure, Electro-Voice, Neumann, and Beyer. Among the instruments are a Hammond B-3 organ and a Neumeyer grand piano. **KENT BANCROFT** and **KEITH HATSCHEK** are the owner operators of Bayshore. *871F Industrial Road, San Carlos, CA 94070. (415) 591-3503.*

■ **AUDIO TRANSFER RECORDERS** (Lafayette, California) announces commencement of work on a new studio facility in the Contra Costa area. Studio design will be by **RICHIE MOORE,** of Studio Operations Service, of San Rafael, California. The initial studio will feature a Scully 8-track recorder fed by a custom console, UREI Time-Aligned™ monitors, and a video interface capability. **JOHN, DUNCAN,** and **PETER ROWE** are the owners. *3327 Mount Diablo Boulevard, Lafayette, CA 94549. (415) 838-0368.*

■ **TEWKSBURY/THE HYDE STREET STUDIOS** (San Francisco) is the new name for the facilities at Hyde and Eddy Streets in the Bay City, which were formerly under the auspices of Wally Heider Recording. The studios were acquired by **TEWKSBURY SOUND RECORDERS,** of Richmond, California, and **RANCHO RIVERA RECORDING,** of San Francisco, and include four complete studios and five acoustic echo chambers. The operation will feature a 40 input Trident console, a 28 input Helios console, and a new Otari MTR-90 24-track recorder, as well as Ampex multitracks and a selection of signal processing gear. *245 Hyde Street, San Francisco, CA 94102. (415) 441-8934.*

Northwest:


Womach Recording

■ **WOMACH RECORDING STUDIOS** (Spokane, Washington) has been opened by owner **MERRILL WOMACH** and features an MCI 24-track recorder with AutoLocator III, and Auditronics 501 console retro-fitted with an Allison Fadex system, and UREI 813 Time-Aligned™ monitors. UREI compressor/limiters are offered along with mikes by Shure, AKG, Neumann, and Sennheiser. The studio itself measures 1,250 square feet. **BOB ZAT** is the chief engineer. *East 122 Montgomery, P. O. Box 5378, Spokane, WA 99205. (509) 327-7784.*

■ **TRIANGLE RECORDING STUDIO** (Seattle, Washington) has reopened after remodeling completed this past summer. The listening room was acoustically engineered to suit the UREI 813 Time-Aligned™ monitors installed in the facility. **BILL STUBER** and **JACK WEAVER** are Triangle's owners and operators, and also offer a mobile recording unit. *4230 Leary Way, N. W., Seattle, WA 98107.*

Guatemala:

■ **DISCOS DE CENTROAMERICA** (Guatemala City) announces the grand opening of their 24-track studio. The just completed facility was designed by Everything Audio, who also supplied the equipment. The studio features an Amek M-2000A console with MCI tape machines and Dolby M-24 noise reduction. The design is South American in feel, with American quality of sound. The studio can be contacted through **JOHN MOORE,** of Audio Express. *P. O. Box 89F, Guatemala City, Guatemala. Telephone: (011) 502-2-316319.*

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producer Herb Alpert

by Robert Carr
Photography by Stephen Burchesky

R-e/o (Robert Carr): Have you wanted to be a musician all of your life, or was it just a matter of fate?

Herb Alpert: I started playing when I was 8, but I studied formally for about 12 years. I didn't realize at that time that I was headed in that direction. I kind of slipped into it through the back door. I don't think of myself as the normal type musician who has been on the road with the big bands, and played with orchestras and groups. I went through another process. I think of myself as a producer of records who has a trumpet in his hands, who is combining both of those. I went to U.S.C. for a short time. That wasn't really happening for me, it was the wrong timing for me and college. I wasn't ready for that experience, although I was playing in the orhestra. I moved my name up on the draft because I was about to be drafted anyway. I went into the army for a couple of years, and was placed in the army band. Halfway through the Sixth Army Band I started to get into jazz, listening to Miles Davis, and the jazzers of the day. Clifford Brown was the one who really struck my ear. He is a wonderful player. I got intrigued with that whole idea of not looking at music and just playing what was coming out. When I got out of the army, around 1957, my ex-wife's best girlfriend was married to Lou Adler, and Lou and I became real close friends. Lou was writing poetry and I had some working knowledge of the piano, so I put some melodies to his words. It felt pretty good relative to what was happening at that particular point of time in the music business, so we started hustling the songs. Lou was always very aggresive — a lot more aggresive than I was. He got us in some doors and we cut some demo records. One company in L.A. liked the records a lot and hired us as staff writers. That's where we really got involved in the recording process.

R-e/p: Were you a song writer for quite a while then?

Herb Alpert: We started out that way in the business. We wrote some songs for Sam Cook — one was very successful, one was moderately successful. All that time I was still playing the horn on weekends, because I was hooked on it and doing it for my own pleasure. When the *Lonely Bull* came about, the trumpet was the perfect instrument for that particular sound that I was interested in laying down.

R-e/p: Did you know at that time that the Tijuana Brass would be a really excellent vehicle, not that I want to dwell on that a long time?

Herb Alpert: No, I had no idea. I knew the idea was sound, and essentially I was trying to translate a feeling I had at the bullfight, my first experience in Tijuana. I was trying to transfer it out to a record, trying to make a visual effect. After we put the crowd sounds on, the "ole's" over the introduction of the record and on the ending as well, the record really came alive. I expected that it had a great chance because it was different.

" . . . I treat every recording session as a rehearsal . . . the panic is not on, and I'm not into neat, clean records . . . I don't want any musician playing under the threat of making a mistake! . . ."

— continued overleaf . . .

"... to get to be a good producer ... you've got to be a real good listener ... and a quick listener ... sometimes it happens in seconds ... you've got to be able to spot it!"

It caught your attention immediately, and in those days the publishers and record people were really talking about hooks — put a hook in the bridge, put a hook here, and put a hook there. This thing really had a hook from go.

R-e/p: So it was just a gut feeling that you had with the song?
Herb Alpert: It was a gut feeling that the record was real good. When we finally released it, it was overwhelming.

R-e/p: It must be a great feeling having another hit record again, and winning a Grammy.
Herb Alpert: Well, I'll tell you, the feeling of having a hit record is certainly great, and also the fact that so many people were pulling for me. For some reason I had it wired wrong. I felt that since I had been so successful in the past as a musician, there would be some little built-in resentment. I found just the opposite out there. The people were really supportive.

R-e/p: You still have the name. You're still really well known.
Alpert: I know, but there's something about when you reach a certain status in the business, you almost kind of pull for someone else to come along. I don't know if it's honestly resentment ... I felt like in some areas, I had this Tijuana Brass image to sidestep. People who remembered me from those days either wanted to hear that type of sound repeated or felt that my music was in a particular pidgeon hole. *Rise* kind of took a left turn. I think Black Radio gave me a shot at it, because they didn't get involved with the Tijuana Brass. They just heard the record, liked it, and played it.

R-e/p: There were a couple of tunes on your recent album that had almost a Tijuana Brass flavor to it.
Alpert: Yes, I'm sure there's traces all through it, but I try to be as honest as I can on records. It's work trying to carve away all your craziness, and all the things you're not willing to show. Little-by-little, I'm showing more-and-more. My sound is pretty much the same as it always was.

R-e/p: You're the only artist/producer who's also vice chairman of his own record company, and you mentioned that you don't write anymore because you feel you're better equipped to listen to somebody else's songs and inject your own ideas. Do you feel that when you wear three different hats it creates a strain on your objectivity?
Alpert: I separate it. When I'm in a creative mood or when I'm recording, I don't take part in the cigar smoke meetings.

R-e/p: But isn't it in the back of your head?
Alpert: No, I've learned how to do both now. We have a tremendous staff of people I feel really comfortable with, and they're very capable. My partner is understanding of that process and I'm able to slip into the artist's groove real nicely. That's what gives me pleasure. If this wasn't pleasurable or a pleasure experience for me, I wouldn't do it.

R-e/p: The music business is cyclical. You had a big hit with "This Guy's In Love With You," in 1968. Has your basic philosophy for production changed very much over the years? I'm sure you've grown over the years in many areas as far as production goes, but has your basic philosophy stayed pretty much the same? Or maybe the cycle has gone around and caught up with you again?
Alpert: I think it's yes to both parts. It has changed, and it hasn't. The thing that I was disturbed about when I was recording for a major company prior to A&M was that it was a very sterile environment — it was white on white. Engineers were running around with buttons on their lapels; my hand was slapped because I wanted to put up a little more bass on the board. They were lecturing me on the unions and how I couldn't do that because this, that, and the other thing. I realized that it was not the way to get the most out of a creative person, and at that point it was in my head that if I ever had the chance to do it, I would do it much differently. It seems so simple ... you can't force anybody to create; you can't give anybody demands to create, but you can at least set the stage right. If the artist feels comfortable there, he feels comfortable to do whatever he or she chooses to do. This is one of the main things I try and touch when I go into the studio. I select the musicians I feel most comfortable with, and the song I feel most comfortable working on at that time and set the environment for them so they don't feel threatened; they don't feel that they have to produce something or else this person who's paying their bills will be disappointed. I leave it very loose. I treat every recording session as a rehearsal. If it doesn't work out, we'll get it next time. The panic is not on and I'm not into neat, clean records. I'm into a certain naturalness that allows spillovers and goofs. I don't want any musician playing under the threat of making a mistake. I'd rather he be adventurous and express his insides through me, having an understanding of what I'm looking for and then allowing the freedom of letting him express himself through those thoughts. It's very important for me to set that stage right.

"... letting the artist express his insides through me ... understanding what I'm looking for ... allowing him the freedom to express himself!"

R-e/p: I notice on your albums that most of your tracks are basically done by the same musicians. There is a definite similarity all the way through. I get the feeling that A&M is a great family-feeling organization. Is that what you're striving for — that family feeling? Do you feel that's very conducive for a creative environment?
Alpert: I don't think we actually identify them like that, but the feeling is here. It started with Jerry and myself in 1962 in my garage, and we've always hoped that every employee we've brought in would reflect our feelings, our philosophy, our general concept about the business — about quality and caring. Jerry and I have always thought that money and all that was just a by-product of doing something well.

R-e/p: And if you enjoy it, then the money will come to you.
Alpert: Yes. So I guess I have just instinctively been very conscious about the environment. I believe strongly that everybody should have their own personal integrity intact. You have to feel good about yourself and what you're doing.

R-e/p: And again, it shows up in the music.
Alpert: It comes back.

R-e/p: Who would you feel is your biggest influence in terms of you being a producer? Is there anybody in particular that you've really looked up to as a producer?
Alpert: No, mine was kind of a seat-of-the-pants operation. In the very early days, I was watching Bumps Blackwell. He was producing Sam Cook. I watched how he was doing it and I think it's just a real personal experience. When I'm producing others, I feel that it's my job to be the middleman between the tape machine and the artist and try to get the most out of an artist. When I'm working on my own thing, I try as best I can to be an audience to what I'm doing, and not just being a musician listening to a trumpet play, wondering whether it's in tune, or out of tune, or all the other things that sometimes we get caught up in. I'm watching for the total picture. It's a real personal experience that you can't transfer from one person to the next. What I think might feel really good might not feel good to the next person. I always liked what Bix Beiderbeck said, *"Just because I ain't receiving, don't mean you ain't sending."* And that's true.

R-e/p: Yes, it's important to have a producer that's in tune with the people he's working with.
Alpert: Yes, of course, I think the major function of a good producer is to know how to set a good tempo. He knows where that flow is.

R-e/p: And then keeps it going.
Alpert: Yeah. There's a lot of producers that know how to get good sounds, and they know how to manipulate an artist, but they don't know where that tempo is. For me, that's the key.

R-e/p: That's where you get the magic from.
Alpert: Right. That's the magic!

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" . . . I couldn't record dry . . . with no echo, no pump up, I couldn't do it that way . . . I like to pump up the track . . . . put-myself-on just a taste . . . get a little rush going . . . it comes to life!"

R-e/p: when you recorded Rise, was everything charted out, or was there a head arrangement, or maybe just chord charts?
Alpert: Yes, chord charts. Randy (Badazz), coproducer, Andy (Armer), *Rise* writer, and myself had talked about the basic concept. I felt very strongly that the tune should be slowed down. We had it much faster. For me it felt like it was in competition with disco type records. I was trying conciously not to make a disco record, although I was trying to make a record that would appeal to dancers.

R-e/p: It was a big hit in the discos anyway.
Alpert: Yes, but it wasn't in their pulse zones. In checking out the majority of the records that were happening in the disco arena, they were like 120 - 135 beats per minute, and *Rise* just felt too darn fast at that tempo for me. We slowed it down to 100 beats per minute. Most of the reactions we got when the record was released were that the record was too slow. That was from the disco people. I always felt that it was a bonus for it being slow. It was a time for the people who were at the 125 clip to slow down to 100.

R-e/p: I noticed you made that back beat really heavy, too. Was that something that was planned before you recorded, or did that evolve?
Alpert: What happened was, we were working for the first time with the 3M digital process, and it just clarified the tones. You could hear all those overtones and it was a real temptation to bolster every sound we had. I think that record has tremendous clarity and depth.

R-e/p: Oh, it does, but when that back beat comes in, it just pins the needles on my machine.
Alpert: But it's a clean back beat. There's a difference between a rude back beat — one that's just there to catch your attention — and one that's working as a motor. For me that works as a motor, because it's not out of relation. The bass is jumping right up your back, too. It was a full range record and there's not that many people playing on it. And, it was done live in the studio.

R-e/p: One take you mean?
Alpert: Well, I think it was the second or third take, but it was done at one time. The only things we added to it were handclaps and a guitar, but essentially that was the recording.

R-e/p: So, you really didn't change anything. The concept you went in with is how it came out.
Alpert: Other than the tempo. And little things happened in the studio. Like getting the guitar player to stay in beat with the piano. I like to stay very loose. I'm not quite sure at all times what I'm looking for. I know what I'm not looking for, but sometimes I'll hear something I wasn't actually planning on. I'll be able to spot that immediately, relate that to the musicians and encourage them to keep whatever it is going.

R-e/p: So you're into a very spontaneous feel. By getting into the feel of the song. Do you ever have a little mind game you might play with yourself to get yourself into the feel of the song? An actor, if he wants to get motivated for a scene, might come up with a thought that would evoke a feeling, and then the feeling would motivate or inspire what he has to play. Have you ever done that?
Alpert: I start with the feeling right from go, because I only choose material that touches me; I already have that feeling. I wait for the moment when the feeling comes through my body and I'm hearing something that's moving to me. I feel it when I'm playing the horn to it. At that stage, I know I'm on the track and I'm on to something that's working for me. I wait for those moments.

R-e/p: You don't just wait and say, "Oh, I have that feeling," and then run into the studio?
Alpert: Well, I usually play it while we're rehearsing. Then there's an understanding before we start recording about what we're looking for, what type of feeling, what type of mood we want to set, what type of picture. I believe strongly in pictures.

R-e/p: So you visualize the music; is that what you're saying — almost a visual imaging of the picture, and that would induce a feeling?
Alpert: I don't know, I think at this point the unknown starts taking effect. Like the bass player might play something that turns on the piano player that turns on the drummer that turns on a couple of the guitar players that turns me on, and all of them are off. For me, it's hard to indicate that on paper.

R-e/p: Have you ever had that session where you didn't have that happen? Did you just keep going at it, or did you say, "that's it for today"?
Alpert: The only one I can vividly recall producing is Gate Barbieri's second album in New York called *Ruby Ruby*. I remember the circumstances were wrong. I had the wrong studio booked, with the wrong musicians, and a series of things that just didn't work out. Usually I'm able to pin it to either the guitar part is wrong or the tempo is wrong, or let's take another song. You can fool around in a number of different ways, but on that particular night, I remember just staring up at the speakers, now knowing what to do. I didn't know why the ---- I was there, or how to react to it. I felt helpless.

R-e/p: What do you do in that situation, just say, "that's it"?
Alpert: We all kind of realized that it wasn't going to work and that it wasn't going to be a real eventful night. Everybody spots it; it's not like you have to convince anybody. When it's magical, when it's really working, when it's happening, people recognize it. When I'm playing something in my office and I get a knock on my window from the guy who's cleaning up outside, asking me "what record are you playing?", you realize most people instinctively feel something when it's working.

R-e/p: Is that the only time you just couldn't save the session?
Alpert: That was an odd situation. In most cases I'd either switch to another song or do something real drastic — like totally change the arrangement. Then, little-by-little something will happen someplace. I usually deal with real creative musicians who understand my process of making records, and they are willing to change on a dime. They don't get locked in. That's one of the ground rules: "Don't get rivited into anything I tell you, because I might change my mind."

R-e/p: Do you think a good producer has to be a good arranger and vice versa, too?
Alpert: To get to be a good producer, you've got to be a real good listener and a quick listener. You have to know when it's happening. Sometimes it happens in seconds and sometimes it doesn't. Somebody might get something that's really working and you've got to be able to spot it. You have to pay attention and be clear. I see a lot of guys that become adrift when things are going on in the studio and they're pinned onto one particular narrow path. They can't see or hear anything else until that one little thing is cleaned up. Sometimes they miss some great stuff.

R-e/p: You mentioned that with Rise, *you didn't want to make a disco record — you wanted to make a dance record. But* Beyond *is a little faster, more like a disco record. Is that what you planned on doing with that?*
Alpert: No, it just seemed like the right tempo for *Beyond*. The thing I like about *Beyond* is that it's an adventurous record. For me, it's a nice union between man and machine.

R-e/p: Did you use a click track?
Alpert: Well, we started with a click track and then laid on the sequencer.

R-e/p: Did you use a drum loop track for that tune?
Alpert: No, that's Steve Gadd playing.

R-e/p: He's an incredible player.
Alpert: Yes, he is. That was his first take. We talked about the type of feel we wanted on it, and he just pulled out his brushes, sat down as relaxed as could be and just played it. I was amazed.

R-e/p: Did you record most of Beyond *on one take, or were you getting the basic tracks?*
Alpert: That particular song was built up. On that process you start with one sound and then from the ground floor, you build another sound to that, and another sound goes on top of that. It's a whole different process. It's one that I was really not familiar with, but as it ended up, I enjoyed it.

R-e/p: Do you think you could get into doing more of that type of production?
Alpert: Well, it's possible, I don't know. I think I have a better feel for the real spontaneous. You get a good song and a group of musicians together and you find out what direction you want to take it.

R-e/p: Let's say you've already cut the initial tracks and you want to go in and listen to it. How important do you think it is to get a real good mix for that initial first time back?
Alpert: Very. I think recording has to be an inspirational experience. I notice there are

some people who record real dry — no echo, no pump-up. I couldn't do it that way.

R-e/p: So you put everything into basic tracks then? All the effects?
Alpert: Not all the effects, but I like to pump up the track. I like to maybe put-myself-on just a taste. I came out of the Gold Star school (Dave Gold's Gold Star Studios, Hollywood) in the early days and they used to really pump up that sound. Sometimes you'd walk kind of dejected from the recording studios into the control room waiting to hear this playback, and all of a sudden it comes to life, because the echo chamber is doing its thing. There are some problems that all of a sudden don't exist. I like to get a little bit of a rush going.

R-e/p: Did you run into any problems when you used the digital machine?
Alpert: Well, we had one of the first three off the line and we were kind of a testing ground for it. There were little odd things that were going wrong — unfortunately at key times. We lost part of the take on *Rise*. We had to reproduce a section, and at that particular point with the 3M process, they didn't have the editing capabilities. The process was: we recorded it on digital, transferred it over to 24-track analog to do our editing, transferred it back to digital and then added any other sweetening like the hand claps and guitar. In that particular case, we mixed to 2-track analog. It wasn't a total digital test, but it was nonetheless an amazing sound.

R-e/p: Did you notice a lot of degeneration when you went to analog and back?
Alpert: Yes. We were very cautious when we plugged straight into the machine, and were very conscious of especially the bass drum and bass. We tried very hard not to lose any of that sound, but you just naturally pick up some tape noise and some of those little things that would be undesirable. We did have some problems with the digital but I still think it's a brilliant way to record. I love the sound. When the little pops and cracks appear at odd times when you don't want them to appear, you have to call in the think tank for them to start tweaking the machine. They're talking a whole different language so it's not easy to track what's happening. One time we had to borrow the Record Plant's machine to finish this, that, and the other thing. But my overall feeling for digital is very positive.

R-e/p: So you would definitely use it again. Well, you used it on Beyond.
Alpert: Yes, we used in on *Beyond* and stayed with it until the mixdown process which we mixed to analog, because I wanted to be able to intercut.

R-e/p: Rise *was a pretty long song. Did you originally record it in that length or was there a great deal of editing?*
Alpert: The long version was that long version. I think we chopped the intro. The short version was obviously an excerpt from the long version. We did change the sequence. The middle section of *Rise* was originally part of the ending. We disliked the groove on it, so we moved it into the breakdown in the middle. We had plenty of ending left over in addition to not repeating anything other than that one part I told you about where there was a verse after the original breakdown on the live tape that was eaten by the machine. Still the editing is somewhat of a hassle. That's kind of a producer's delight if he edits here and there, but it's not that easy to do on digital. I was willing to wait out the process, but it kind of digs into momentum. You want it done now so you can go to the next thing. Everything goes "ho hum" while they're setting things up to make the editing happen.

R-e/p: Were there any problems that related specifically to production, other than the machine breaking down, that you might like to see changed?
Alpert: Well, the editing process up until now has been very complicated. And there was some type of problem interfacing a 32- with a 4-track and getting all that into sync. I don't quite understand how the whole thing works. I'm kind of in awe with these two machines, editing and doing all sorts of space age stuff.

R-e/p: Just the sound you got back, the clearness, the three dimensions from it, and so forth . . . did that make you alter your production technique at all or maybe let you expand on it? Did it make you change in any way?
Alpert: I was certainly aware of the possibilities. It's a great experience to ping pong on the machine, because you can go endlessly from one track to the other without degradation in sound. That process is really very attractive. The choice of microphones sometimes would be different, because you can capture some real tingle on digital that you don't get on analog. At times I would go for a more sophisticated microphone, like the AKG-451. That's a real nice sound. There are some things that I didn't recognize before. In fact, on one of the cuts on the *Beyond* album, there was a little saliva in the trumpet that I didn't realize at the time and it got picked up. It's dull-like when you listen to the record gurgling away. It's certainly human, but analog wouldn't have picked up that sound. The sonics are terrific.

R-e/p: Did you transfer that back to digital in order to master it, or did you keep it on analog?
Alpert: We mixed analog and stayed there. The next one it's possible we'll continue on all the way.

R-e/p: Did you pick up any new production tricks by using the digital in the last couple of albums — things you might not normally do?
Alpert: I don't know if they're tricks as much as each sound has its own little frequency response and you can hear layer sounds a lot better than you can on analog. You're not dealing with hiss factor and the build-up of all the nonsense. You can put on a lot more if you're careful not to jam frequencies. For example, on the last cut on the second side of *Beyond* is a track called *The Factory* which has a lot of different sounds going on — backwards stuff, bottles dropping, and chains falling, pianos with MXRs, and a lot of stuff that I'm not sure would have been that easy to do on analog. It's so clear that the horn just sits right on top of it. It's nice that you can hear every bit of what the horn is doing and also hear all the little details that are happening. We became aware of both possibilities and at times went for some more salt and papper than we would have normally.

R-e/p: Have you done much work with the automated consoles?
Alpert: Just in the real early stages.

R-e/p: Do you have any here at A&M?
Alpert: We have one coming, and it should be here next week. It'll be hooked into our Trident board.

R-e/p: How did you like working with it?
Alpert: This was years back when they first became available and I liked it. It was working when I was able to keep my hands on the board and then make some moves myself. I don't like to sit back and clinically make decisions that the bass it too soft or we need more piano. It was nice when I was able to move it and then come back to it two days later and have that same mix up and just touch up a spot without trying to get too clinical. The problem I feel is that you could get too analytical about what you have, be over intellectual about the process and clean it up to the point where it doesn't live; it doesn't breath.

R-e/p: When you do a tune, do you do it all at once? For example, you go in and cut the basic tracks, sweeten it, and mix it at one time, or do you do a few different rhythm tracks, sweeten them all at one session, and then come back and mix all the tunes at one time?
Alpert: It's done in different ways. Usually I go for various tracks. I have a day for sweetening and collecting ideas; layer it, and take a look at it. I do it over a period of time. I find that for me it works best when I don't labor too much on it. I like to go with my first instinct. If something hits me right, I go with it. I try not to overthink it.

R-e/p: Do you find that if you go from day-to-

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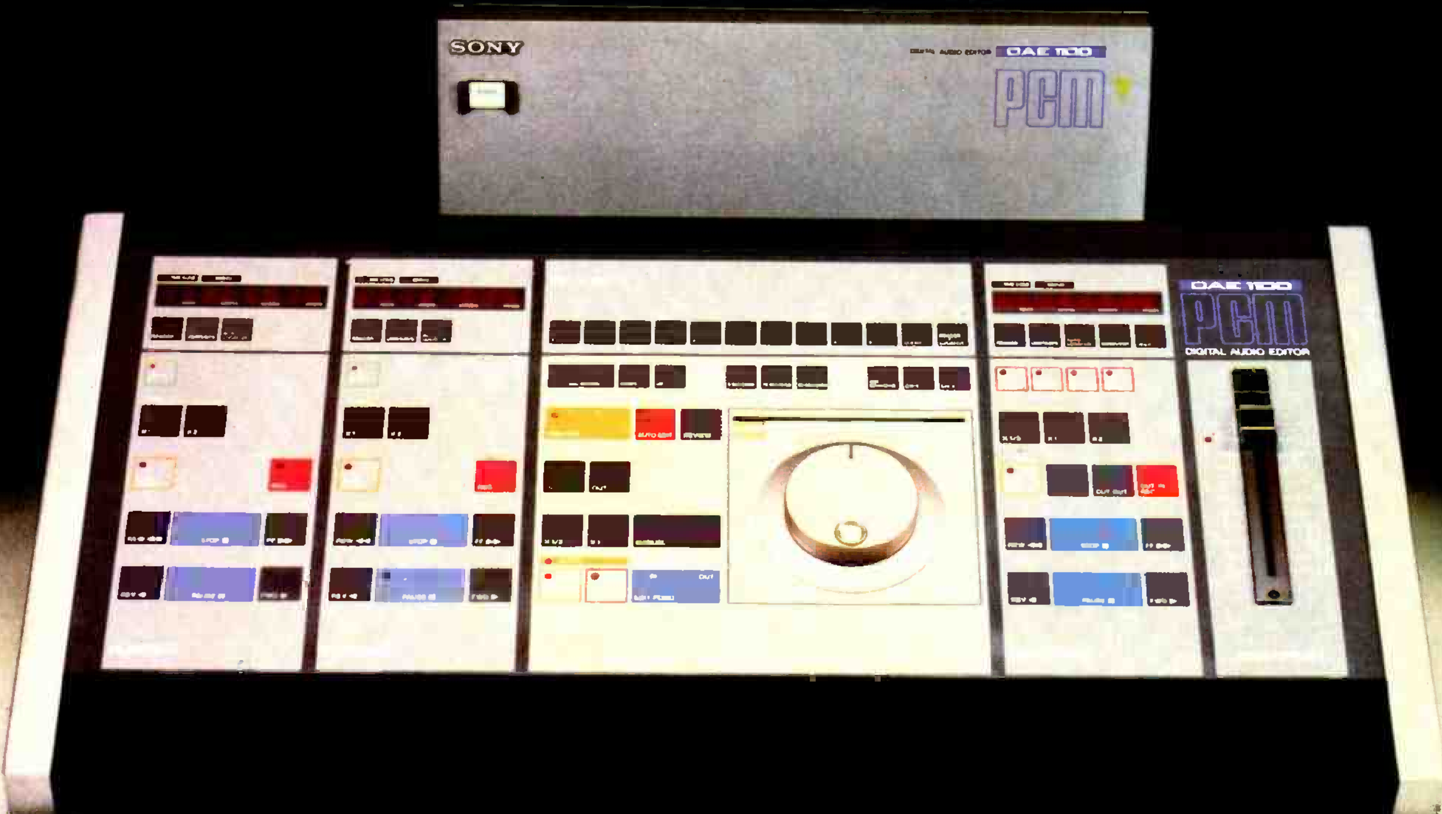

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"Mixing . . . I don't labor too much on it . . . I like to go with my first instinct . . . I try not to overthink it!"

day — let's say it takes you a week for your basic track, sweetening mixing, and the other stuff — then your feeling, of course, would change over that whole amount of time? Does that bother you at all?

Alpert: It does, and I try to stay on it and not have too much time between the time the track is recorded and the time it's finished and the last sweetening is put on because you do change. You can get tired of it, too, if you hear it over and over. It feels like a hit record you don't want to hear anymore or a record you don't want to hear, period, which sometimes has its plusses. When we record more than we need for an album, after hearing them over and over, the ones that don't work start sticking out, start bugging me, and I'll without hesitation either eliminate it or change it drastically.

R-e/p: What's the first thing you do when you're approaching a mix? Let's say you have all the basic tracks, you have everything you need, you're going to sit down and mix it. What would you usually go for first?

Alpert: The right room and the right engineer. Make sure you have the right engineer — one you trust — and a room that you feel comfortable with. Musically, you have to get a good bass sound. The next thing is to lock in that good bass sound with a good bass drum sound. That's the foundation of the record. You can't build a top floor before you have a ground floor. Then you go for the drum sound — the overall kit — and there you have choices. You can spread them 'way out, or move them into the center, or sometimes if it's spread too far left or right, it can sound like the Jolly Green Giant. It's not always comfortable and we prefer to keep them pretty close to the center. Most of the time I don't choose to go to the extremes. I think the drums and bass sound are really the key.

R-e/p: You mix those basically in the center?

Alpert: Well, the bass stays in the center and the bass drum stays in the center as well.

R-e/p: That's pretty much your basic concept — dealing with those two instruments first?

Alpert: Yes, for starters. Then I lock in the rhythm section. It's important that the rhythm section be one section. They have to be tight — not letter perfect, but tight in sound. Everybody should have their place. I'm not crazy about equalizing either, unless you're going for a special effect.

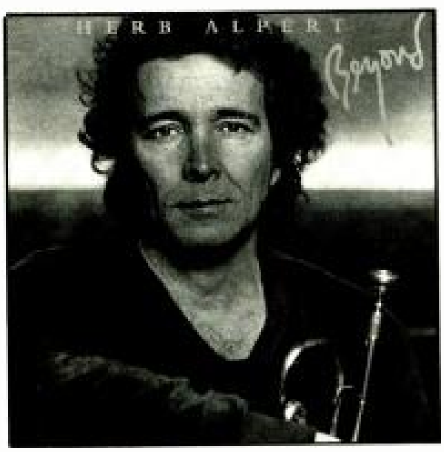
R-e/p: When you're working with the drummer, do you let him basically tune his own set and find his place in the track, or are you very concerned with the way the drums are tuned?

Alpert: Oh, sure, I'm concerned with that, and I usually use the drummers that are aware of that, too. They're aware that you can cancel out sounds. If the bass drum is too close to the frequency of the bass, he's not going to be heard. Most drummers want to be heard so they're becoming aware of that problem now. It's important also to get a drummer that has a good kit — a good sounding set of drums. There are a few drummers in town that have everything together but the sound of their drums. The cymbals are a little bit too skinny, the snare drum's a little undernourished, and the bass drum is not tough enough. It's all real personal.

R-e/p: How do you get that sound on your trumpet? Is that an echo, or a reverb. There seems to be a characteristic sound that runs through all the cuts on the trumpet. There seems to be a little reverb or echo, or digital delay, or something.

Alpert: On some cuts I use some additives, a little Echoplex here and there. For the most part, it's just the sound. I use the C38 Sony microphone, the old tube mike. I like that sound as opposed to the more sophisticated microphones. It gives a little warmer sound. I'm conscious of mike placement, the distance between the bellhorn and the microphone, the acoustics in the room, and I'm using a real mellow horn with a rather large mouthpiece so I have that particular sound. I don't affect to get it. That's the sound I get.

R-e/p: I notice a distant sound to it.



Alpert: On the *Rise* cut we used quite a little bit of just natural chamber or EMT, I should say. I think there was an interesting relationship between the horn and the track. I purposely didn't want to stick it out too far. I wanted a haunting sound like a Continental sound, like recorded in Mexico.

R-e/p: On Rise, you said you recorded it all at one time. Do you like a little leakage on your trumpet mike?

Alpert: Not really. On the horn, I prefer to keep it pretty well isolated because sometimes, even though I like to play live while we're doing the tracks, it gives me the indication whether the groove is right, or if the fill-in parts are working against what I'm doing. I still like to have that option of being able to take it off and re-recording it. If there's too much leakage, you get the shadow horns and you have a real problem trying to Kepex sounds out.

R-e/p: What about the rest of the instruments? Are you very concerned about keeping them all clean?

Alpert: Sometimes. I never used to be. It was never even a concern of mine until digital. I started to realize you could get both cleanliness and sloppy at the same time. You can get good clean sound but it doesn't mean that every one has to be letter perfect — especially on the drums. It's nice to get a good tight sound on that snare without a bunch of additives coming into that microphone. You don't want guitars and bass leaking into that snare mike if you want a fat solid snare sound, because you're going to have to deal with all those sounds when you mix down. A lot of records are made by putting on the drums after the fact so you can isolate the snare and get a big sound. That's like letting the snare hack some wood at you. I haven't found it to my liking to do it that way. I like when there's a response. There has to be an interaction.

R-e/p: That energy carries through the music and through the record.

Alpert: Yes, energy isn't just a thing that one person creates. In fact, that's one of the things I picked up on when I did the album with Gato Barbieri on the *Caliente* album. I was listening to his music and I realized he had a tremendous amount of energy. For the most part the guys behind him were kind of snoring away. They were afraid to really get up there and shout with him; there was a governor on them. When I was doing the *Caliente* album I was very conscious of getting guys like Lenny White, a drummer and bass player who were willing to go with him, be adventurous. I think that's what jazz is all about — the interaction.

R-e/p: Do you tend to monitor with the speakers very loud, or very soft, or in between?

Alpert: I listen all different ways. I have staying power. I can listen for a long time and not get distracted by centering in on what I'm doing.

R-e/p: A lot of people, when you talk to them, say, "Well, I can listen for a while but then I start losing the top end; I can't hear the top end." They keep boosting it and when they listen again, they're shocked at what they did the last time.

Alpert: I've never really run into that problem. Here, again, it's important to have a good engineer at your side. When you start questioning whether something has enough top end, you turn to somebody that you trust. It's important to get that feedback. I never had that be a problem. Once I start recording, everything else stops. I'm just into what I'm doing.

R-e/p: Do you bounce the sound back and forth between different sets of speakers?

Alpert: Yes.

R-e/p: Which ones are you using?

Alpert: We're using the Altec crossover system in the studio at the moment. And then I'll use the small Auratones at times. Don Hahn, the engineer, has his own favorite little set. I don't think he has a name on those. He calls it job security. I recognize that all systems sound differently. I once went into a stereo shop and took in one of my records. I switched between eight major systems to find that they're all totally different in the bass response. Sometimes it seems like the echo disappears, or the middle tucks away, or juts out at you depending on how much midrange is boosted. It's hard to make a record or tape to satisfy all systems. I try to get it in the middle there someplace by listening loud,

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soft, medium, on different types of speakers — sophisticated ones; some need cheapos.

R-e/p: You're a real speaker junkie!
Alpert: Yeah! In the room here I have the Altec 604s that I've had for years. They're a little harsh but I know what they sound like.

R-e/p: You made a statement, "Things have changed so much, I don't know if there's a demand for a tour." What could you do to launch a new group in the 1980s? Do you think touring is important for success or do you think video cassettes and video discs will become a replacement for touring?
Alpert: I don't know. It's a little too new to have enough information. Suspicion has it that's it's going to be a major event. In terms of myself touring, I've had offers and I'm considering doing some interesting things, like to do a few weeks on Broadway — a show called "*Herb Alpert Now and Then.*" Maybe have two groups — some of the members of the Tijuana Brass and then the things I'm doing now. We'll bind them together on stage, and thread it together with some kind of story.

R-e/p: Sounds like it could be a lot of fun.
Alpert: Yes, it could be a lot of fun if all the elements got together right. But touring in the traditional sense the way we used to do in concert halls is pretty tough. It's expensive.

R-e/p: Do you think touring's pretty much out for any group? Even if you had a new group?
Alpert: Well, it doesn't seem to be out for Led Zepplin. The big rock groups find no problem with it.

R-e/p: But they've been established for quite awhile. With them it's a media event — a nostalgic social gathering.
Alpert: I think the older generation is a little frightened, they're not used to going out any more. They're used to doing the summer circuit, the outdoors, the amphitheaters, but I don't think they're conditioned anymore to go to the Civic and listen to someone they can identify with.

R-e/p: They've matured now; they have money.
Alpert: I don't think it's that, I think they're just been frightened. They hear about the craziness that goes on — like what happened at the Who concert, the drugs, and they don't know how to deal with that. The sound of it, I think, is frightening to them. I think they're conditioned not to do it at this point. It makes it harder for an artist like myself to think of touring because it's a little different out there. It might be able to work in Las Vegas, or Tahoe.

R-e/p: I have a feeling that's where it's going to go; more intimate showrooms like that — a little classier atmosphere where you get dinner and watch the type of music you were raised on.

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Alpert: But that's another category. I don't know if it turns me on to get into something like that.

R-e/p: Do you think it's a little too clinical?
Alpert: I don't know if it's clinical, but that's "show time;" that's getting the lights right. It's another way of doing it. It's not just making music. It becomes an eye spectacle. Music is secondary to seeing the artist and hearing vaguely something that resembles the record.

R-e/p: Do you think when video discs do come out, that they'll find enough established artists to work in that medium?
Alpert: Yeah, I do.

R-e/p: What type of format do you think could work?
Alpert: I just don't know, but I suspect it would be a lot of young talent to be involved at the other end of that camera. Some new ideas and some innovative things are going to start coming out of the woodwork. It's here, and I think it's going to be real exciting. It's going to be good and it's going to be healthy.

R-e/p: Do you think there'll have to be a whole new breed of entertainers? It's hard for me to imagine most established rock stars making the transition into video, into a visual media.
Alpert: It depends on how it's done. It's certainly not going to be like the Scopatone, or whatever it's called — the juke box with the pictures, that corny stuff. It was static and didn't go anyplace. There was music but nothing happened. It was the novelty of seeing. It's not going to be that; it's going to be a far cry from that. It's going to be very sophisticated with optics and all sorts of effects.

R-e/p: So you don't necessarily think it will be the entertainer in the picture?
Alpert: I think it will be a combination. The guy behind the scenes will be as much a part of the entertaining as the entertainer. No, I don't think it's going to be a band playing their song on a stage. That isn't going to satisfy anybody. You see that on *Midnight Special*. That's not going to happen, but I think there's going to be some exciting things.

R-e/p: Pink Floyd has done some things with good visuals along with the music, but if you see it a couple of times, it tends to become boring.
Alpert: Yes.

R-e/p: I'm just wondering what kind of staying power it will have to get people motivated to buy software like that. If they will only watch it a couple of times. They could always just listen to it, of course.
Alpert: I suspect there's going to be some eye appealing things happening that are going to make you rewind it and check it out again.

R-e/p: Have you seen the thing that McCartney did with "Comin' Up?"
Alpert: I heard a lot about it.

R-e/p: He was all 10 of the musicians, each a totally different character. There was a personality there that I was excited about. I wanted to see it several times and check out each one of the characters that he was portraying. To me, that was something I wanted to see over and over again.
Alpert: I've heard about it and just by knowing what he's done in the past, I'm sure it's good. He has a lot of charisma, so he can carry something like that. I don't think that just a regular old stare-you-in-the-eye artist is going to come off in that media. You need a little action there. They're going to have to think of it more than just making music. It's going to be visual as well, and McCartney does have the instincts for something like that.

R-e/p: Do you think you're going to be getting into very much video production?
Alpert: Oh, sure, we're tooling up the big sound stage right now. We spent last month revamping it.

R-e/p: What kind of plans do you have as far as artists and format?
Alpert: I think right now we just have to wait and see. Let the biggies fight it out and see what kind of configuration they're going to go for and let them test the market. We don't have that type of money in our slush fund to toss around, but they'll give us the indication and when it's there we'll jump right in the deep water and go for it. At the moment we're kind of just holding back.

R-e/p: It's just a matter of being ready for that particular time.
Alpert: Yes, we recognize it's around the corner.

R-e/p: Let's say you take a music group, make it visual, and broadcast it across the TV through a small speaker. Do you find that you would have to do a different kind of mix for that kind of speaker? Maybe different than you would use for records?
Alpert: Oh, sure.

R-e/p: What kind of compensation would you make?
Alpert: You know, I don't know enough about that process and how that sound is transferred to be accurate. I noticed when we were doing TV specials that if you pushed the bass too much and it hits that limiter at the TV station you get a lot of mush. It rolls and folds everything in.

R-e/p: Did you make any compensations on those shows as far as the mix? I'm anxious to know how you changed your mix or if you just pulled out a record mix and used that.
Alpert: No, a lot of times we heard it through television sets — not mixed it through, but at least at the last stages we heard what it would sound like through television sets, which is certainly not a great indication, because you don't have all those additives. I believe on the

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last two shows we did somehow convince them to turn off their limiters on a national level. I know it sounds kind of grandiose, but we were reasonably happy.

R-e/p: So the source tape you use is basically the same mix.

Alpert: Well, it was and it wasn't because we were dealing in TV and dealing with mono and it's certainly different from making stereo records. Some of us use the CSG process and some of us get the mono mix from the stereo mix.

R-e/p: What is the CSG process?

Alpert: Well, it compensates for the mid. When we play a 2-track tape through a mono system, it would be the same as squeezing a hot dog bun with a hot dog in it and having the hot dog squirt out in the middle. Everything in the middle would come up about 6 dB in proportion thereon. So the CSG process allows you to push back at the center portion of the record in 3 dB steps. For the most part everything else remains intact. That's what allows a mono station to play a stereo record and sound much better than if they were just playing a tied stereo record. That was one of the problems we ran into in the earlier days when we switched from mono to stereo. A lot of stereo just sounded terrible on mono AM radio. The bass kind of disappeared, and nobody knew where to put the bass. We were too involved in effects. Put the bass here, put the choo-choo train over there, put the guitar over there — spread that sucker out and it sounds sounds like hell on the radio. We had to find a way to compensate for that. And then Howard Holtzer came up with this CSG — compatible stereo — that has been very good. Also, in TV you're fighting major problems in that the American system, for the most part, is not too efficient. In Europe it's different. The Telefunken television sets are more concerned with sound and get a much better bass response.

R-e/p: They have these receivers now where you can pick up TV signals and play them through your stereo system for good sound.

Alpert: I know. Eventually it's going to go to that in TVs themselves; it has to. Of course, there's simulcast, too.

R-e/p: Well, this whole 25 - 45 age group has been raised on quality sound — records, discos, concerts, and so forth. There has to be a better sound on TV to get the people to watch.

Alpert: You're right.

R-e/p: You've been quoted, "If I ever had a record company, I would definitely give more importance to the artist because it all centers around the artist." Is that still your philosophy with money being pretty tight and the record business cutting back?

Alpert: Yes, but that doesn't mean they can't make independent decisions about what they want to record. It's important for the artist to feel comfortable, that they have a home and that the company is emotionally and musically behind them. They need a certain amount of freedom within some limitations to be able to do what they have to do. I don't think you can just deal with bottom lines, and how much you want to spend, or "give us this type of record, or else." It might work for some, but it wouldn't work for our company. It's an artist oriented company, for sure. Jerry and I always base all of our decisions on how our stomachs feel.

R-e/p: That's usually what's in tune?

Alpert: I go straight for that and really deal with first instincts. There were a couple of times in the past where I passed up a good record or good song and my first instinct was to go with it. A couple of times it came back to haunt me.

R-e/p: What do you do to stay in tune with what's going on in the business so that your instincts stay in tune?

Alpert: I have a good balanced life.

R-e/p: So you're not out seeing a lot of groups?

"Monitoring Fatigue? . . . it's important to have a good engineer at your side . . . when you start questioning whether you have enough top end . . . you turn to somebody you can trust!"

Alpert: No, I don't personally go too often. For some reason, I'm just not a great audience to know all the different types of groups and artists. I just don't get off doing that. I appreciate great artists, and the arts, but I can't . . . I'd rather spend the time developing my own, practicing the piano, looking for songs.

R-e/p: To get back to my other train of thought . . . when money is tight and times are real hard, people tend to spend more money on entertainment. Do you think the recession that we're going through now will be healthy for the entertainment business?

Alpert: Only time is going to tell that. It's pretty hard on everyone right now. Everybody's pruning down pretty well. It's been healthy in a sense for the record business, because we've taken down a lot of dead wood; we've taken some marginal artists away. We're all trying to get to the creme-de-la-creme and put out select records and quality records. In our case, it still leaves some room for the off-the-wall stuff — the stuff that you would least suspect, because we're right in the middle of the target that deserves to be heard, which is again, real personal opinion.

R-e/p: Do you see the record industry teaming up with the TV industry in the next decade?

Alpert: I see us teaming up with the radio industry, because we have to work a little closer together. We're on the same team. At times you get the feeling we're working at cross purposes. They need our music, and we need them.

R-e/p: I'm thinking more in terms of visual media. With video discs and video cassettes becoming more popular, do you foresee more of a union between the TV and the record industry? Do you foresee the two getting together to go after that market? Video discs, video cassettes?

Alpert: I don't know. TV's a mystery to me. For my personal purposes with TV, there's so much garbage on it. Unless something drastically changes on it, I don't see much hope for it.

Well, you can get good learning tapes and great quality tapes by your favorite musician. I think it's going to replace all these sit-coms and these nonsense shows that they're tyring to shove down our throats and are pretty successful at the moment. I think it's going to catch up to them. You can only eat so much of that and then you realize you haven't had any nourishment. You realize it's not doing anything for you and you have to find something else that will.

R-e/p: I noticed you had on the back of your album, "Thanks to Randy and Andy for helping to show me another color in my rainbow."

Alpert: Real poetic, huh? I was happy that I got together with Andy and Randy. There were some things that they were listening to that I really wasn't aware of on account of my background and my age, and they made me aware of and vice-versa. It was a give and take. They certainly added a lot to the things I'm doing. I recognize it and appreciate it.

R-e/p: So you'd say you are a man who embraces growth and development?

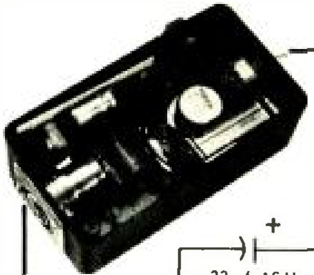
Alpert: I like to think so. Still water gets stagnant. You stop living if you stop growing.

R-e/p: What do you see your direction being on your next album, or next couple of albums?

Alpert: I'm not sure. I would like to improve on what I'm doing. I've reached another place with the horn and I'm starting to become more comfortable. I think I'm able to express myself in a little more adventurous way than every before, and my experience with these two albums will certainly be an aid to anything I will do in the future. It's always a weird question, because I'm not really creative until it's time to dig in. If I was an architect and somebody said to me, "Man, whenever you get the plans finished for this house, put it on my desk," they'd never be on his desk. But if I have until September 30th to get them on his desk and $30,000 to do it, they'll be there and be darn good.

R-e/p: So you're saying you need a little bit of pressure then?

Alpert: . . . Yes, maybe it is pressure. I need some guidelines, but once I start getting creative and getting into that mood, things start to get into focus. So if you ask me that in a couple of months, I'd be more than glad to answer. I'm not in this business for the money. I'm in it because I love music. I don't have any hang ups about accepting the money, because I feel I've earned it, but if all you go for is the bucks, you end up with a hole in your stomach. I've been lucky to be successful at something I love — playing and producing music. □□□



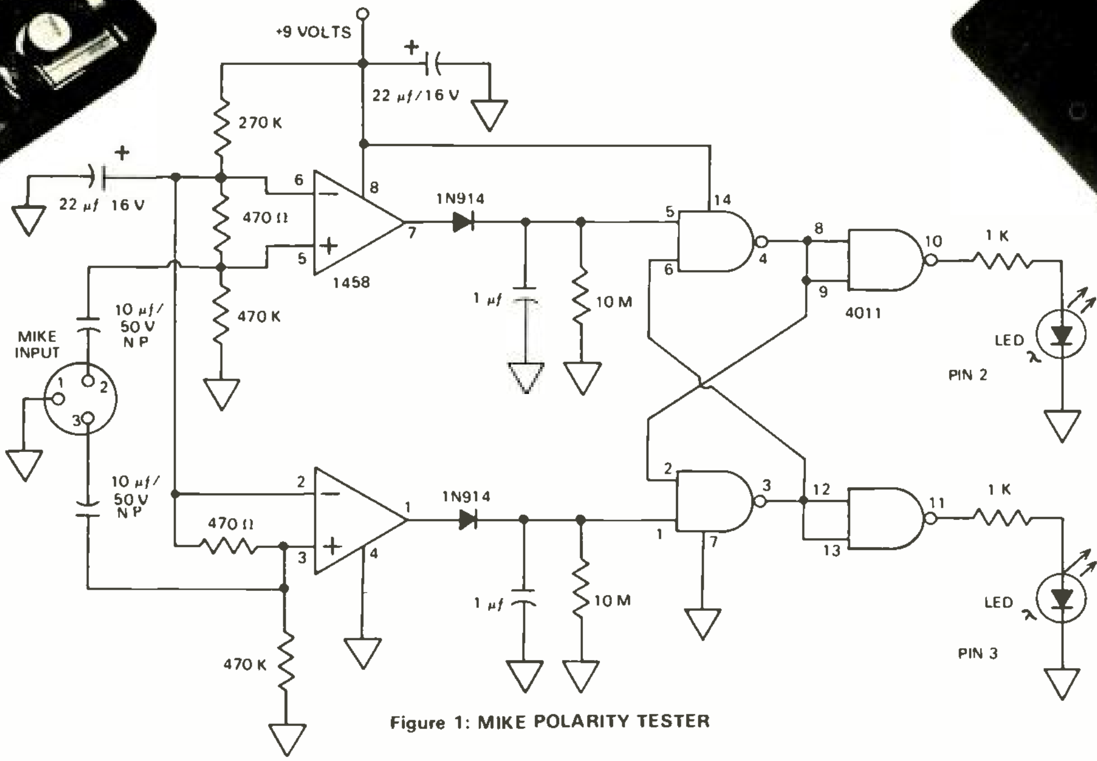


Figure 1: MIKE POLARITY TESTER

BUILD A MICROPHONE POLARITY TESTER

BY ETHAN WINER

Photos by Andy Goodspeed

No one would dispute the importance of proper microphone phasing, or more correctly — polarity. In multiple miking situations, even one mis-wired mike or cord can throw the whole mix off. Similarly, monitor systems with multiple drivers must be wired for "in sync" operation as well. Now, I'm not talking about time alignment, but simply the direction the diaphragms go when presented with a single pulse input. All of this has been covered before, of course,[1] and is not the ultimate point of this article, though there still seems to be some confusion as to the difference between phase and polarity. Not wanting to be accused of beating this thing to death, let me just say that phase is both frequency and time dependent and can vary anywhere in value between 0° and infinity. Polarity on the other hand, has nothing to do with any of the above and can only be either 0° or 180°.

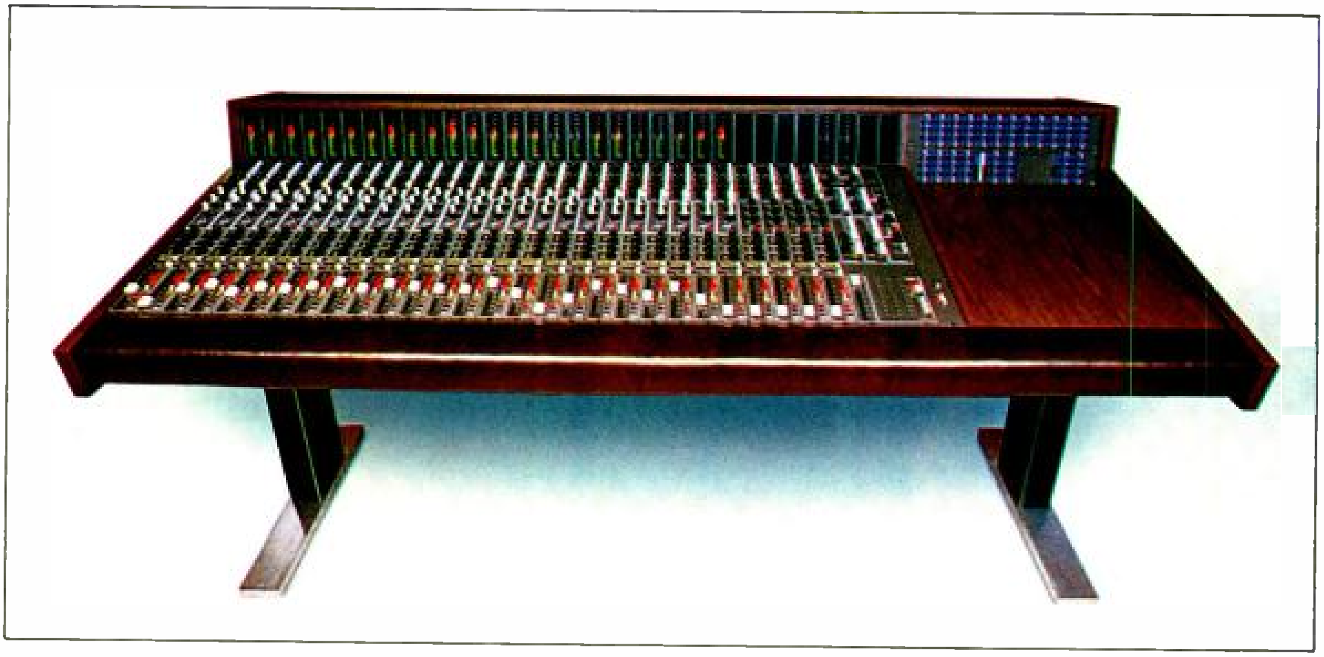
With that out of the way we can now get down to the project at hand which will be useful to anyone who regularly deals with a lot of microphones. Though the market seems full of various devices that can instantly check the integrity and phasing (oops!) of microphone cables and snakes, it is clearly more difficult to determine the absolute polarity of the mikes themselves. Granted, this is not something that you will need to do everyday, but when the need does arise, our little gadget will be indispensable. (Are you sure that the used mike you just bought is correct?) You can also check the drivers in any speaker system by using the line output to feed the power amp — or even the drivers directly if desired — and then using any mike, check to be sure the individual speakers all move in the desired direction. And, of course, you can always check cords with it, too! Operation is simplicity itself: plug in a mike, point it at the built-in speaker, and push the button. An LED will light showing whether pin 2 or pin 3 is positive for a forward pressure on the diaphragm. Battery operation makes it portable and nearly pocket size. Additional batteries can even be included for phantom powering condenser mikes and should last for years since they'll only be used for five or ten seconds at a time. About the size of an AA cell, these 22½-volt batteries should be available at any well-equipped camera supply store.

How It Works

When the pushbutton switch is activated, a single brief voltage pulse is applied to a small loudspeaker which is used as the "sender." By connecting the speaker's plus terminal (marked with a red dot of paint or a "+" symbol) as shown, the diaphragm will jump forward creating an increase in pressure in front of the mike under test. When the microphone is plugged in, the signal wires are then connected, each to its own comparator/detector circuit that can sense a positive voltage as small as 5 mV. This method, while elegantly simple and straightforward, does have one potential shortcoming in that most mechanical transducers will continue to vibrate or "ring" after the application of a single pulse. Since a certain amount of this ringing is inevitable in any speaker and especially the little two-incher used here, the microphone will often put out more than one cycle of voltage making it important to determine not only which pin went positive, but which went positive *first.* The mikes themselves may also ring as will many of the loudspeakers that you try to test. To combat this problem, a digital "lock-out" circuit was designed that causes the first comparator that is triggered to prevent the other's LED from flashing. Only two common, inexpensive ICs are used in this project and the total parts cost including a case should be under $20.00.

You will notice that non-polar capacitors have been specififed for the tester's input since they may have to swing either way. If an unbalanced mike is plugged in, one input may be grounded; yet when a phantom supply is connected, the input terminals will be more *positive* than the rest of the circuit applying a reverse voltage across the caps. See how often polarity pops up! These capacitors are not that rare, though if you can't find any, it is not very difficult to make your own using two ordinary electrolytics of twice the desired value. In this case you would connect two 22 uF 50-volt units in series — either plus-to-plus, or minus-to-minus — it doesn't matter which. In fact, this is all that's inside one of the non-polar types that you buy, only it's already done for you in one neat package. Also, it should be pointed out that while unbalanced mikes can indeed be tested with this circuit, it is important that neither the hot wire nor the shield should be connected to pin 1 which is

Ethan Winer is co-owner and chief engineer at the Recording Center, in Norwalk Connecticut, where he records, arranges, and performs on commercials and film soundtracks done by the Center's in-house production company. He also teaches recording there as well as at two area universities. Ethan informs us that when he's not engineering or teaching, he can be found fighting with local and state traffic department heads over the "horrible things they do with traffic lights."



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- High resolution, multicolor VU bar graph displays featuring average, peak, and peak hold modes for both level monitoring and spectrum analysis.
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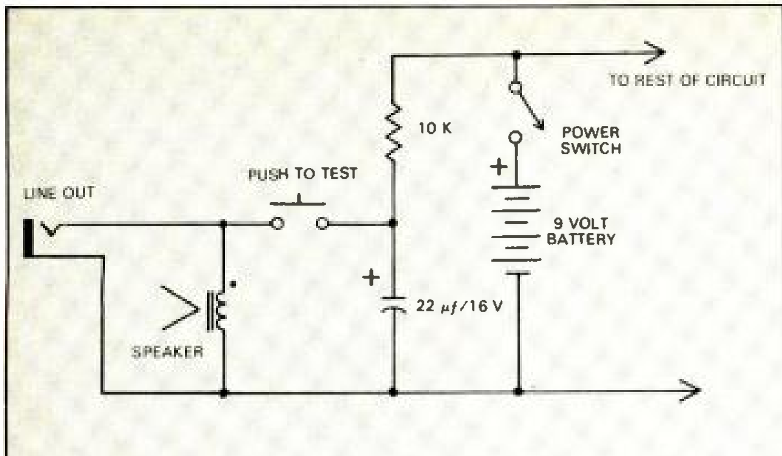

Figure 2: BATTERY AND "PULSER"

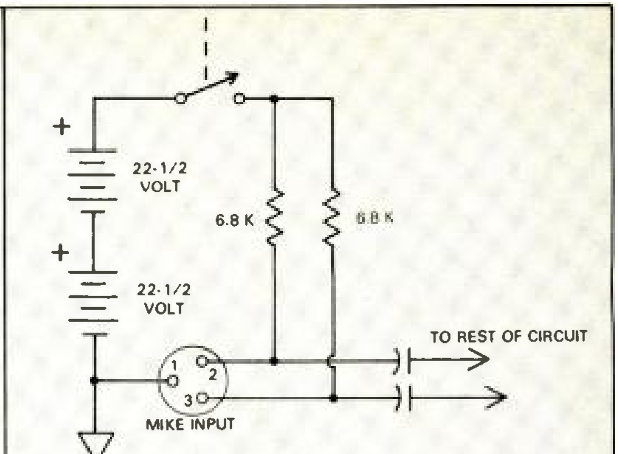

Figure 3: OPTIONAL PHANTOM POWERING

ground. While the non-polar caps will prevent any damage to the tester, the polarity being indentified is between pins 2 and 3 only, and with a phantom supply connected, you would be trying to power a mike that wishes you wouldn't. About the batteries for phantom powering; connect them as shown in Figure 3 using a double pole switch for power on off so both supplies can be shut off at once. Actually, you don't really need a switch for the phantom supply at all as long as you remember not to leave a mike plugged in when it isn't being used.

Construction is not difficult, though if you are trying to squeeze the whole thing into a fairly tight package, you might be careful to make sure that everything will really fit as intended before you actually start drilling holes. One wiring item that you will want to be careful of, however, involves the speaker's negative lead. This should be connected directly to the minus of the 22 uF pulse capacitor as shown in Figure 2. If the current surge is allowed to flow through the circuit's ground wiring, there's no telling what might happen. For the .1 uF detector capacitors, it is best to use either mylar, polystyrene, or some other low leakage type due to the high impedance of that part of the circuit.

Use a speaker of 8 ohms or less to be sure of getting a sufficient signal to the microphone being tested. If you use the speaker recommended in the parts list, you will be assured of adequate output though I'm sure many other would be just as suitable. In fact, while I would just as soon avoid plugging any particular parts supplier, it is useful to be able to go into any Radio Shack store around the country and get the exact same part.

The speaker comes without any obvious method of mounting, though I have found that a fine bead of contact (not rubber) cement applied to both the gasket and the mounting surface will hold adequately.

Checkout And Use

When power is first applied, one of the LEDs will light for a few moments but will go out once all the capacitors have charged and the voltages have stabilized. Connect the microphone to be checked to the tester's input and aim it directly at the built-in speaker getting as close as necessary to obtain an indication. With the majority of dynamic types, you will need to be within an inch or so. Most condenser models have a greater output on the average and won't need to be that close. Be assured, however, that the tester will never give you a wrong answer (assuming, of course, that you build it correctly), since neither LED will light if the mike isn't close enough. □□□

Reference:
1 - *Impulse Alignment of Loudspeakers and Microphones, Parts I and II*, R-e/p, Dec. 1978; Feb. 1979.

POLARITY TESTER PARTS LIST

Case
2¼" Miniature Speaker (Radio Shack part #40-246)
9 Volt Battery
XLR Connector, 3-pin Female
¼" Phone Jack
Power Switch
Momentary SPST Push Switch
2 - 470-ohm ¼-watt 5% Resistors
2 - 1K
10K
270K
2 - 470K
2 - 10M
2 - .1 uF Mylar or Polystyrene Capacitors
2 - 10 uF 50-Volt Non-Polar (Radio Shack #272-999)
3 - 22 uF 16V Electrolytic
2 - Silicone Diodes (1N914, 1N4148, or similar)
2 - LEDs with mounting hardware
1458 Dual Op-Amp
4011 CMOS Quad Gate
For optional phantom powering —
2 - 22½ Volt Batteries (Eveready 505, Radio Shack #23-510)
2 - 6.8K ¼-watt Resistors

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Union/Industry "Recording License:"

THE NEW PRLA
(Phonograph Record Labor Agreement)

by Will Connelly
President/Producer, Star Jazz Records

Many people involved in the record-making business will tell you that you need a "license" from the American Federation of Musicians to make "legal" recordings, and that you will pay a "fee" for this "license."

You may already have guessed from the use of quotes that there is a divergence of popular opinion and the fact is: ***there is no such animal as a "recording license," and no "fee."***

What there is is a series of three closely related agreements that record companies and the AFM sign, and these agreements define the whole set of arrangements that come into play when a company hires AFM musicians and releases recordings containing their performances.

These agreements are the Phonograph Record Labor Agreement (PRLA); the Phonograph Record Manufacturers' Special Payments Fund Agreement (SPF); and the Phonograph Record Trust Agreement (PRT).

The PRLA lays out the ground rules for employment of musicians, and includes the wage scales and the 'work rules.' It also addresses such matters as what companies can and cannot do, what they must do and the administration of the agreement. It also mandates that the company be a party to the SPF.

From a purely practical standpoint, the PRLA fixes many of the "up-front" cost elements that a company has to consider in planning a new record.

The SPF requires record companies to contribute part of the money received for records it sells to a Trust Fund from which that money, after payment of administrative costs, is distributed to recording musicians. In theory, every musician who makes a record will receive an amount equal to his original *scale* earnings from the sessions over a five year period after the record is released. It also works this way in practice so long as record company contributions are sufficient.

The PRT requires manufacturers to contribute another part of the money they receive for record sales to the Musician's Performance Trust Fund. This money is distributed (again after administrative costs are paid) to local unions and other organizations who use the money to pay musicians who perform at free community concerts and similar public performances.

Both the SPF and the PRT involve cost to the record company, but not until after a record has been released and then in an amount that depends on sales income. The PRLA is the agreement that guides the day-to-day activity of working producers and musicians; oddly, very few of the people who are most directly affected by this important document have ever seen it, or know what it contains.

A new PRLA was negotiated between the 'record industry' and the AFM in November, 1979. Although all interested record companies, large and small, are invited to

BASIC SCALE FOR SIDEMEN (Double for Leaders/Contractors)

	Nov. 1, '79 to Oct. 31, '80	Nov. 1, '80 to Nov. 30, '81
Regular 3 Hour Session	$137.21	$146.82
1/4 Hour Overtime	22.87	24.47
1/2 Hour Overtime	45.74	48.94
Special 1½ Hour Session	90.56	96.90
1/4 Hour Overtime	15.10	16.15
1/2 Hour Overtime	30.19	32.30

BASIC SESSION CONDITIONS AND SIDEMAN WAGES (Double for Leaders/Contractors)

TOTAL TIME HOURS	USABLE MUSIC	SIDES/CUTS PERMITTED New Music	Sweetened	INDIVIDUAL WAGES FOR SIDEMEN 11/1/79 - 10/31/80	11/1/80 - 11/31/81
Regular Sessions					
3	15 min.	No limit	4	$137.21	$146.82
3¼	15 min.	No limit	4	160.08	171.29
3½	20 min.	No limit	5	182.95	195.76
3¾	20 min.	No limit	5	205.82	220.23
4	25 min.	No limit	6	228.69	244.70
4¼	25 min.	No limit	6	251.56	269.17
4½	30 min.	No limit	7	274.43	293.64
4¾	30 min.	No limit	7	297.30	318.11
5	35 min.	No limit	8	320.17	342.58
Special Sessions					
1½	7½ min.	2 sides	Not allowed	90.56	96.90
1¾	7½ min.	2 sides	Not allowed	105.66	113.05
2	7½ min.	2 sides	Not allowed	120.75	129.20

Above rates, wages apply from 8:00 a.m. to midnight weekdays, and from 8:00 a.m. to 1:00 p.m. on Saturdays when the day is not a holiday (see PRLA U.S./Canada Holiday list). At other times, apply following multiplier factors to all wages (leaders, contractors, sidemen):

Midnight to 8:00 a.m. weekdays; 1:00 p.m. Saturday to 8:00 a.m. following Monday, excepting holidays: Factor 1.5

Weekday holidays from 8:00 a.m. to midnight, Saturday holiday from 8:00 a.m. to 1:00 p.m.: Factor 2.0

All other holiday time: Factor 3.0

Sessions which cross the boundries from one time frame to another must be computed pro-rata for the various rates involved. Work up from basic scales.

DOUBLING: 20% of full session wages for first double by each individual, 15% of full sessions wages for each successive double by same individual. See PRLA doubles list.

DUBBING, TRACKING, SWEETENING: Generally equivalent to adding a sideman for each added musician service (but see PRLA for exceptions and details).

HEALTH AND WELFARE CONTRIBUTION: $3.75 per person per session.

PENSION FUND: 10% of total wages (including overtime, premium time, doubling, tracking, but excluding H&W contribution and Cartage).

CARTAGE: See PRLA for list of instruments for which cartage must be paid.

Figure 1: SUMMARY OF COSTS AND WORK RULES UNDER PHONO RECORD LABOR AGREEMENT

Brandon Wades Crosstown Recording, Kalamazoo, Michigan.

attend and to participate in the negotiations with the union, the new PRLA (and nearly all other union agreements) was negotiated on behalf of the *'industry'* by the lawyers from a relatively few of the major companies. Smaller companies either could not afford to send representatives to New York, or were content to let the big guys speak for them. For whatever reason, smaller companies do not get involved in the negotiations, they still wind up being bound by the agreements if they become ***signatories*** to the agreements — which they will be if they want to make *'legal'* (in the union sense, not a statutory sense) records with AFM member musicians.

When a company signs the PRT, it is (for practical purposes) required to send a minimum $100 deposit against its future obligations under that agreement. This is the loot that is commonly referred to as the "license fee" which, of course, it is not.

So the PRLA is the key document from the day-to-day operational viewpoint, and it is something that companies and their producers really need to understand; they can be sure that their musician employees have a pretty good feel for at least the wage scale and work rule parts of the agreement!

Although the new PRLA was negotiated in October of 1979, the actual document (a 56-page booklet which also includes the SPF agreement) was not printed and sent to record companies by the union until mid-August, 1980. This delay occurred because, after the eyeball-to-eyeball negotiations between all the lawyers, disputes arose on how the verbal agreements would be translated into the arcane jargon of legal agreements, and what everyone had actually agreed to. These disagreements were not resolved until mid-summer. The PRLA, however, is retroactive and covers a twenty-five month period that began on November 1, 1979 and continues until November 30, 1981. The major differences between the new PRLA and its immediate predecessor are:

- Scale wages increased by 8% for the first year, and by another 7% for the remaining thirteen months, for a total cumulative increase over the whole period of the agreement of 15.56%.
- New sections are included that relate to the recording of classical music forms, including chamber music.

Other provisions make changes in the company contribution rate for Health and Welfare; modify some scale bases; change some of the instrument cartage rates; and increase per diem rates for orchestrators.

But let's, as they say, 'take it from the top.' The text of the PRLA will appear in my forthcoming book, **The Musician's Guide To Independent Record Production,** (to be released in January by Contemporary Books, Chicago), and this review will not cover everything that's in the full text. However, it should cover most of the points that are of interest to producers who are doing new records with live musicians (as differentiated from those who are re-releasing old material, air shots, or other previously recorded music).

The parenthetical numbers in the discussion that follow are the clause, or article, numbers of the PRLA.

Coverage (1)

For all practical purposes, the PRLA covers the services of the U.S. and Canadian musicians who record in those countries, their territories and possessions. It also covers certain services that musicians perform elsewhere. The term "musician" means not just instrumentalists, but also includes leaders, contractors, copyists, orchestrators and arrangers of instrumental music. It is important to note that singers/vocalists are ***not*** included: musicians are paid under the PRLA only for the services they perform with respect to instrumental music, and if they sing in addition to playing, there is no extra compensation for doing so. Generally speaking, vocalists are members of another union, the American Federation of Television and Radio Artists (AFTRA), which has its own version of the PRLA.

The PRLA does not prohibit the use of non-union musicians: it simply imposes on the company the obligation to be sure that any non-members become AFM members within 30 days after they are first hired (12). This is in the U.S.; in Canada, companies can only hire AFM members (11). In all cases, musicians must be members in good standing, i.e., not under suspension or on an unfair list. Producers can check the standing of musicians they're planning to use with the local union in which the recordings are going to be made.

Payment of Scale (3)

The company agrees to pay its musicians at least scale wages, which are discussed at length in Exhibit A of the PRLA. Obviously, a company can (and may well have to) pay more than scale and in this connection, it's worth noting that companies generally file recording contracts at scale wages even though they may pay the players more, the difference being covered by a separate check. The reason for this is that the company is under no obligation to pay Pension Welfare Fund benefits on the over-scale portion of the agreed wage, and usually prefers not to disclose the amount of over-scale it is paying to third parties (Exhibit B, (1)).

Notification of New Releases (5)

Companies are required to notify the AFM monthly of all new releases, and to send a copy of the release on request. The importance of this clause to a producer is that the AFM does run spot checks on record *times*, to insure that the amount of music released does not exceed that for which the musicians have been paid under the work and use rules relating to session scale.

No Commercials, Accompaniment Records (7)

The PRLA does not cover the recording of radio/TV commercials; these fall within the purview of a different ("Jingle") AFM agreement. This clause also prohibits making records for entertainers to use in their acts (i.e., sing-along records that would be used instead of live musicians in a live performance).

Musician Services (11, 12)

These clauses deal with such matters as what happens in the event of strikes, lockouts and the like, and also addresses the requirement that musicians be (or become) members in good standing.

Recording Contract Approval (13 (b))

When musicians are hired for sessions, the basic hiring is done by means of the AFM's Form B Recording Contract. In theory, this contract is always filed in advance, and on that theory, the AFM reserves the right to veto contracts before services are rendered. In the real world, which varies from one local union jurisdiction to another, recording contracts are usually not even typed up until after all the work is done and the producer has the information necessary to complete the contract form. In this real world, producers are exposed to a mild risk of retroactive disapproval of the contract. In Local 47 (Los Angeles), planned sessions ***must*** be reported in advance and a telephone is manned day and night to facilitate just such reporting; failure to pre-report can put the musicians (including the contractor and leader) and the producers into very hot water with the Local, which is engaged in a vigorous war against "bootleg" sessions. In most other locals, no problems arise if sessions are reported promptly after the fact.

Union Rep At Sessions (14)

AFM or local union business representatives must be admitted to studios during sessions expressly to conduct union business. I am not aware of any abuses of this discretion, but a producer has the obligation to be sure that any such visit is for a business purpose; it is, after all, his budget that is paying for musician and studio time.

Audit (15)

While this does not directly affect producers, it is a very important part of the PRLA that the AFM has the right to audit the company's books.

Limited To Phonograph Records (18)

The PRLA deals only with the making of records (including tapes). Other agreements cover video, radio, live performances, jingles, and other aspects of musical performance.

Other clauses not mentioned are important, but are mainly important on the business-side of the record-making house, or to producers who are working with "old" (previously recorded) music. Matters addressed include licensing, assignment, transfer and sale of masters; assistance to third parties; transfer of the Agreement; and termination in the event of a change in control of the company (as, for example, through acquisition or merger).

Exhibit A of the PRLA is the part containing the wage scales and the related work rules. Section I relates mainly (but not entirely) to performing musicians, Section II wholly to Arrangers, Orchestrators, and Copyists.

A further subdivision occurs in Section I between the rules and rates peculiar to recording symphonic music and those relating to everything else. On the premise that any company or producer who's going to hire a symphony can damned well afford a lawyer to interpret the agreement, I will address only the parts of Section I that relate to the non-symphonic music that dominates recorded music sales. So what follows

. . . continued on page 70 —

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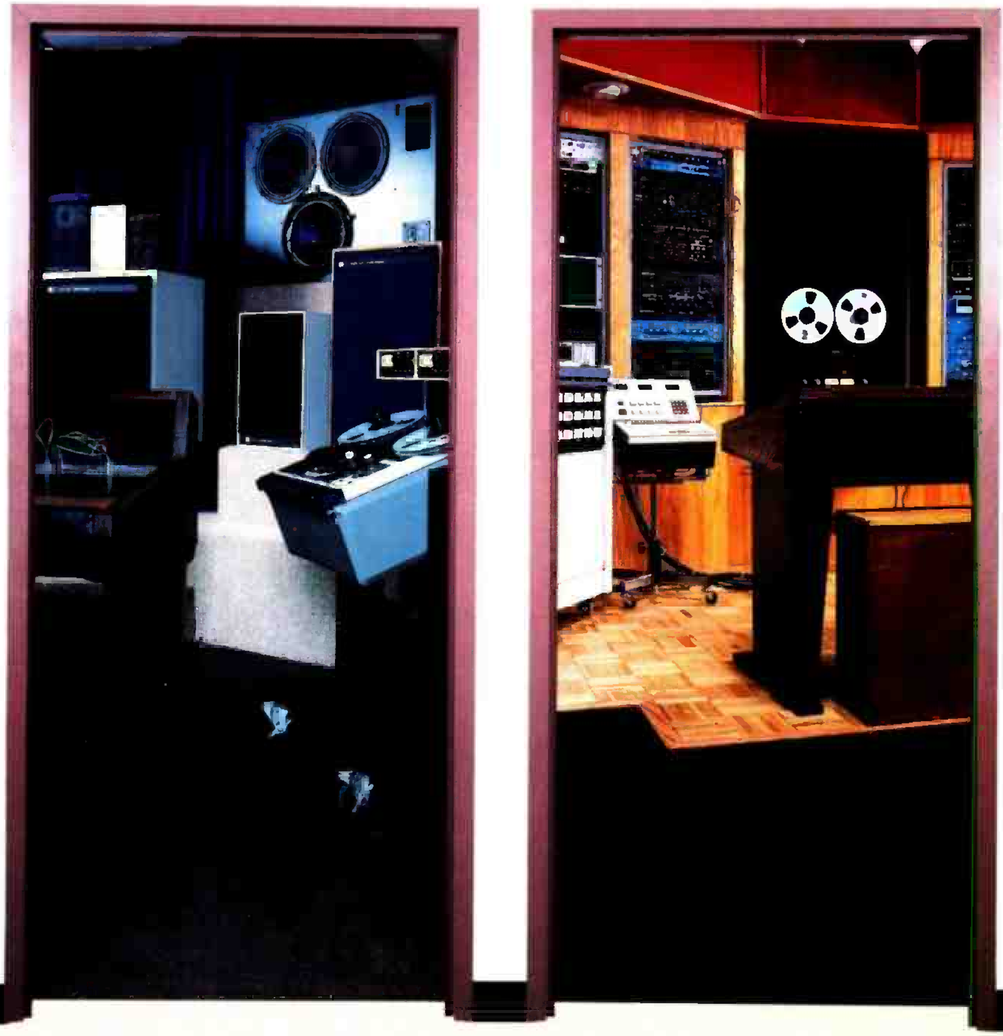

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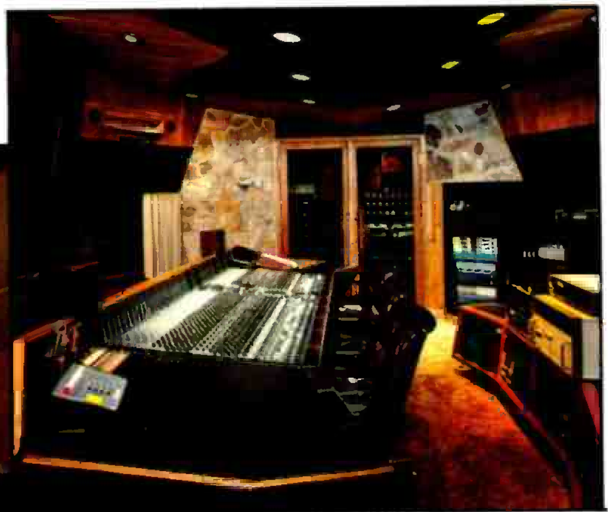
Electric Lady Studio

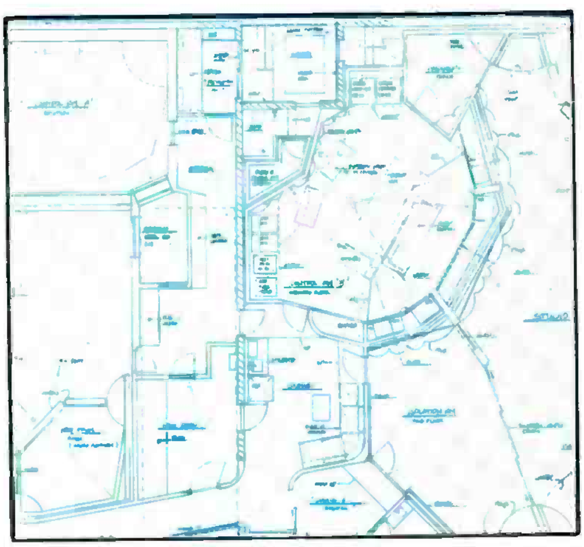


Westlake show room & facilities in Los Angeles exemplifies Westlake's continued dedication to the state of the art.

Westlake's HR-1 Phase Coherent 4 way monitoring system, as well as "Controlled Travel Path"T.M. acoustic design assures accurate recordings.

Radio & Records uses its Westlake Production room for multi-media and syndicated radio production.

Designing For Innovation, Quality a

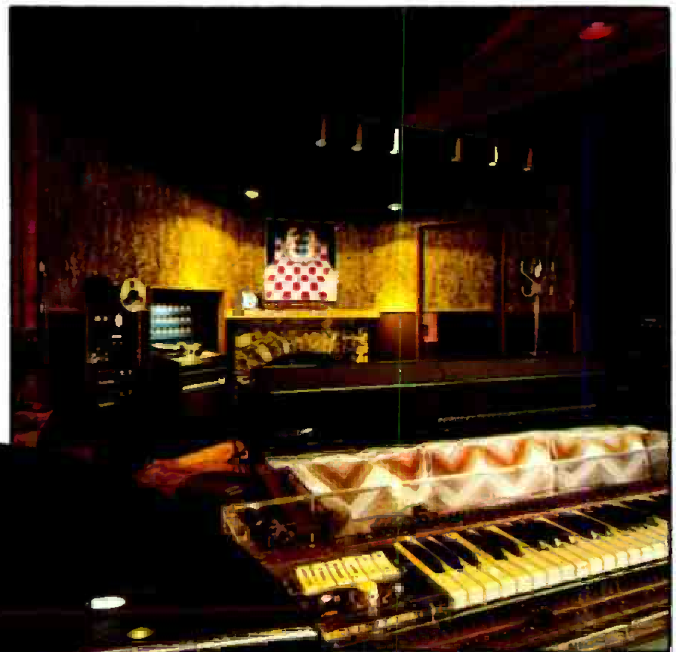


Our detailed design leaves little doubt what the results will be. Westlake has "pioneered" guaranteed acoustical designs.

In taking the state-of-the-art to new limits, Westlake has developed many proprietary products.
The D-1 Active Direct Box with 106 dB of dynamic range is a product of that research and development.

As we move into the 80's, television sound will take on new dimensions. Westlake recently completed renovation of the KCET (P.B.S.) facilities in Los Angeles.

The professional home studio is no stranger to Westlake. People such as Danny Serephine, George Duke, Michael Lloyd, and Georgio Moroder all chose Westlake to create their professional listening and recording environments at home.



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Consider our reputation! With more than 100 major installations, Westlake has many satisfied clients. So whether you are looking to purchase recording equipment or a complete turn-key studio, Westlake's professionals are ready to earn your confidence.

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"Recording License"
THE NEW PRLA
(Phonograph Record Labor Agreement)

applies to the recording of rock, jazz, country, hillbilly, fusion, and all the other stuff broadly categorized as "pop" music.

There are two kinds of Basic recording session: Regular and Special.

Regular Session

A Regular session is three hours long, and from each regular session the company can use 15 minutes of the music recorded. If a session goes very well, and it turns out that the company is able to finish a 30 minute album within the three hour period (it may surprise some of you to know that that's not impossible), the musicians get paid for *two* sessions, even though they all leave the studio (in some cases to work someone else's session!). In other words, the company pays for either the amount of time that the musicians are present at the session, or for the number of minutes of music it extracts for release, whichever results in the higher payment to the musician.

The base wage scale for a Regular session is $137.21 for the three hours minimum session period. That's the least anyone gets for the session. We'll get to extras shortly.

If a Regular session is to be extended, the producer has three options:

- He may extend the session by 1/4-hour, with no extension of the 15 minutes of usable music.
- He may extend the session for one increment (or more, if he has informed the musicians of the possibility of overtime at the time of hiring) of 30 minutes from which he may extract 5 minutes of music.
- Or, if he gives the musicians a half-hour break between sessions, and if the musicians are willing to continue, he can schedule another full three hour session, which allows another 15 minutes of usable music.

The base wage scale for overtime on a Regular session is $45.74 per half-hour, which works out to twice the rate for the straight time portion of the session, but which also permits usable music to be taken at twice the straight time rate (except for the quarter-hour increment, which can only be used once per session and provides no added usable music time; a quarter-hour of overtime is $22.87 base).

Which option a producer takes depends upon how much more time he thinks will be needed to complete the record and his estimate of how fired-up his musicians are. The union rules force a firm decision based on matters upon which the producer can only make something between an informed judgement and a wild-assed guess.

Special Sessions

Regular sessions can be used for making albums or singles or for sweetening (about which more later). There is also a Special basic session that is designed to permit cutting two sides of a single. The Special session is 1½ hours long and permits the use of 7½ minutes of music. The special session cannot be used for sweetening and the maximum permissable overtime is one-half hour, payable in quarter-hour increments.

The base wage scale for a special session is $90.56, and overtime is $15.10 per quarter-hour (essentially the same as the straight time rate).

Double For Leaders, Contractors

Leaders and contractors are paid at double the base wage scale. This applies to both straight time and overtime.

Health and Welfare

At each session, each musician (including the leader and contractor) earns $3.75 which is paid to the local union *if* the local has set up an H&W plan/fund, or directly to the musician if the local hasn't set up such a plan/fund. An additional payment for H&W is not triggered if the session runs into overtime.

Premium Rates

Basic session (Regular or Special) wage scales mentioned above apply only if the session is held entirely (from start to finish) between 8:00 a.m. and midnight on a weekday that is not a holiday, or between 8:00 a.m. and 1:00 p.m. on a Saturday that is not a holiday.

If the session is held entirely between midnight and 8:00 a.m. on a non-holiday weekend, or between 1:00 p.m. and midnight on a non-holiday Saturday, or on a non-holiday Sunday, all of the base wage scale and overtime rates are increased by a factor of 1.5.

If the session is held entirely on a holiday (New Year's, Labor (Labour) Day, Thanksgiving, and Christmas in the U.S. and Canada; Washington's Birthday, Memorial Day, and Independence Day in the U.S. only; and Good Friday, Easter Monday, Victoria Day, and Dominion Day in Canada only), all of the base wage scale rates *double*, again including the overtime rates.

If a session falls partly in straight-time and partly in premium time and/or on a holiday, the wages paid are prorated so that, for example, on a Regular session that started at 10:00 p.m. and ended at one o'clock the next morning (and assuming that the morning was not a holiday or a Sunday), then two hours of the session would be paid at the base wage rates and one hour would be paid at the 150% premium time rates.

These "golden time" rates are cumulative. If a session were held on a Christmas that was also a Sunday, the multiplying factor would be base wages x 1.5 (for premium time) x 2.0 (for holiday), or *triple scale.*

Breaks (Rest Periods)

Musicians are entitled to two ten minute breaks in a Regular session, one ten minute break in a Special session and 5 minutes during each hour of overtime.

Sweetening

You can sweeten four LP cuts or singles during a Regular session, and one cut or single for each half-hour of Regular session overtime. No sweetening is allowed in Special sessions.

Contractors

If a session involves twelve or more sidemen (not counting the leader), a contractor must be hired. The contractor can be one of the players other than the leader, or a non-playing AFM member; but the contractor cannot be the producer or an engineer on the session or a representative of the company even if he is an AFM member.

Advance Notice Of Sessions

If a company knows in advance that it is going to hold a session, it ***must*** give advance notice. This is a somewhat elaborate clause but its requirements can be met easily enough by calling the local union business rep and answering his questions when you know about a session.

Cancellation Of Sessions

Once a session has been called, it cannot be cancelled, delayed, or re-scheduled on less than seven days notice without the permission of the AFM, New York. This is a very strong incentive not to schedule a session until it's absolutely certain to come off.

Doubling

When a musician plays a second instrument that is considered a double (piano and celeste are not, a twin-neck guitar is). For illustration, the player earns 20% over scale for the first double and 15% over scale for each additional double. This doubling surcharge applies to all session wages (overtime, premium time, holiday time) even if the musician only doubles for a brief period in either the ordinary part of the session or the overtime.

If a player is asked to bring an extra instrument for a possible double, but he doesn't wind up playing it, he earns $5.00 for having brought it along. This is independent of cartage.

Location Recordings

The rules for non-studio recording are a little complex and I won't recite them. If you're planning to record at a club date or a concert, call the local union business rep and get the costs from him.

Cartage

The company pays for cartage at the higher of either the actual transportation cost (when the instruments are hauled by a common freight carrier) or $6 per instrument for: Accordion; string bass; Tuba; all amplifiers; baritone sax; bass sax; cello; contrabassoon and contra bass clarinet. For harp, the cartage fee is $26, increasing to $28 November 1, 1980.

Payment

Pay everyone within 15 days of the time you receive the Form B contract and W-4 tax forms for all the musicians and you will avoid the penalties that are specified.

Overdubbing, Tracking, Sweetening, Multiple Parts, Etc.

These somewhat convoluted rules boil down to this: if a musician re-records during a session to sweeten or to add a harmony part to his earlier recording, or in any way winds up doing what the producer would have otherwise hired some other musician to do if he weren't there, the producer will wind up paying him exactly as if he were another musician. The only non-artistic benefit is that you may wind up not running into the point at which you have to hire a contractor. As noted at the beginning of this article, though, no one gets anything extra for vocal work (at least not under this agreement).

Royalty Artists

A "royalty artist" is an instrumentalist who has a deal with his record label providing for a royalty of 3% or better, or who is part of a recognized group which jointly receives a royalty of 3% or more of a record's list price (after adjustments). This doesn't include sidemen or others who don't share in the royalty earnings.

The payment of royalty artists is simplicity itself. For each song they record, they receive the Regular Basic Session base wage scale. There is no doubling for leader, no overtime, no premium or golden time pay, no addition for doubling, tracking or sweetening.

This arrangement is not optional under the PRLA; but no particular brilliance is required to work out an alternative arrangement that both complies with the agreement and avoids what could be higher up-front cost than a company may want to invest. Whether the royalty artist deal or the customary wage and rules deal is better depends upon the facts of every particular production and such elements as the degree of player skill and rehearsal of the material involved. For super-famous groups or stars, it's all academic anyhow.

Section II of PRLA Exhibit A, covering the Arrangers, etc., is simply too complicated to review here, and the review wouldn't help, anyhow; when all's said and done, what you'll pay for charts is chiefly going to be determined by counting the written bars or by some fixed quoted price that will be well over scale, anyhow.

Exhibit B of the PRLA deals with one item only: the AFM Pension Welfare Fund. What it boils down to is that the company pays 10% of all earnings in addition to everything else. The 10% applies to all wages, including the overtime, premium, and golden time earnings and to doubling and earnings resulting from sweetening, tracking, dubbing, etc. It does not apply to the H&W contribution or to cartage, neither of which are 'earnings.'

In addition to everything that's stated in the PRLA and its exhibits, there is one other major cost that is not mentioned directly, but which applies nevertheless: payroll cost. The Form B contract that is signed with the leader or the contractor is an employment contract, and the people who are hired become the company's employees. The company is responsible for the actual acts of and the cost burdens associated with withholding income taxes; FICA contributions; Workman's Comp, and a variety of other expenses. In some local union jurisdictions, these costs are handled by working through special paying organizations (e.g., Crew and Cast in Los Angeles, Paymasters via the local in Miami) who take care of all the necessary paperwork and submit a bill. These excess costs typically add another 13 to 16 per cent of base wages (only a small fraction of which is for the payroll service: the bulk of the loot goes to appropriate taxing authorities).

The employer/employee concept is reinforced throughout the PRLA.

This article should be taken as for guidance only. There is a great deal in the PRLA and its Exhibits that has not been covered here, and there are some loopholes that I have not discussed because they apply to very special situations.

Figure 1 summarizes the nitty-gritty of Exhibits A and B. I would be very much surprised if there are many after midnight, Fourth of July big-band recording sessions held. □□□

THE WONDROUS WORLD AND VOCABULARY OF THE AUDIO/VIDEO SWEETENING SESSION

by WINN SCHWARTAU
Empirical Audio
Ossining, New York

A Very Typical Session . . .

Astor Productions signed the contract with MGM for their first feature film production almost two years ago and it took nearly that long just to complete the script (as bad as it was) and finally (a big sigh) complete the film shoot and street scenes to the overly finickey producer's satisfaction. M. Fitzgerald had the reputation of standing for nothing less than the best; but his filming was a compromise every inch of the way. The technical crews were plagued by problems at each turn and the problems were compounded by the general unease created by the mood of Fitzgerald. They lost **camera sync** more times than they cared to remember (their audio tape recorders had more **wild** dialogue than not) almost all of the audio recorded **live** had to be **redubbed** or **lip synced** at a later date.

It only took Fitzgerald and his crews a couple of weeks to finish the **film editing**, and to everyone's surprise, he was actually satisfied with the film — at least visually. The sound was atrocious! By transferring the **Nagra** recorded tapes onto a string of **film dubbers**, they had attempted to achieve some sort of **sync** with the picture, but only a few seconds of audio could be expected to stay in lock with picture because of the bad camera sync on the shoot. Frustrations reached a fever pitch, and it was decided to go into a recording studio and virtually start from scratch. At one point, even, **Vidimag®** was considered, but again rejected because the general sound quality, even if it were in sync, was less than acceptable. Going with an A/V interlock using **SMPTE** seemed the one way to solve all of Fitzgerald's problems.

The Booking

Sally Hughes, Fitzgerald's assistant, began the arrangements to find a **post production** studio which could handle their project in the required time scale. (They were already late on delivery.) She began with the **video production** houses

What It All Means . . .

SYNCRONIZATION: The process of accurately controlling the position and dynamic relationship between two or more audio or video machines or other storage media.

CAMERA SYNC: The vertical sync signal from the camera itself. The sync may be internally crystal generated within the camera, or may be generated from an external source and distributed to a number of cameras. (See: V-DRIVE)

WILD: Audio tape (or video) recorded out of sync, or without a sync reference to the video or film picture. The audio track is used in post production and must be re-synced with the picture.

LIVE: Audio recorded at the actual time of filming. Not overdubbed in a studio situation.

REDUBBED/LIPSYNCED/ELECTRONIC POST SYNC (EPS): An audio recording in the studio where additional sound is added to the picture. The recording of actor's dialogue at the same time picture is being shown. Sync signals are added at this point for accurate synchronization in the mix.

FILM EDITING: The mechanical process of shortening or lengthening particular film segments to the producer's taste. The overall process of creating a completed film segment.

NAGRA: The mono or two track audio tape machine which the film industry uses as their standard for remote recording. The Nagra has the capacity for locking to cameras, generating its own sync, being RESOLVED and running on batteries.

FILM DUBBERS: An audio tape recorder which uses either 16 mm or 35 mm sprocketed magnetic film. The machines are available in mono (STRIPE), 3 track or 4 track formats (FULL COAT). In the FILM MIX, the picture on film, is synchronized with the dubbers through a Sel-Sync motor drive. The dubbers (up to 30 or 40 is not uncommon) are interlocked with the Sel-Sync drive system. Each dubber is treated as a single track on a multitrack machine would be, but with the advantage of being able to move each sound element on each dubber in time with respect to the overall audio mix and picture. If the original audio was recorded without a sync reference, the audio on the dubbers would only stay in lock for a few feet. The only way around this is to "Ride-Sync;" a manual (and time consuming) method of retaining a "marginal" lock with the picture.

SYNC: An abbreviation for synchronization. (Audio equivalent is Sel-Sync or selective channel synchronization). Staying in SYNC means that all machines are moving film or tape at the same speed with reference to each other. Sync may also refer to the particular type of synchronization technique used. (See: V-DRIVE)

VIDIMAG®: A 35 mm sprocketed film system which permits the recording and playback of video signals. Vidimag® will interlock with Film Dubbers and permit a conventional film mix facility to effectively mix the audio for a video production.

POST PRODUCTION: The process for completing an audio/vidual production. The synchronization, mixing, dubbing, and recording of audio signals for a video/film project.

VIDEO PRODUCTION: The taping, editing, dubbing, gen-locking, etc., of video signals from a number of sources. Generally several video tape machines are computer controlled to create a visual montage and single video output. CMX editing equipment is common in large video installations.

in town and they could certainly take on the work for the **Telecine, video-dubbing,** and **regeneration** of code necessary to achieve an **audio-video time lock,** but they didn't have the audio recording facilities required for large dates, sound effects, libraries, and the subtleties of an automated multitrack audio production. The audio production, as viewed by both Sally and Fitzgerald would require some 16 - 22 tracks and a reasonably sophisticated mix.

One of the video houses Sally spoke to referred her to a large recording studio, which also had some video, post-production services available. Sally called the studio and spoke to the studio manager, Susan S.

"Hi, Susan? This is Sally Hughes from Astor Productions. Mort over at the video house said you could handle some audio **sweetening** for us on a feature film we are finishing up."

"Sure can. I'll just need a few details so we can get all the equipment together for you."

"O.K. Shoot."

"Have you transferred your film to video tape yet?"

"No, not yet; but Mort has all the Telecine facilitites we need. What **format** would you like us to use, **quad?"** (Her TV experience showed.)

Susan glanced down the list of audio and video interlock formats their new system could handle. "We can lock with either **Beta** or **VHS**. But most of the engineers prefer to work with VHS on a ¾" **VCR**."

"Fine," replied Sally. "The film is all edited so we can transfer each reel to its own **cassette** on Mort's Telecine equipment."

"Are you going to be scoring everything here or do you have pre-recorded effects?"

"Actually both, Susan. The audio was an incredible problem. But we have to use some of what we shot anyway. The stuff from Europe made for the most trouble. By the way, can you **resolve** both 50 and 60 Hertz signals?"

"We generally resolve on our Nagras, if that's O.K."

"No problem. For the scoring, we'll need some sort of projection. But preferably not film. What've you got?"

"I'm glad you asked! We just

TELECINE: The machine which transfers film to video tape. The Telecine equipment electronically compensates for the frame rate difference between film (24 frames per second) and video (30 frames per second).

VIDEO DUBBING: Copying of video tape.

SMPTE GENERATOR: SMPTE stands for the Society of Motion Picture and Television Engineers. In 1972, a standard digital code was devised which would permit several audio or video tape machines to have a common reference. The concept of SMPTE time code being an "electronic sprocket hole" is quite apropos. The SMPTE code is a series of digital pulses which sequentially (with reference to the house sync or internal sync signal) identifies a particular frame of either black and white or color video. The code is 80 bits wide and each frame of signal has a unique code. The code identifies the frame, the time in hours, minutes, and seconds, and has a series of user and internal use bits. The SMPTE Generator is a machine which generates this code; either starting at 00:00:00:00, (no hours, minutes, seconds, or frames) or at any predetermined starting point.

REGENERATION: The copying of SMPTE signals directly from one tape to another is an absolute "no-no." Regeneration is done by actually reshaping the SMPTE signal itself back to an undistorted form and retaining the original code information. (See: JAM-SYNC)

AUDIO-VIDEO LOCK: Lock and sync in this manner mean roughly the same thing. An Audio/Video (A/V) Lock is the synchronization of two or more audio and video tape machines.

SWEETENING: The addition of audio signals to an existing soundtrack to enhance or finish the audio production. In a recording studio, a string of overdub sessions is also called sweetening. In the A/V post production, a sound effect (SFX) like the slamming of a door, could be called sweetening. Often, sweetening is done to picture.

FORMAT: In audio we call the number of tracks and width of the tape the FORMAT. In video systems, the manner in which the video signal is recorded, the width and speed of the tape, the number of audio tracks and control tracks, all contribute to the Format designation.

QUADRUPLEX: Quad recording was, until recently, the absolute broadcast video standard. The large 2" machines use four recording heads and penetrate tape transversely.

BETA - VHS - U: These three formats are the most likely types of video cassette recorder (VCR) configurations the studio engineer is going to run into on an A/V post production basis. First, they are in popular usage both commercially and industrially. Second, they are reasonably inexpensive so most studios can afford them, and the interface and operation of the machines is quite simple. The Beta system is the Sony ½" VCR recording system. The Beta tapes are recorded with what is known as the AZIMUTH RECORDING TECHNIQUE. The Beta formats are Beta I, II, or III. The Beta II is the consumer version of the Beta I, utilizing a narrower record head and a slower tape speed, achieving more recording time per reel. Beta I tapes will play on a Beta II machine, but not vice-versa. Most of the newer Sony machines have a Beta II/Beta III switch to decide the tape speed and format. The U-format, as in U-Matic VCRs, is a HELICAL SCAN TYPE of recording process, but does utilize the AZIMUTH RECORDING TECHNIQUE. The VHS (Video Home System) recording is also a HELICAL SCAN format, but differs from Beta and U in that the tape path utilizes the "M" wrap; two pins pull the tape around the head drum in the shape of an "M," so there is 570 degrees of tape angle. See manufacturers literature for the various models and technical specs for your applications. (See: AZIMUTH, HELICAL SCAN, QUADRUPLEX)

HELICAL SCAN RECORDING: A type of video recording in which the video heads and the tape meet at such an angle that the resulting pattern of information resembles a series of diagonal stripes on the tape. Each diagonal stripe contains one field of video information. (Two fields equal one frame.) The name Helical is derived from the shape of the tape path around the head.

AZIMUTH RECORDING: One problem with Helical Scan recording was the crosstalk that resulted between the audio and video channels. The solution found initially was the use of guard bands between the various layers of information on the tape. This, though, takes up space on tape and can reduce the overall quality of picture and sound. With Azimuth Recording the audio and video tracks are recorded several degrees out of phase with each other, thereby eliminating much of the crosstalk present. The heads are positioned to both record and playback the phase variant information.

CASSETTE: Just as audio has cassettes which permit an enclosed tape system; so does video. The cassettes are one direction only (as in certain audio formats) and have a tape width of either ½" or ¾". The tape within the cassette is looped around a head drum within the cassette machine by a mechanical arm.

RESOLVING: The process of playing back a pre-recorded reference signal from a tape, comparing that signal to some internal fixed reference in the resolver and creating an offset control voltage or frequency by comparing the difference between the two. The offset voltage ultimately determines the tape speed. In principle, resolving is similar to the system used with a servo system. Most early resolving was done with either a 14 kHz or 60 Hz reference signal. Other techniques of Pilotone and Neopilotone also became popular in Nagras and Magnatech equipment. (See: DC/FREQUENCY SERVO)

Empirical Audio

wishes to thank

SOUNDMIXERS

for their continued support of

TRIDENT AUDIO DEVELOPMENTS

Paul Sloman,
Executive Director of SOUNDMIXERS:

"SOUNDMIXERS was the first major New York recording facility to install a TSM. It wasn't an easy decision, but studio "B" is now our most successful room by far; primarily due to the Trident. Now it's an easy decision to go TRIDENT in studio "A".

TRIDENT AUTOMATED 4032 TSM RECORDING CONSOLE IN SOUNDMIXERS STUDIO "B" installed a new **video projection** system, and is it ever easy to use! The screen is almost as large as the old film projection system, but with greater **gain.** Your video signal in the control room just feeds into our **DA** facility and we can give you video anywhere in the house."

"Fabulous! So give us Studio B, if you can, for a couple of weeks. By the way, who's gonna be our engineer?"

"As I remember, you used to use R.O. over at NBC. Would he be O.K.?"

"Susan, you're a gem. See you next week."

The Setup

As has become the habit in the majority of large recording facilities, the booking office lays out all the requirements for a session on a daily log and the maintenance department uses this as a guide in setting up the session. For an audio/video lock, though, there is a little more required than just saying 16 or 24 track and what type of tape and bias are to be used.

Stephan and J.J. of the night shift were responsible for setting up the next day's session.

"These first lock up sessions are always a pain, J.J. They need all the equipment in the world to set things straight, then once we've laid the code sync, and achieved **lock,** it's just like any other Sel-Sync date."

"Yeah, I know. Let's get on with it. Let's patch the **house sync** feed into the **SMPTE generator** and check it."

"Fine. Did they specify **frame** or **drop-frame** code, J.J.?"

"They didn't, so let's use the **drop-frame,** as usual."

"We're getting **valid code**. The reader is coming back clean and green."

As the LED display on the SMPTE reader read out the code in hours, minutes, seconds, and frames, Stephan rolled in the Studer machine, which the client did specify. "So you want to use the **DC** or **frequency servo** cards, J.J.?"

"Well, if we let the Studio be the **slave** and the VCR the **master**, I vote for DC. By the way, which VCR are we gonna be using on this one?"

"Probably the **JVC**. The **Sony** still hasn't been modified for that **looping/unlooping** function. The

VIDEO PROJECTION: The aim of television system designers has been to create a large, bright, and exciting picture for industrial and home use. The video projection system is a midway step for the ultimate — flat screen TV. Basically, three projection tubes representing one of the primary colors are aimed and REGISTERED on a large reflective surface. These systems are usually fairly large and once setup, should be left untouched if possible. For studio applications, plugging a VTR into the projection system and pressing PLAY is a lot simpler than running a 35 mm projector from a projection booth.

GAIN: The gain of a screen determines how much light is reflected from the screen to your eyes. The more efficient the screen, the higher the gain and you can operate your video projector in higher ambient light environments.

DA (Distribution Amplifier): In audio a DA amplifies and isolates signals from each other and provides a low source impedance, high level output to feed either long lines, or a number of separate devices. In video systems, you can't simply "daisy chain" or parallel TV monitors, or loads as simply as we can with audio. So, in a complex layout, say between several VTRs and an array of monitors, it's easier to run each video source through a DA and then patch the desired interconnections.

LOCK: In this case, lock means that both audio tape recorders, or audio and video recorders are synchronzied; running at the same speed, and in the proper time reference to each other.

HOUSE SYNC: In studios or video facilities which have a number of different SMPTE generators, video sources and long lengths of wire from room-to-room, using one single master reference signal for all of the video equipment is desirable. The master sync generator will control all timing functions in each independent device and insure that all equipment is related to a single time-base frequency. (See: V-DRIVE)

FRAME: The video signal we see on a TV (or film for that matter) is not a stream of continuous motion. The video signal is broken into 30 discontinuous frames of information, each representing the equivalent of a still photograph. Each frame equals one complete TV signal and is further divided in half, into two FIELDS. Each frame has 525 horizontal lines of information which is scanned at 15,750 Hz. (That's the whistle you hear on your home TV or studio monitor.)

DROP FRAME/NON DROP FRAME CODE: There are two varieties of SMPTE code which are used in video or A/V facilities. The NON DROP FRAME code is the standard SMPTE code which will run sequentially from any given starting point and count 30 frames per second, all the time. The DROP FRAME code actually does NOT COUNT or OUTPUT from the generator frames :00 and :01 (once a minute, on the minute) except for 10, 20, 30, 40, 50 and 60 minute flags. This means that the DROP FRAME SMPTE generator would read 1:00:00:00 (1 hour) and the NON DROP FRAME generator would read 1:00:03:18 (1 hour, 3 seconds, 18 frames) if both generators started and stopped at the same instant.

VALID CODE: In SMPTE readers, (a device which detects and displays the SMPTE time code), there is an indicator lamp or LED which turns on when the SMPTE code being fed into it is of the right level and waveform. Occasionally when the tape has had dropouts, low level code, distortion, etc., the indicator will extinguish and potentially one can lose lock. There are techniques for generating new code when this occurs. (See: JAM-SYNC)

DC/FREQUENCY SERVO: A servo driven machine (either audio or video) is a self-correcting motor. A signal is picked up by a tachometer at the motor, compared with a known constant and a correction signal is sent to the motor to change its speed. This process is done hundreds of times per second, so the minute fluctuations which do occur in the tape speed are unnoticeable. Either a DC voltage is used as a reference for the motor speed, or a stable frequency base is used. All major multitrack machines use servos and the external servo input on the tape machine is the point which a synchronizer will hook up to control the speed of a machine.

SLAVE/MASTER: The MASTER machine in a synchronization setup is used as the reference for all other tape machines in the operation. The SMPTE code coming from the MASTER machine is compared to the SMPTE code coming from the SLAVE machine. The SLAVE machine receives the signals from the synchronizer which changes the speed of its tape travel, to match the position and speed of the MASTER.

LOOPING/UNLOOPING: On a VCR, the tape is pulled by a mechanical arm assembly from the cassette and around the head drum. In some machines this occurs every time you press PLAY (looping) or stop (unlooping). For the production engineer, this makes re-positioning the tape on the VCR a time consuming job. There are VCRs though, which do not unloop the tape from the head drum unless you want it to. Some machines, even permit variable speed shuttling of the tape across the heads and the retention of picture signal and sync. In a synchronization setup, the VCR must be able to do these functions, if the VCR is declared the slave.

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the Wondrous World and Vocabulary of the Audio/Video Sweetening Session

engineers say it slows them down too much. Let's play it safe."

The VCR is rolled in and the deck controls of both machines and the servo of the Studer are connected. The **keyboard** is placed on the remote table next to the console and the **brain** hidden away from sight. "Ready to give it a go?"

"Let's rock."

As the two maintenance engineers pressed the appropriate buttons, both the Studer and the JVC began moving tape. A slight whining sound could be heard as the Studer approached lock with the VCR.

"We have **frame lock** . . . Now. And we have **bit lock** . . . Now. That's a little slow, you know? How about tweaking the **slew** rate?"

"You're right, J.J. Engineers always want instant lock. Anything over a millisecond is too long for them. Tweak away."

A few tweaks and tries later, both maintenance folk were happy with a 6-second lock and ready to align the Video. "Got the **test tape?"**

"I've got it, but how about setting up the **monitor** first? We need to feed the DA and split the **Trini** and the **beam**."

KEYBOARD: On a synchronizer, all of the data entry and visual displays are generally on a single panel. This panel is the basis of the Man-Machine interface.

BRAIN: A generic name for the processing section of (in this case) a synchronizer. Generally, a maze of microprocessor based PC cards which control the operation of external equipment. Some manufacturers use this term in the description of their gear. (See: SOFTWARE)

FRAME LOCK: The conditon when two or more A/V recorders are playing back (or recording) in sync with each other. Each frame identified by the SMPTE code occurs at the same time on both machines.

BIT LOCK: The SMPTE code is broken up into 80 bits of information. A tighter or more accurate lock can be achieved, if instead of only locking the machines together on a "per frame" basis, we then let the synchronizer compare and lock to the bits of the codes as well. This type of lock is roughly 80 times more accurate.

SLEW: When two machines are attempting to lock together, the slave machine must vary its speed in order to reach the same SMPTE location on its tape as is on the master. The rate at which the machine changes its speed (either up or down) is the SLEW. Engineers, of course, want the machines to lock as fast as possible, so optimizing the SLEW to the particular machine is of great importance.

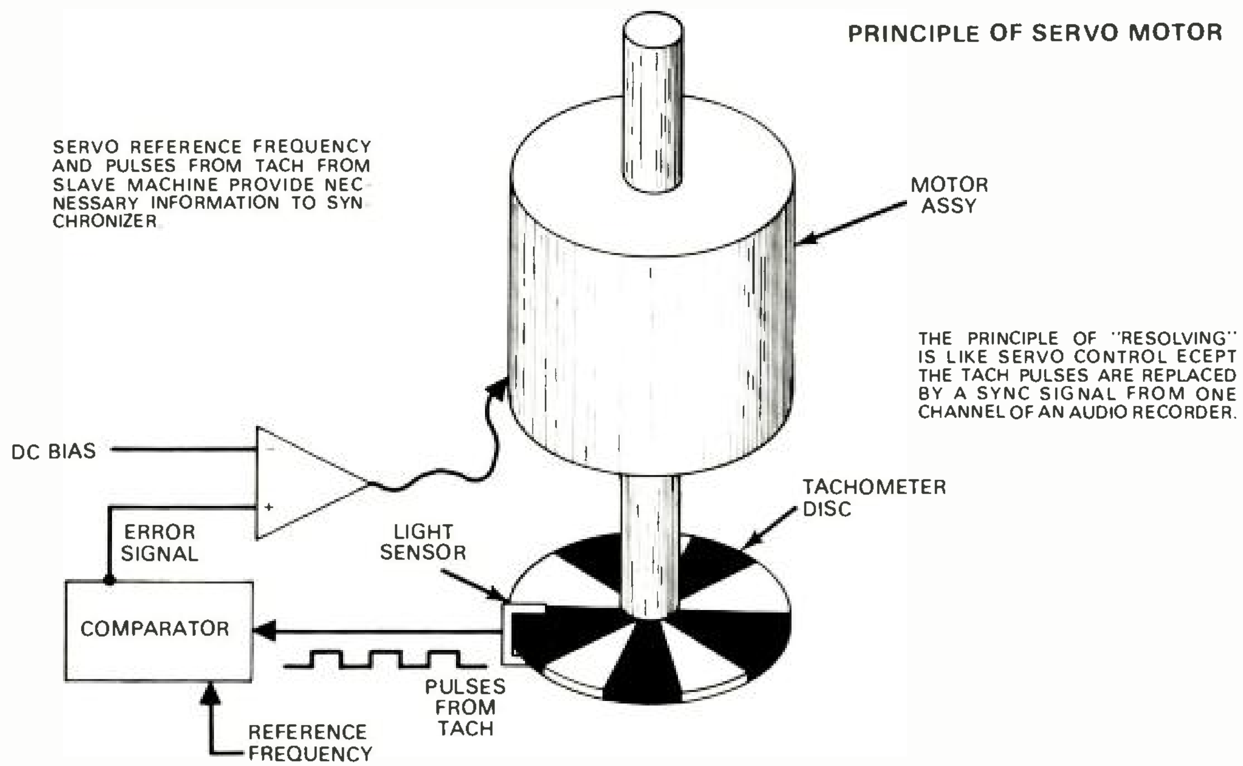
TEST TAPE: Video test tapes contain a variety of information to align video systems, just as audio test tapes do. They will include video images such as the familiar test patterns we see as stations go off the air, an array of dots on the screen to align for position and color addition and color bars for chroma adjustments. Most A/V production studios only have to worry about these factors once in a while; but in a broadcast situation, these alignments are as common as adjusting bias for a new roll of tape.

MONITOR: A television is a video monitor. There are, though, two types of monitors which you will encounter. One is the typical TV type receiver; you can feed an input to the receiver via the antenna terminals and tuning the receiver to either channel 3 or 4 (depending which part of the country you are in). The RF feed into the antenna from a VCR or camera contains all video, sync, and audio information on the same pair of wires. A monitor, though, in distinction to the receiver, has a separate COMPOSITE VIDEO input, and a separate audio input. There are also loop through connectors on the rear to have a video output to feed another piece of equipment down the line. Some monitors include a tuner for normal channel selection, as well as a VTR input; others are a straight monitor only with controls to adjust the picture itself, and without any sound.

COMPOSITE VIDEO: The total video signal including all sync information, (not SMPTE), but without any audio information.

TRINI: Shortened form of Trinitron, the Sony patented single-gun picture tube. Instead of using three separate guns, one of each primary color, a single electron gun is used which produces three meshed beams to reproduce the three primary colors on the phosphor screen.

BEAM: An affectionate name for a projection TV system.



. . . continued on page 81 —

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"Stephan, why don't we just loop through and move the termination?"

"Cause none of the **connectors** match up. Some got **UHF,** some have **BNC**. One day they'll standardize. The DA room has all the adapters built into the patcher. Besides, each load is terminated, and we have to use a DA."

Once the connections were made, the **NTSC** test tape went into the VCR and final alignment of the Trini and the beam were made. **registration** of the Trini was rarely needed, but because of the size of the screen on the projector, **barrelling** or **pincushioning** was easily seen and with the pickiness of the video people, you always checked.

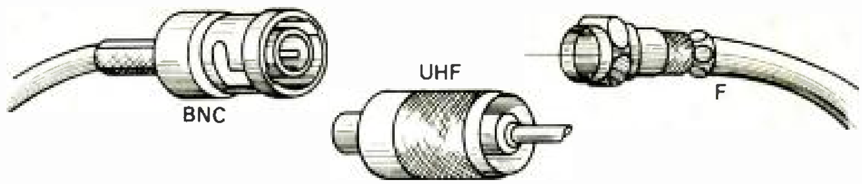
"J.J., have you checked **jam-sync** yet?"

"Done. I hope no one touches anything until the session tomorrow."

"Don't they always? All right, I'll take the mono record in Edit 2. You set up the tape copies in Production."

The Session

Fitzgerald, for all of his years of film making, (Cannes Reviews "A" Grade, American Sales "C" Grade) had never really gotten used to the idea of using the video medium for his audio production. The rows of dubbers in another room in a film mix were a safe and happy way for him to produce a film. He didn't believe in the video techniques used for typical television; **split screening, gen locking, character generator, negative imaging,** and the rest of the special effects that video **software** technology had recently developed, were a far cry from the hands-on **hardware** with which he had always worked. Sally Hughes, though, had considerable experience in amateur video production during college and her early job experience. She used to carry **Porta-Paks** around and video tape both in real time and slow motion. Her production courses offered her the chance to experiment with **insert editing** and **assembly** through the school's reasonably sophisticated **switcher** and **SEG's.** She felt completely



CONNECTORS: There are three basic connectors you will find in video applications: the BNC, the F, and UHF. The UHF connector is generally used with a composite video signal and is wired to coax type cable (RG-58, RG-59). The F type connectors and the BNC connectors are typical for interconnecting RF signals, carrying both modulated video and audio.

NTSC: National Television Standards Committee. A broadcast advisory group established in the 1940s to recommend standards to the FCC. They established the 525 line, 60 field television scan rate. (See: PAL, SECAM)

PAL: Phase Alternate Line scanning system used in the U.K., Western Europe, Scandanavia, Australia, and South Africa. 625 line, 50 field standard.

SECAM: A color broadcast system developed by the French; the Sequential Color and Memory system is vastly different than the NTSC and PAL color systems.

REGISTRATION: An adjustment for color TVs or monitors which aligns electron beams from the gun to the proper phospor dots or stripes on the screen.

BARRELING: Barrel distortion makes the TV picture look like a view through a wide angle lens. The edges are rounded and out of proportion with the center of the picture.

JAM-SYNC: A form of sync regeneration. SMPTE code is entered into the Jam-Sync generator. Instead of merely reshaping the waveforms of the pulse train, the generator actually recreates a new SMPTE code. The new code will have the same information as the old code, but may be delayed 1 - 2 frames for the internal processing time. Some Jam-Sync generators have a "look ahead" feature which will permit the continuation of code generation, although there is a loss of input code. This is quite useful when tapes have been mangled or erased, and new code is required to finish an interlock project.

PINCUSHIONING: Pincushion distortion is almost the exact opposite of Barrel distortion. The picture distorts outward at the corners.

SPLIT SCREENING: A technique allowing multiple images to appear within the area of the TV screen. Many varieties of screen split are possible; having two separate source images vertically adjacent or even a circular or square insert in one portion of the picture. Most SEGs offer a number of split screen arrangements.

GEN-LOCKING: A technique of synchronizing several television (video) sources together. The sync signal from a camera and a SEG can be used to control the VTR so both the camera and VTR signals can be combined in the SEG.

CHARACTER GENERATOR: A device which electronically creates letters and numbers on the TV screen.

NEGATIVE IMAGING: Reversing the black and white sections of the picture. Just as in the negative of a photograph.

SOFTWARE/HARDWARE: Hardware is the actual equipment to be used in a particular application: i.e., camera, tape recorder, speaker, etc. Software is the electronically programmed decision-making equipment which controls the hardware functioning. Software may be altered fairly easily.

PORTA-PAK: A small camera and video tape recorder which can be used on location. It usually runs on batteries.

INSERT EDITING: The insertion of a video segment into an already recorded series of segments on a video tape. The inserted segment replaces one which must be of the exact length. Insert edit demands that the segment be edited in and then edited out at the proper moment since already recorded information exists immediately following the edited-in segment on the original tape. The use of SMPTE code to define the exact edit-in and edit-out points makes this operation quite simple. Rehearse record programming on a synchronizer lets the operator practice through E-E MODE.

ASSEMBLY EDITING: A method of electronic video tape editing. Various video taped segments are re-taped in a predetermined sequence to produce a coherent whole.

SWITCHER: A type of SEG which allows the operator to switch between several cameras or other video signal sources. Some switchers have internal sync generators to keep all video program material in proper time relation to each other. This insures quiet and glitch-free picture transitions.

SEG: Special Effects Generator. A device which permits the manipulation of several video sources; gen-locking, split-screening, fade-ins and fade-outs, are some of the features available on the better switchers.

— continued overleaf . . .

comfortable in the video medium, although the multitrack audio/video interlock was new to her.

The session was scheduled to start at 10:00 a.m., but the first set of musicians would not arrive until some 1½-hours later. Both the studio and Fitzgerald wanted to lay down the SMPTE code and **V-drive** on the video tape and 2" audio tape, prior to the session. Fitzgerald and Sally arrived shortly before ten, with an armful of video tapes. Fitzgerald's skepticism of video was only further enchanced when they played the first reel of tape from the movie. The video picture on the monitor and the beam was **rolling** and the **retrace** lines could be seen. Maintenance was called and in a few minutes, they cleaned the **head drum** of the VCR, and re-tweaked the **skew** and adjusted the vertical on the Trini. Fitzgerald, though, still somewhat apprehensive, was impressed with the **resolution** and hue of the picture. The perennial **ghosting** he got on his home TV was totally absent here.

As R.O. was playing the VCR for Fitzgerald, he was also preparing to lay code and V-drive on tracks 23 and 24 of the multitrack and SMPTE on the spare audio track of the VCR. He would also, on each pass of new video, **dub** over the **scratch track** on the VCR to the multitrack. This would be used as a reference audio signal while the music and dialogue was added onto the Master. The first reel of video (film) was almost ½-hour long, but would easily fit onto the audio reel. R.O. programmed the generator preset for 1:00:00:00, checked **E-E mode** and started both machines. Several seconds later (making sure both machines were up to speed) he began recording time code, V-drive, and the scratch track. Fitzgerald watched in fascination as his movie played on the 19" screen in front of him. It was slightly annoying, not being used to seeing his film on a video screen, but the picture quality was excellent.

After the reel was dubbed over, R.O. wanted to check the quality of code and lock. By initiating both master and slave control to the synchronizer, he pressed locate and both machines began their search back to the **keyboard memory** position of 1:00:30:00 minus the preprogrammed **pre-roll** of 15 seconds. The machines rewound

V-DRIVE: The vertical sync pulses which control the vertical field by scanning of the TV screen by the electron beam gun. The V-Drive is 59.94 Hz (for color) rather than the standard line frequency rate of 60 Hz for black and white. On VCRs or VTRs which have an external sync input, the scan rate can be locked to other equipment, such as cameras, SEGs, etc. The vertical sync pulse should be laid down when interlocking an audio and video machine, as a backup to the SMPTE code. If the SMPTE code gets lost, or erased, or becomes unusable, the V-Drive signal can be used to get both machines running at the same speed, so that new code may be laid down. House sync generators can provide V-Drive of either 59.94 Hz (29.97 frames per second) for color, or 60 Hz (30 fps) for black and white. (See: JAM-SYNC/HOUSE-SYNC)

ROLLING: When the video picture appears to be moving either up or down across the screen, a loss, or loss of lock of vertical sync is the cause. The vertical control on your TV can control this.

RETRACE LINES: These are the lines in a TV picture which you normally cannot see because they are "blanked out." When the TV picture is rolling, you can see black lines which are the vertical retrace lines. Horizontal retrace lines (there will be many more in each field of video) can be seen when a TV's horizontal controls are not set properly.

HEAD DRUM ASSEMBLY: The recording and playback section of the VTR. In either Quadruplex, or Helical Scan Recording, the head drum spins so that the actual heads will pass over the tape at the proper time. The head drum servo speed is determined by the recovery of a 30 Hz cue track signal on the video tape and comparing it to the VTR's internal reference. Adjustment of the phase relationship between the two signals is called tracking. (See: RESOLVING)

SKEW: The tape tension of the video tape across the heads. Improper tension can be seen by FLAGGING, or bending of the TV picture at the top of the screen.

RESOLUTION: The picture detail on the TV screen.

GHOSTING: Multiple images of the video picture adjacent to the primary image. Can be caused by electromagnetic reflections in a TV receiver, or by mis-terminating a composite video line.

DUB: Copy.

SCRATCH TRACK: A reference track to be used while building the audio elements on the master multitrack. Also, helps determine the degree of sync achieved between the two machines by listening to them simultaneously.

E-E MODE: On an editing VTR/VCR one may want to preview prior to recording, or monitor during recording, the video, audio, or sync signals. E-E MODE is the equivalent of "input" on an audio tape machine. E-E (electronic-to-electronic) MODE got its name by bypassing the record amp and tape; one just looks at the signal as it passes through the input amps and playback electronics. This mode will not tell you exactly what is on tape.

SYNCHRONIZER: The piece of electronic gear which compares two SMPTE code sources, and provides an output to make one machine match the speed and location of another.

KEYBOARD MEMORY: A built-in memory provided in most synchronizers. It can refer to a preset location on the tape, or an arbitrary "0" which is always referred to.

PRE-ROLL: This is the amount of time (in SMPTE terms) prior to the actual desired start of audio/video program. The pre-roll position allows both machines to get up to speed and achieve proper lock prior to the beginning of program material.

— continued overleaf . . .

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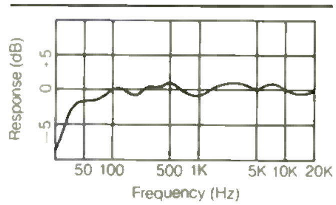

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silently; the tape was not travelling against the heads as this synchronizer used **tachometer** pulses for the **fast search**. Only as the machines decelerated to near the **search point**, did the head lifter retract to sample the code momentarily. A few seconds later, the machines were stopped in their **park position**: Barely 2 frames off from each other. To test lock, R.O. pressed **play** on the synchronizer, which now controlled play on both machines. The raster on the Trini began to wiggle, the countdown began and picture and sound appeared.

"We have frame and bit lock," R.O. said, "in less than 6 seconds. That's great. Let's hear how close we really are." R.O. began to A/B the audio from the VCR and the dubbed audio on the multitrack.

"Sounds O.K. Let's hear 'em together." As he played both machines' audio signals at the same time through speakers, there was no **phasing** at all. Fitzgerald was duly impressed and seemed to relax as they transferred the next reels of video while laying down the SMPTE code, sync signals, and scratch track.

The band arrived and as usual, the studio was impeccably ready for them. The schedule was to record all of the orchestral music in one session and then at a later date add the single and dual instrument cues.

The band was pro. A single run through on each number and they were ready to roll. The conductor

TACHOMETER: A pickup device on a motor which provides output pulses according to the number of rotations of the motor. These "tach" pulses can drive counters and be used as fairly accurate tape position devices.

FAST SEARCH: Fast searching, or high speed locating of the desired position, can be done by a high speed sampling of the SMPTE code, but the tape must be against the heads to properly read the code. Two problems result. One, the frequency of the code is multiplied as a function of the speed of the machine, so a wide band preamplifier is needed for the track with the SMPTE code. Second, not all audio multitrack tape machines have a muting function when in high speed winds, and the squeal can be atrocious. The concept of a Fast Search using a tachometer and its pulses to approximate the position of the tape eliminates both of these problems. The SMPTE code then, only has to be sampled once in a while, when the tape speed is below twice the normal play speed. This technique provides a highly accurate means of location. This can save the wear and tear on heads and ears.

SEARCH POINT: The location (in SMPTE) that the machine is searching for on the tape. This parallels the search to zero, etc., concept that autolocators, etc., have had for years.

PARK POSITION: The actual SMPTE location where the tape stopped, on both the VTR and multitrack machine. The object is to have both machines "park" as close to each other (SMPTE-wise) so that they can proceed to lock up as quickly as possible. Synchronizers have a park "window" or range of 30 minutes for usable re-lock. Most synchronizers can park within 2 - 3 frames.

PHASING: This means phasing just as we know it. By listening to two machines with the same audio, we can audibly determine if lock is acceptable by listening for any phasing (caused by wow, flutter, etc.) between them.

had his own monitor for viewing the scenes, while the band watched the larger projection screen 20 feet in front of them. The lights had barely to be dimmed because of the **luminance** provided by the screen and projector tubes.

The session went exceptionally fast and both Fitzgerald and Sally almost felt like outsiders for lack of anything to do. R.O. was **storing** the major cue points in the synchronizer's memories as they came up and was able to recall them for a replay whenever Fitzgerald wanted one. The **cycling** feature of the synchronizer fascinated Fitzgerald more than his own film at this point. He didn't completely understand why the tape machines would stop then rewind, then play again always at the right time, without anyone touching any of the controls on the keyboard. Before they knew it, the session was over, the musicians gone and Fitzgerald wanted some playbacks.

The cues were remarkable, he thought. This video and audio lock-up is really O.K. Each one worked as well as the next. Except for one. At the beginning of one scene, a

LUMINANCE: Brightness.

STORING: Placing SMPTE points into the synchronizer's memory banks. These may be retrieved for rapid location of any point in the audio or video program. Generally, an engineer will store scene beginnings, musical intros, outros, etc., for rapid cueing.

CYCLING: The program within a synchronizer to perform over and over the same function; for example: PLAY from 1:06:25:00 (1 hour, 6 minutes, 25 seconds) to 1:14:18:21 (1 hour, 14 minutes, 18 seconds, 21 frames). STOP: REWIND to 1:06:25:00 plus 30 second PRE-ROLL (1:05:55:00). STOP: enter PLAY go to . . . etc., until the operator manually overrides the CYCLING. Any CYCLING routing of this nature can be programmed into the better synchronizers; even RECORD and REHEARSE RECORD.

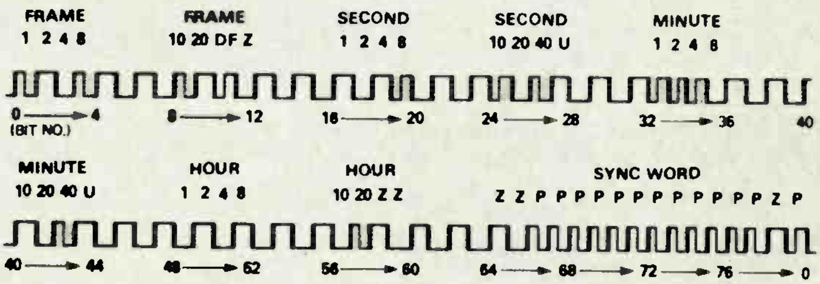


ILLUSTRATION SHOWS: 20 HOURS. 47 MINUTES. 58 SECONDS. 19 FRAMES

DF = DROP FRAME BIT. (USED FOR COLOR VIDEO ONLY)

P = PERMANENT ONE U = UNASSIGNED BITS. Z = PERMANENT ZERO

SMPTE Time and Control Code

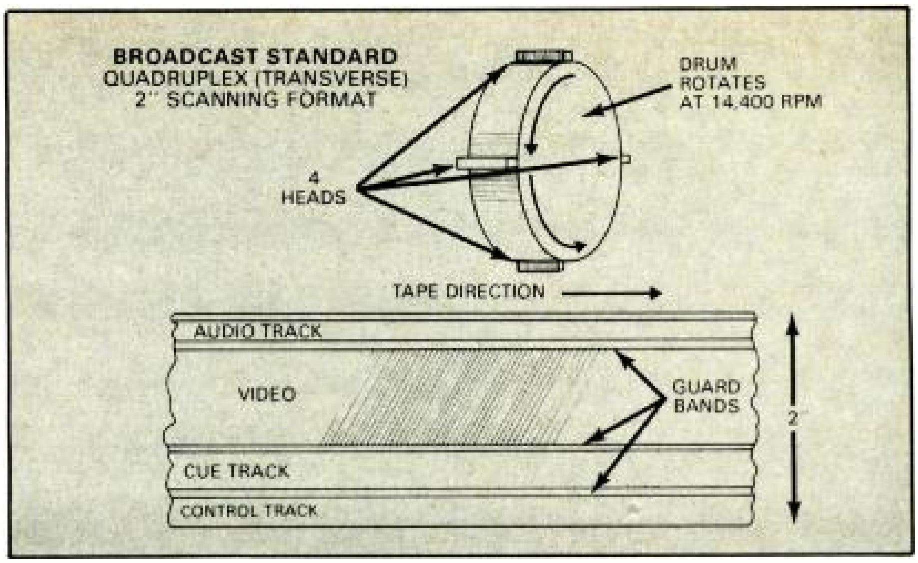
— continued overleaf . . .

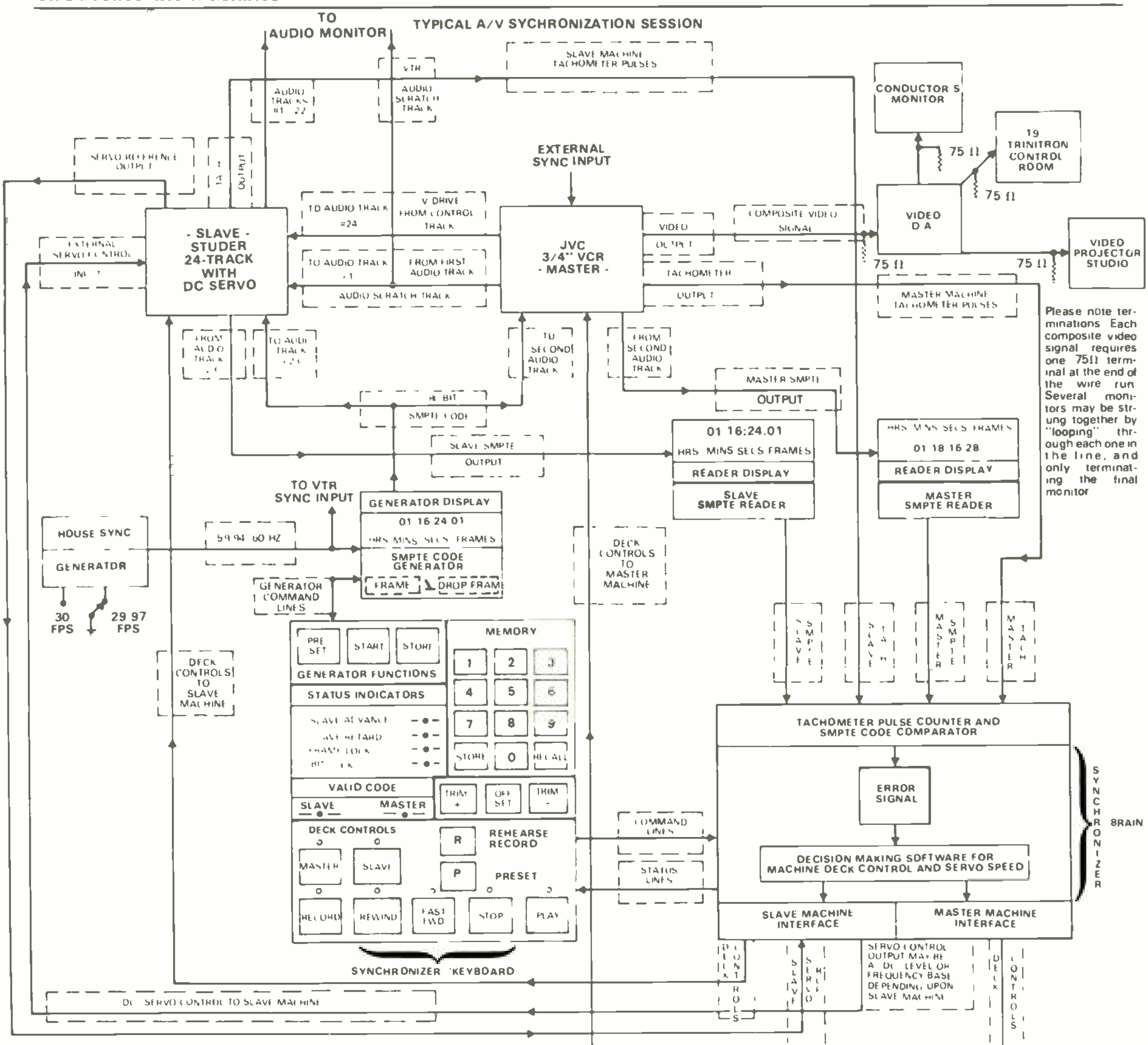
door is slammed shut and someone starts running down a long hallway, through a myriad of corridors, out into a parking lot and finally slams a car door. There were two important cue points which had to correspond to the music; the first door had to be exactly on cue and the car door slam had to be on cue also. The first door slam was perfect, but the car door was slightly out of sync with the musical downbeat.

R.O. noticed Fitzgerald's silent panic; rapid note taking and getting ready to edit the film again. "Do you want both doors to be at the fortissimo?" R.O. asked.

"Can you do that? Easily?"

R.O. rolled the machines to that





NOTES
In the better synchronizers, the SMPTE code generators, readers, software, and interfaces are in a single unit. All displays, status indicators, control functions, memory select, etc., should be in the "keyboard."

The synchronization equipment here does not represent any particular make or model. We merely want to show the basics required for sync. Either audio or video machines may be utilized, and are not limited to two units.

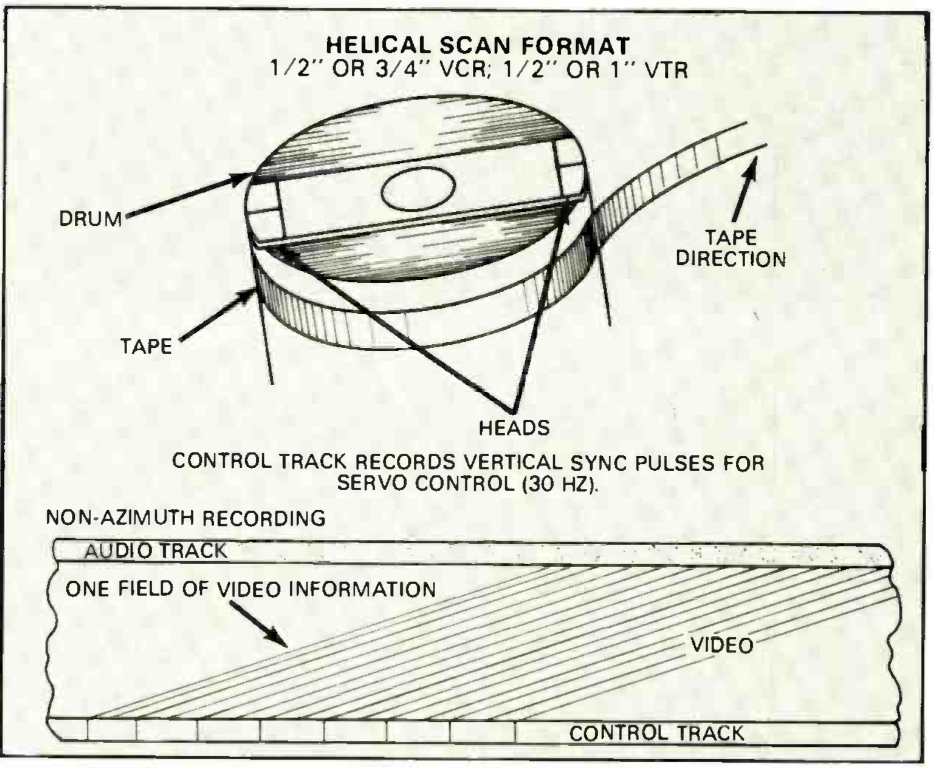
point prior to the car door slam. At the moment of the door slam, he transferred the reader display to the keyboard display memory. He then transferred from the reader display the moment of the musical cue needed. There was a difference of only 17 frames; but 17 frames is slightly over ½-second — a very noticeable difference for visual cueing purposes.

R.O. stored the 17 frame difference in the **offset** memory and pressed **locate** to start replay of the scene. The opening of the scene was perfect; just as it had been. While the corridors were being navigated by the movie's runner, R.O. pressed **trim offset**. By the time the car door slam arrived, the musical cue and visual cue were right in sync with each other. The 17 frame offset was introduced into the slave machine very slowly, by the synchronizer's internally programmed **slew** control, such that there was no perceptible change of speed while the slave moved from its true **in lock** position, to the offset position. Fitzgerald, obviously impressed, calmly forfeited his re-editing notes and they called it a day. □□□

OFFSET: The amount of time and frames that one wishes to either lead or lag one machine ahead of another.

TRIM OFFSET: The command to initiate the amount of offset (plus or minus) into the slave machine.

SLEW: Here, as before, slew is how fast the change occurs. □□□



GROUNDING, REFERENCING AND PATCHING FOR SINGLE-ENDED SYSTEMS

by

Fred Addison

Many high quality audio systems have single-ended as well as balanced or floating (two-ended) circuits appearing at the patchbay. Various grounding schemes have been used to avoid hum problems caused by ground loops created when patching between two physically remote single-ended amplifiers. Most schemes are limited to local crosspatching or require auxiliary transformers when patching in external equipment. Transformers not only isolate hum, but also avoid excessive crosstalk and instability due to ground modulations and poor amplifier referencing. The problem, then, is to develop a consistent and predictable scheme that would allow interfacing all types of equipment at the jackbay and the connector panel without the use of special patchcords or auxiliary transformers.

Grounding Geometry

Hum potentials exist across a large loop of wire in a room because of 60 and 120 Hz radiation from power transformers, flourescent ballasts, or even AC currents in the wall wiring. When the loop is closed, hum potentials are created at high resistance connection points in the loop, that is, within the elctronics. When the loop is opened, the total hum potential (a function of field strength and cross-sectional area of the loop) is seen across the break. Thus, hum potentials can exist even without a closed loop if there is hum induced in the ground reference between source and receiving amplifiers. Hum can also be induced into a relatively straight wire run close to an AC power transformer.

The most general solution to induced hum problems* is geometric — that is, to establish **one central** ground reference in the console and to distribute the common via one heavy busbar to the electronics, through the patchbay and then to connector panels and racks. It is important to keep the wires in bundles physically close together and then run straight as possible so that the interface circuits to the patchbay, etc., present no cross-sectional area to the hum field. Minimizing the cross-sectional area between the wires that the hum flux can pass through minimizes ground loop potentials. If there are two physically separated racks with separate cable runs to the console, only one should be used for single-ended equipment. This will minimize the difference between ground references when patching between single-ended units.

** Hum can also be caused by leaky power supply transformers, and this can usually be corrected by swapping AC phase.*

Referencing Long Signal Chains

Hum potentials and cross talk appearing on grounds can be amplified, or poor muting may be caused, if the amplifiers are not referenced to the proper common. Each stage of a system is often treated as a separate black box which has its own power return common. The black box circuit is then integrated into the system by tying it to the power bus rails and providing signal at its high input where ν_s is the signal and ν_x is the hum and crosstalk potential between source and amplifier. With this scheme, the difference between the commons of the source and receiving amplifers, ν_x is amplified by the gain of the circuit.

If the input of the amplifier is referred to the source amp, then the ground potential is not amplified as shown in Figure 2. Here ν_x which represents ground modulation from other current returns appears in both the output high and low, and is not seen by subsequent amplifiers referenced to this output.

This type of circuit works well for long signal chains or sections of a system where good muting or fader kill is required as shown in Figure 3. It should also be pointed out that voltages induced in the wires between source, control and feed are common mode and are not amplified.

Switched Or Patched Circuits

For single-ended circuits that are to be switched or patched, some means of

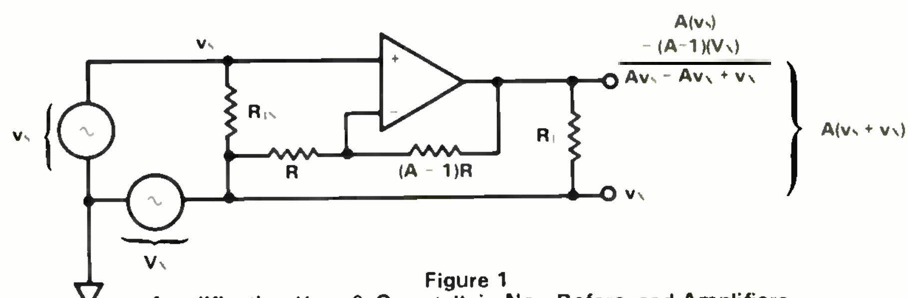


Figure 1
Amplification Hum & Crosstalk in Non-Referenced Amplifiers

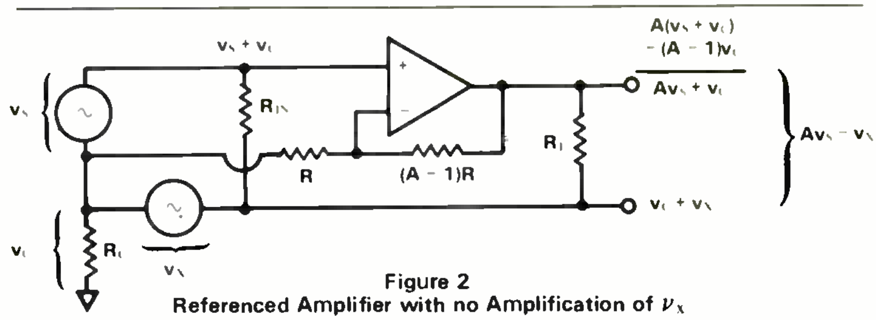


Figure 2
Referenced Amplifier with no Amplification of ν_x

— The Author —

***Fred Addison** earned a Bachelor of Science from Yale University in Applied Science and Electrical Engineering. His major field of study was Control Theory and Circuit Analysis. For over ten years he has been active in console design, construction, installation and modification. His experience includes various technical positions with API, Cetec Audio, Quad-Eight, Filmways/Heider, Marantz, Accurate Sound and Rupert Neve, Inc.*

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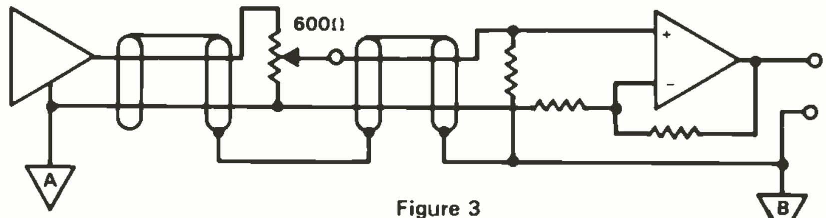

Figure 3
Amplifier Referenced to a Remote Gain Control

translating the signal from one reference to another is required. The techniques necessary to achieve quiet source selecting and patching depends of the intimacy of the common references involved. Selecting between two amplifiers referenced to the same common point requires only high-end switching. Selecting between two amplifiers sharing the same common bus at different points may require switching the amplifier reference as well. Selecting between separate systems may require the common mode rejection afforded by a transformer or different amplifier.

Similarly, if only crosspatching is desired in a compact single-ended system, then a wiring and cabling scheme that precludes ground loops may be all that is necessary. Patching between remote sections in the same system is possible by wiring amplifier input references to the patchbay to achieve higher isolation. Much higher isolation is required for patching between amplifiers with separate power supplies referenced to separate "real earth" ground points.

Many professional electronics devices are provided with single-ended inputs and floating outputs. This has resulted from the use of step-up transformers at amplifier outputs to achieve high output swing with low rail voltages. In an effort to minimize the number of transformers, yet retain patching capability, some systems have all outputs floating and all inputs single-ended. These single-ended inputs cannot be patched with impunity to external single-ended equipment. Recently, differential amplifiers have been employed where ultra-high fidelity and high common mode rejection are required, such as multi-track monitor inputs and monitor select inputs. These are wired to patchbays and connectors as traditional balanced inputs.

A Consistent Interface Scheme

The patching/grounding scheme presented in Figures 4 and 5 represents a consistent wiring system that takes into account the various natures of the patched sources and inputs. The basic rules adhered to in this system are:

1 - Significant output load currents are returned to their source common via the "ring" patch connection and the output "low" wires. Drive currents return by the same path so that no hum potentials can be induced across the fed input. Return currents to internal transformers from single-ended inputs are carried on the input shield for convenience (otherwise mono jack cords would be required for patching to single-ended inputs).

2 - Input shields are grounded via the common close to the amplifier input (good RF practice) with no common present on input "low" (ring) to prevent ground loops.

3 - Internal single-ended output shields are tied to the output low wire via patch normals so that the shield is switched to the shield/sleeve connection of a patched input. This arrangement proves useful in the forward referenced amplifier system described later.

4 - All connections are interchangeable, i.e., a single-ended or floating device may be connected or patched anywhere.

Shield Connections

Because RF shielding is often improved when shields are attached to ground at both ends, coupling capacitors are placed between commons and shields at all interface outputs (a .01 μf disk capacitor should provide a good RF shunt). External amplifiers are referenced to the systems common via output shields. Since this shield would be the only return path for drive current from the console, the output and input lines must follow the same path between the console and rack to prevent hum potentials at the external input. RF shielding would be improved if these

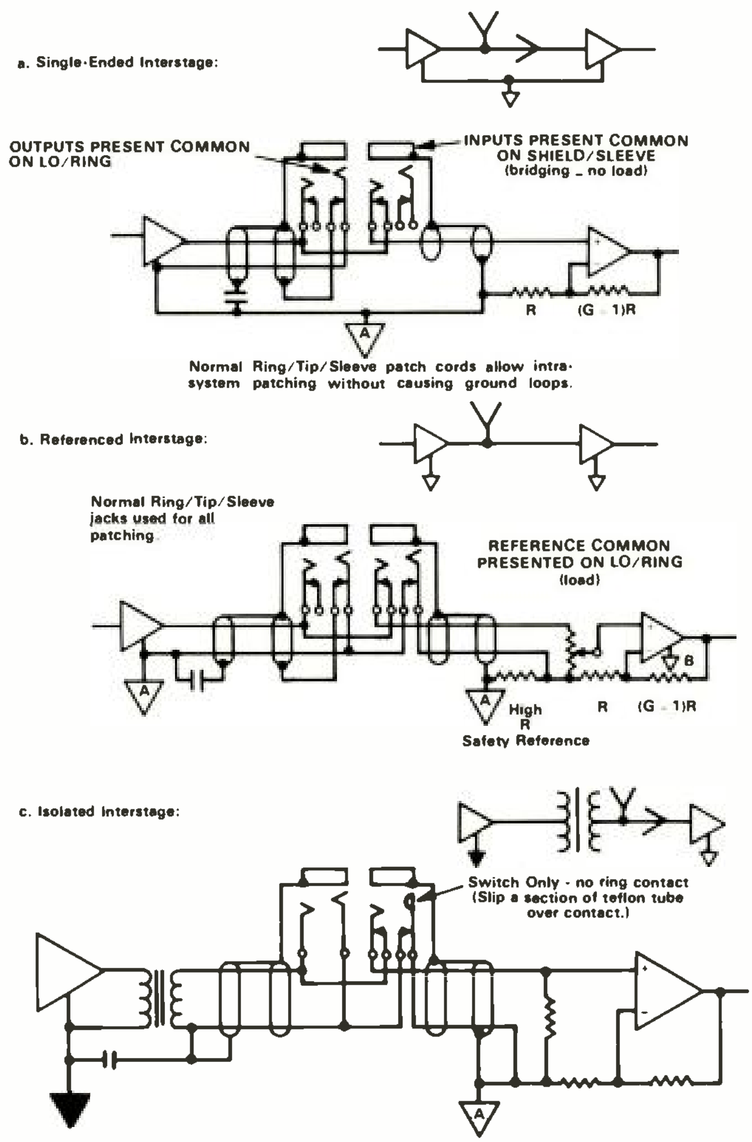

Figure 4 - Interstage Patching

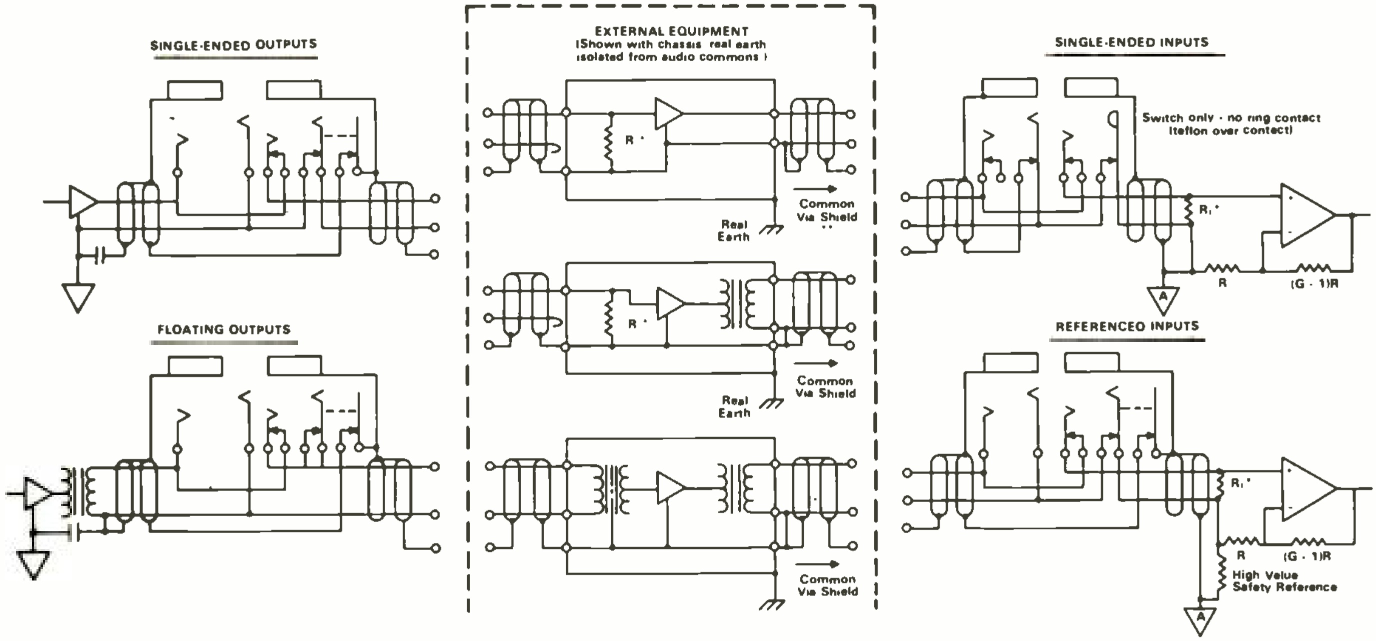

Figure 5
Patching Scheme for Interfacing External Equipment to Console with Single-Ended Inputs/Outputs

NOTES
* If $R_1 = R_i$ for unity gain external equipment, then return currents will cancel, reducing crosstalk.
** Input and output wires should follow same routing to console to prevent ground loops. Also, the use of an earthed conduit will reduce RF on single conductor shields.

lines were run within an earthed conduit.

Grounding External Single-Ended Equipment

The audio commons of the external equipment in Figure 5 are shown isolated from the local chassis ground. This is necessary to take advantage of isolation afforded by the power transformer in the remote unit. If we have a stereo processing unit outboard with its own power supply, then its audio common could float up and down with the console, while hum due to ground potential differences would be strictly a function of power supply leakage and routing geometrics. Even if all console inputs are differential, crosstalk among many remote single-ended units sharing the same floating power supply would still be a problem. The input and output load return currents from one channel would modulate the commons of other channels in relation to their source. This causes the ground potential differences between the source and feed to be amplified. Multi-channel remote units with common power supplies must therefore have referenced or floating inputs to avoid crosstalk if the console has single-ended outputs.

It is fine to discuss this kind of approach in theory, but it is often difficult to modify outboard gear to break the audio common from chassis AC ground or to install differential inputs on multi-channel units. One answer is to eliminate the ground loop between AC ground, outboard common, and console geometrically, by running the 3rd wire AC ground back to the console along **the same path** as the audio wires. (This must be an extremely low impedance path to satisfy most safety regulations. There are AC receptacles on the market now that allow the 3rd wire to be isolated from the conduit.) With no cross-sectional area between the audio and AC/chassis commons going between the console and external equipment, the central console common may be tied to the best real earth ground available. While this method eliminates hum (potentials are induced on the way to

the rack and un-induced on the return), the possibility of crosstalk due to contamination of the remote common by return currents, as described above, still exists.

Forward Referencing Of Amplifiers

Even with the elaborate scheme for patching and wiring described in Figures 4 and 5, completely transformerless interfaces between single-ended systems on separate power supplies are not possible without differential amplifiers at every input. The system thus defined would have all differential inputs and all high level single-ended outputs. To provide a system with the capability of interfacing to remote single-ended equipment, a transformerless isolated output is required. A differential amplifier may be used to reference an output to the following amplifier to achieve common mode isolation.

The differential amplifier in Figure 6 senses the common potential at the amp it is driving via the shield wire. This signal is connected to the unity gain reference leg of the differential amp and injected into the output. The drive to the following amp therefore contains v_x, the difference between the common references, which is not amplified. If the forward referenced differential amp is a true operational amplifier operating on bi-polar power rails, then it can be set up to have differential inputs and forward referencing. Figure 7 shows a forward-reference push-pull output amplifier, which is equivalent to a step-up transformer.

This offers the possibility of designing a system with all outputs consisting of forward referencing differential amplifiers and all inputs consisting of bridging single-ended amplifiers. Because most consoles have more inputs than outputs, this system would be more cost effective than providing all differential inputs, which would be required only when significant input load currents must be returned to their source via the "low" wire.

Bi-Level Patching System

By using the patch system shown in Figures 4 and 5 in conjunction with balanced outputs, a consistent bi-level patching system may be achieved. The output low wire/ring connection is designated as the inverted phase line, which may be driven by a balanced transformer or a mirror output amplifier (Figures 8 and 9). This system precludes the use of referenced single-ended inputs (Figure 4B) because the reference point (low of input loads), cannot be connected to the inverted phase line.

Bridging single-ended inputs receive η the signal across the tip and shield connections for direct interstage patching at nominal levels using RTS patch cords. Isolated inputs (differential amps or bridging transformers) receive full signal across the tip and ring connections. Balanced input transformers are functionally superior in this application in that they would produce the same internal level whether patched to a −2 single-ended source or a +4 balanced source. Differential inputs will achieve the necessary isolation but would always have the same net gain. If a gain control is ahead of the differential amp, the setting loss is simply 6 dB more for +4 dBm inputs. If the gain control is after the differential amplifier, however, sufficient headroom must be allowed for +4 dBm inputs with

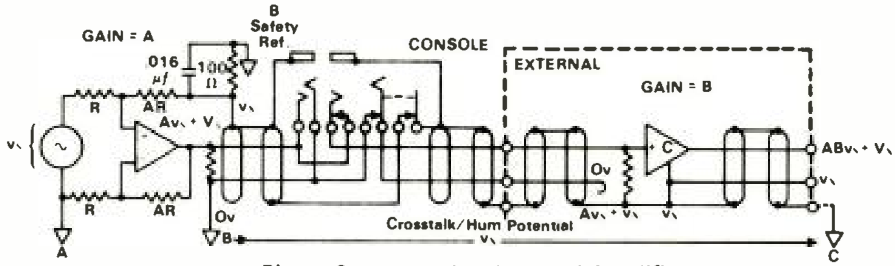

Figure 6 - Forward Referenced Amplifier

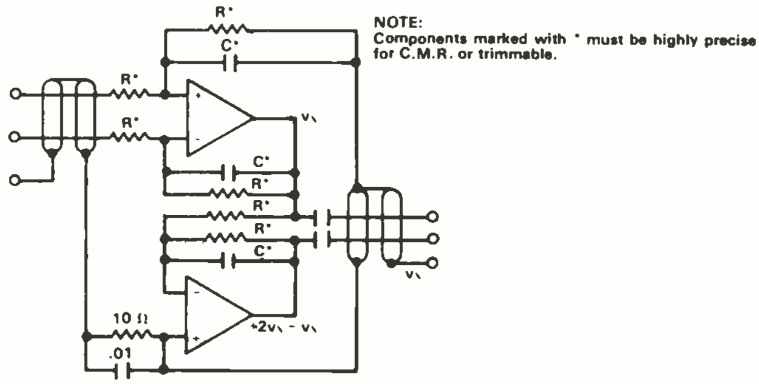

Figure 7 - An Electronic 6 dB Step-Up Output with Forward Referenced Isolation and Differential Input

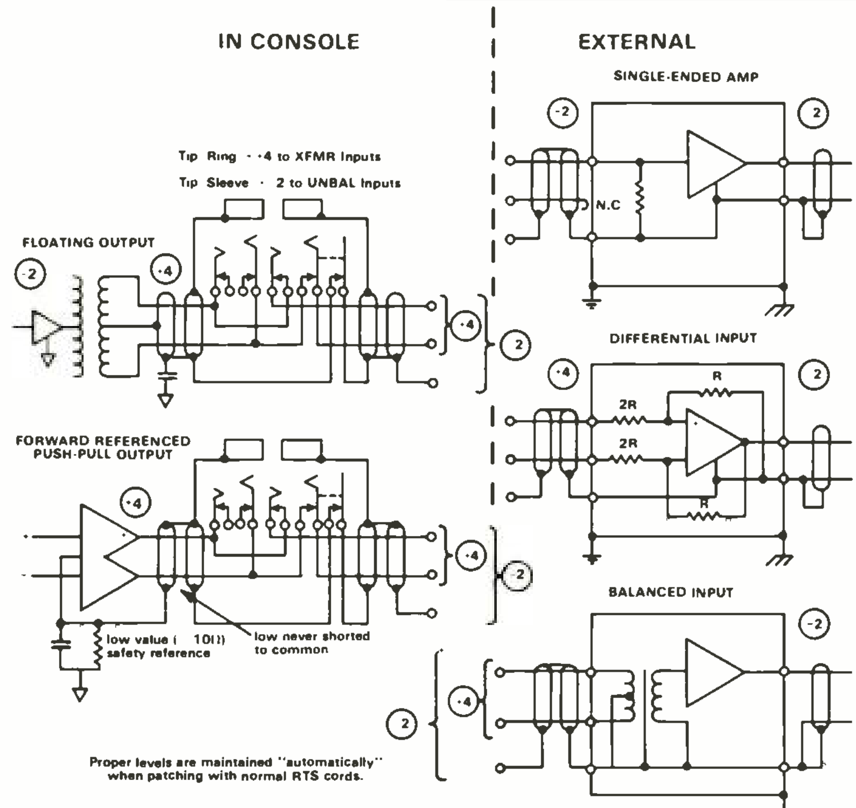

Figure 8 - Step-Up Outputs with Bi-Level Patching

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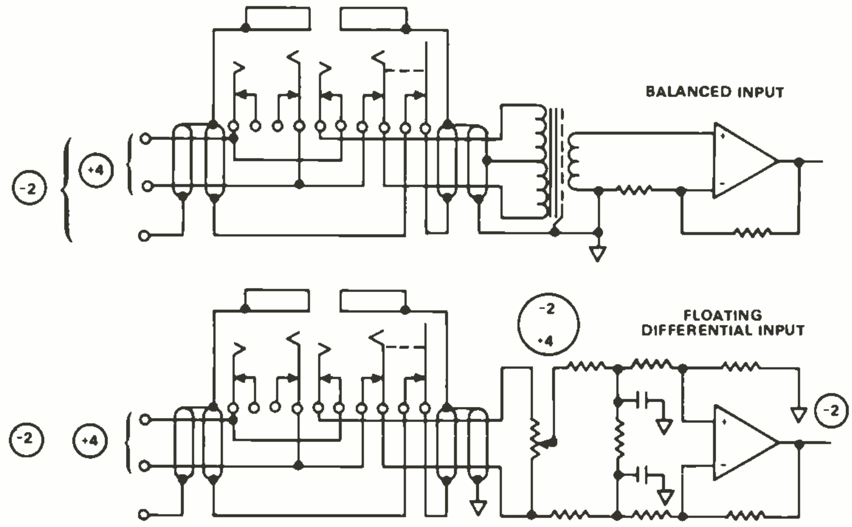


Figure 9 - Isolation Inputs with Bi-Level Patching

sufficient setting loss in the fader for −2 dBm inputs.

Mixing

Mixing systems consist of parallel chains of amplifiers feeding the same output. The input chains are often tied together by one common bus (▽). The crosstalk appearing between channels at the output of each chain is determined by the resistance of bus (▽) and the relative return currents from each input section. If the central common (▽) is located at the center of bus (▼) then the average return current common modulation will be the average module current times the module common resistance plus ϕ of the resistance of (▽).

One method of overcoming such crosstalk problems is to overpower it with massive common busses having such low resistance that significant voltage drops don't occur. If each amplifier is connected by a separate wire to such a bus, then the difference between its common reference and the rest of the system's amplifiers is due to its own return current only. This philosophy leads to systems where the patchbay shield connections are all bused together and connected to the massive common bus within the console.

In some cases the console chassis is actually used as the main common. This is poor practice for systems that may be used in high RF environments. If the console chassis provides any reduction of the RF field, then RF currents are induced on its skin. Referencing two amplifiers to different points on this common results in RF potentials at their outputs. Successful systems grounded in this manner employ all class A amplifiers that are bandwidth limited so that the RF is not demodulated or amplified.

Modulations due to hum fields will be a function of the cross-sectional area between outputs of the parallel chains to be mixed. Traditional systems overcome this problem by using transformers for isolation. Contemporary systems employing active (virtual ground) mixing have compact, shielded sum amp ground planes which carry as little current as possible (the sum amps drive high impedance loads) and are connected to the center of the input common bus (▽). If the system is physically large, then there can be significant hum differences between the commons of mixing chains referenced at the extreme ends of the main bus. The technique shown in Figure 10, can solve this and the crosstalk problem.

By making the last amplifier in each input chain differential, it may be referenced to later summing stages. The average common modulation of the

- COMMONS -

A **Audio Commons**

R **Reference Common** - a low current bus associated with the summing amplifier reference common.

P **Power Supply Return** - large decoupling capacitors between voltage supply lines and return line should be connected at the central ground reference point. Power supplies must be disconnected from 3rd wire real earth ground, if the secondary is not floating.

S **Power Supply Sense Ground** - specific stabilizing techniques may be required for remote sensing.

Chassis Common - racks; panels; frames.

L **Logic Common** - logic circuits may share the same supply as audio circuits with good power supply noise rejection, but should be provided with a separate common for return circuits to prevent clicks and pops.

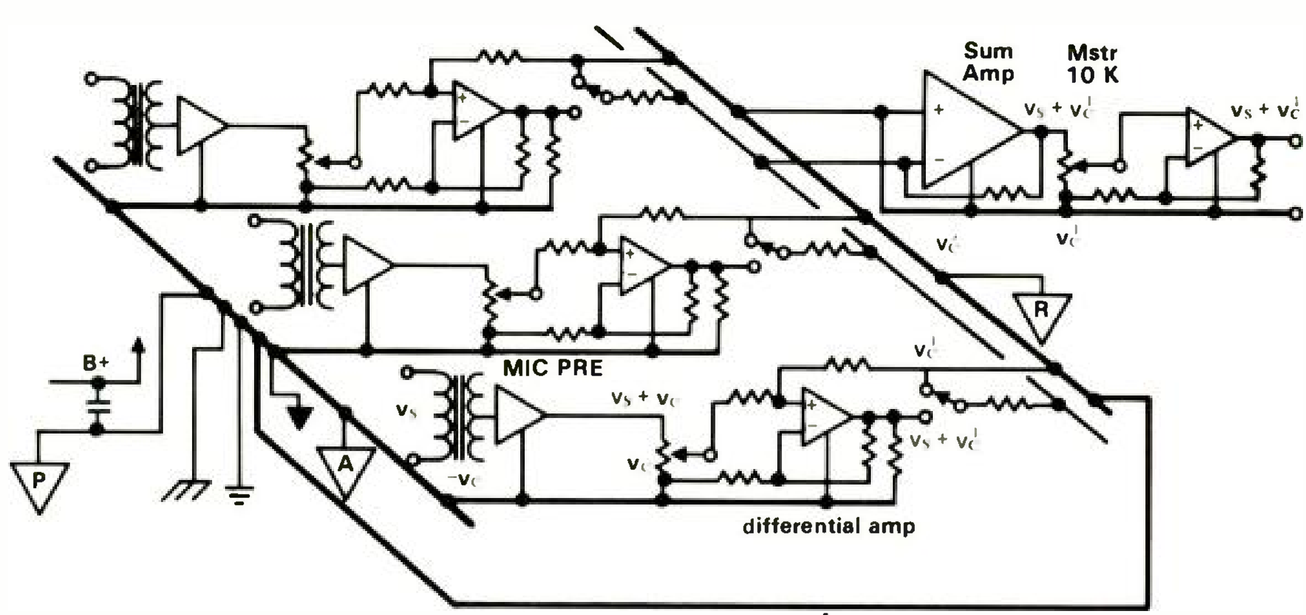


Figure 10 - Forward Referenced Summing

input section is v_c: that of the output section is v_c'. The output of the differential amplifier with no signal is v_c' with the common modulation of the input chain, v_c', being rejected. The sum amp sees v_c' at both inputs, therefore, there is no hum or crosstalk from the input section. It is important that the reference bus R parallel the routing of the actual signal so that any potentials induced on it and the sum busses are identical and would not be amplified.

Conclusion

It may be impractical to incorporate the patching scheme presented here into existing systems because of the difficulties in re-wiring patchbays and remoting cables. However, making the changes required in an existing system may be well worth the trouble, if transformers are being deleted. Since many systems use true op amps in output stages, it is not inconceivable that these could be modified to be forward referenced while all inputs are left single-ended. Using this scheme, transformers could remain in non-critical signal paths, while retaining the ability to patch freely. □□□

INTERFACE INDUCED AUDIO ANOMALIES

by Dave Baskind

Inconsistant performance of practically every piece of audio equipment has been or will be encountered by virtually everyone in our industry who has worked in multiple environments. The most common inconsistancies occur in steady state frequency response and transient response, but every performance perameter a device has may be affected by HOW and WHERE it is installed.

The purpose of this article is to inform the reader of how performance anomalies are generated and to offer simple procedures for preventing, identifying, and correcting them.

Table 1 lists primary parameters affected by interfacing equipment with typical subjective listening results.

— TABLE 1 —

PARAMETER		PERCEPTION
Frequency Response	—	Equalization
Transient Overshoot	—	Harshness
Supersonic Ringing	—	Dissonance
Subsonic Resonance	—	Washed Out Bottom End
Oscillation	—	Noise/Dirtiness
Total Harmonic Distortion:		
Even Order	—	Increased Brightness
Odd Order	—	Dissonance
Intermodulation Distortion	—	Dissonance
Overload	—	Brightness, Dissonance, Dirtiness
Low Frequnecy Distortion	—	Smeared Bottom End; Lack of "Punch"
Parasitic Oscillation	—	Dissonance, Dirtiness, Noise

Balance vs. Unbalanced Lines

Professional audio equipment has traditionally featured transformer coupled, balanced inputs and outputs. With the introduction of high performance, cost effective unbalanced devices, myths concerning the absolute need for transformers and balancing are fortunately being erased.

In practice we are finding that balanced line are needed only in high noise environments or when long runs of cable are required.

While truly balanced systems are generally unnecessary, it is still good practice to wire all audio systems with accepted traditional techniques. The general procedures are:

1 - Shields should be grounded at one end only to prevent current flow and possible ground loops.

2 - Inside each shield there should be two conductors.

3 - One conductor should connect the high or "signal" side.

4 - The second conductor should connect the low or "common" side.

5 - Grounding between equipment should be via tinned copper braid of as large a size as practical to insure good RF grounding.

Illustrations pertinent to the above are shown in Figures 1 and 2.

— *continued overleaf* . . .

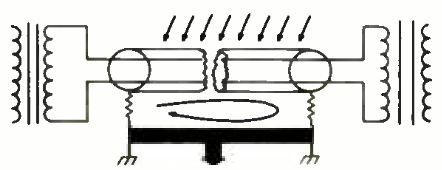


Figure 1: SHOWING GROUND LOOP INDUCTION INTO SHIELD

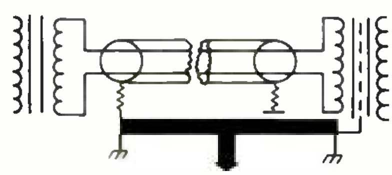
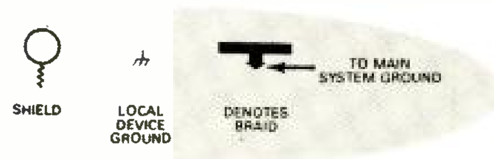


Figure 2A: BALANCED TO BALANCED COUPLING



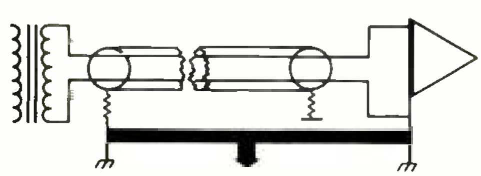


Figure 2B: BALANCED TO UNBALANCED COUPLING

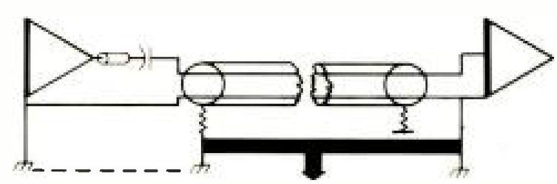
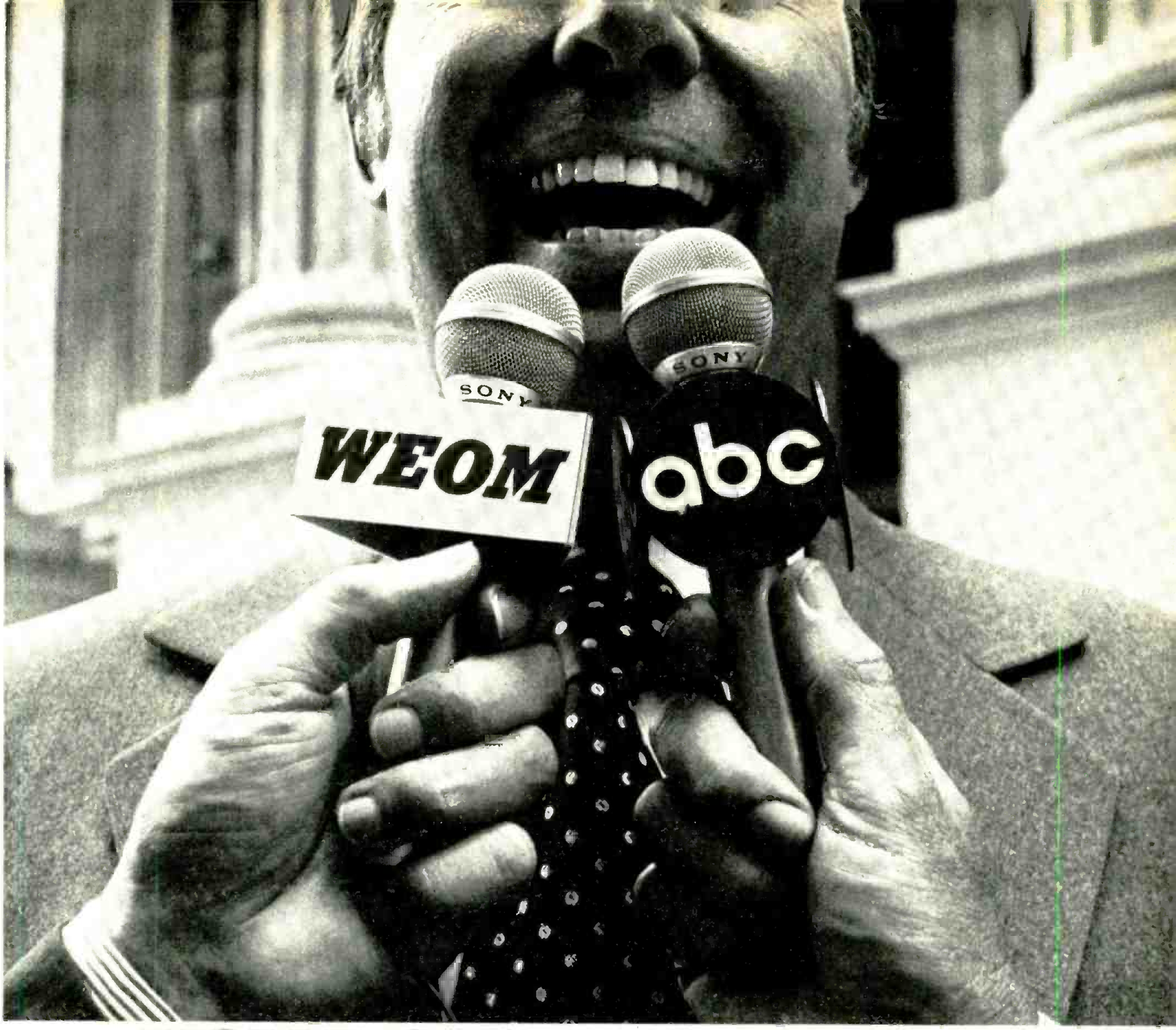


Figure 2C: UNBALANCED TO UNBALANCED COUPLING



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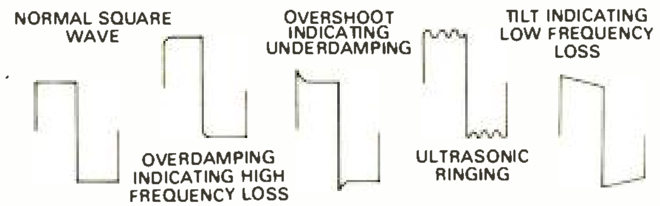


Figure 3: SQUARE WAVE RESPONSE PATTERNS

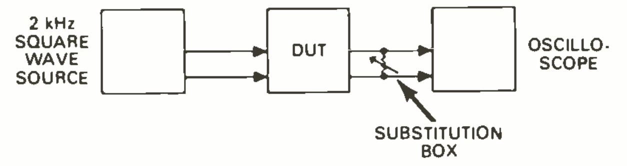
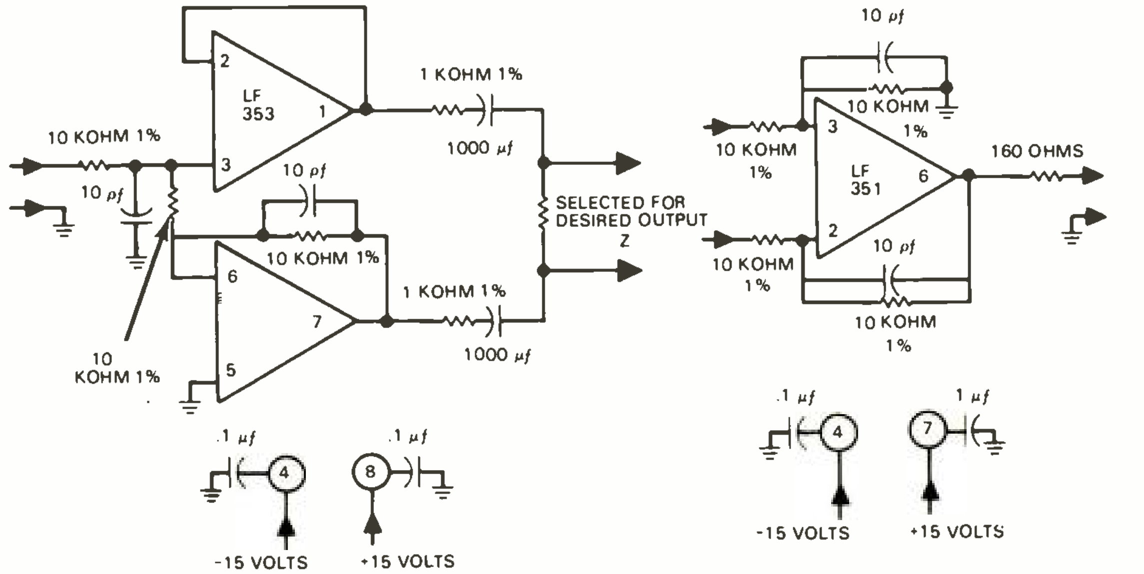


Figure 4: BASIC SQUARE WAVE TEST SETUP



(A) SINGLE ENDED TO DIFFERENTIAL CONVERTER

(B) DIFFERENTIAL TO SINGLE ENDED CONVERTER

Figure 5: CIRCUITS FOR CONNECTING UNBALANCED TEST EQUIPMENT TO BALANCED DUT'S

Cause Of High Frequency Anomalies

High frequency anomalies resulting from equipment interfacing are generally caused by two basic mechanisms:

1 - Transformer resonance.

2 - Effects of loading on preceding stage gain.

In the case of transformers, two factors must be considered:

1 - Effective load impedance.

2 - Cable capacitance.

The effective load impedance may be optimized by the use of a resistor substitution box, a 2 kHz square wave source, and an oscilloscope (see Figures 3, 4, and 5). Critical damping is usually desired and is reached when the output of the DUT (device under test) most closely matches the shape of the input square wave.

When the proper resistor value is found with the substitution box, then a permanent fixed resistor is wired in its place.

Cable capacitance need only be considered for extremely long cable runs. When long runs are encountered, a step-down transformer may be needed to minimize high frequency losses.

When the frequency response or transient performance of a device having no output transformer is affected by interfacing, it is usually the result of the load becoming part of the device's feedback circuits.

— continued overleaf . . .

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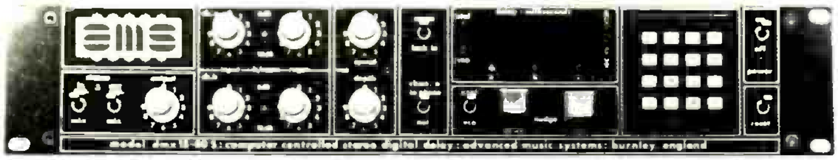

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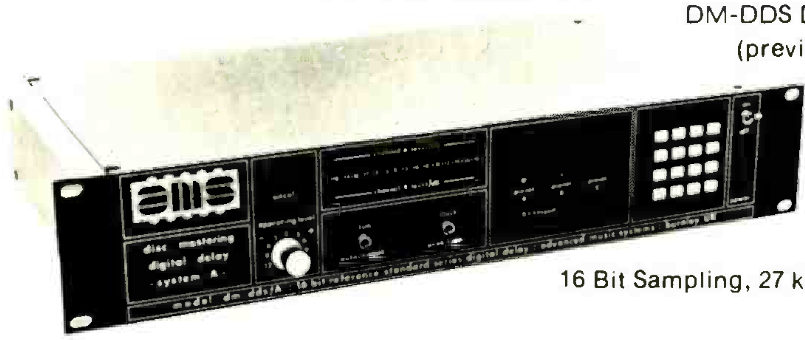

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INTERFACE INDUCED AUDIO ANOMALIES

A typical solution to this type of problem is to insert a resistor of 47 to 270 ohms or a ferrite bead between the device and its output line (see Figure 6). It is recommended that if a resistor is used that it be as small a value as practical and that square wave testing be used to confirm optimization.

The above paragraphs discussed corrections for high frequency problems resulting from terminating device outputs; input termination procedures are discussed below.

There are typically two types of input termination problems:

1 - Source impedance.
2 - Input level.

Source impedance is usually a problem unique to devices having input transformers. It is especially noticable in microphone preamplifiers with high turns ratio input transformers.

Figure 5A shows a simple circuit for generating a balanced signal for testing mike preamps for optimum source impedances.

All microphones should have their output impedances selected or built out with resistors to match console preamp requirements (see Figure 7).

Input level is the second major factor to be considered in optimizing input interfacing. If levels are too high, the stage can overload and if levels are too low then device noise floor can degrade the system dynamic range (see Figure 8).

For devices with transformer inputs, low frequency (50 Hz) sine wave testing will allow one to predict overload capability while for transformerless inputs any audio frequency sine wave may be used.

Once overload characteristics are determined, the user may select the desired headroom for the device and pad or boost the input circuit as needed (see Figure 9).

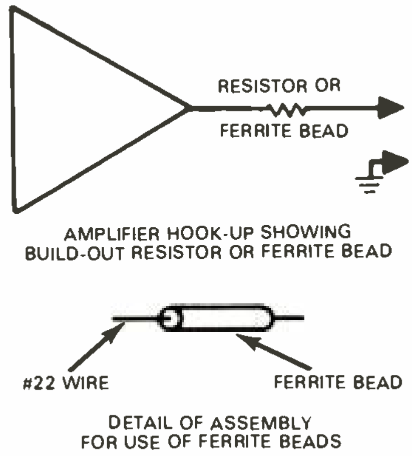


Figure 6: ADDING BUILT-OUT RESISTOR OR FERRITE BEADS TO ISOLATE AMPLIFIER OUTPUTS FROM I/O LINES

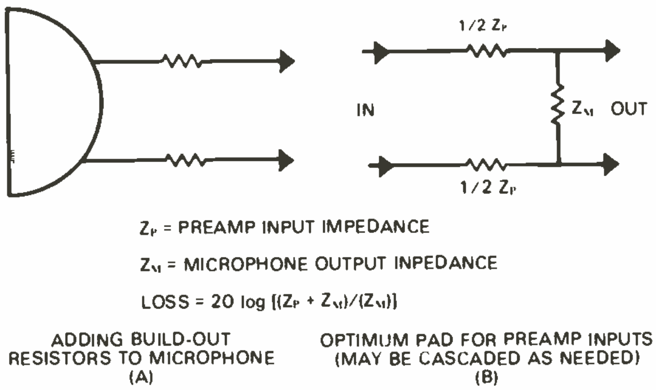


Figure 7: MICROPHONE INTERFACE NETWORKS

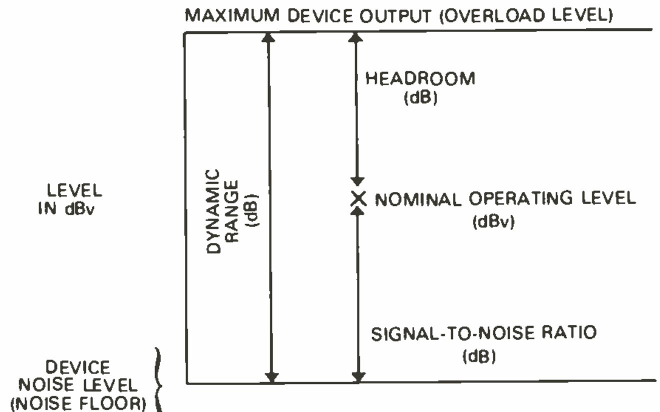
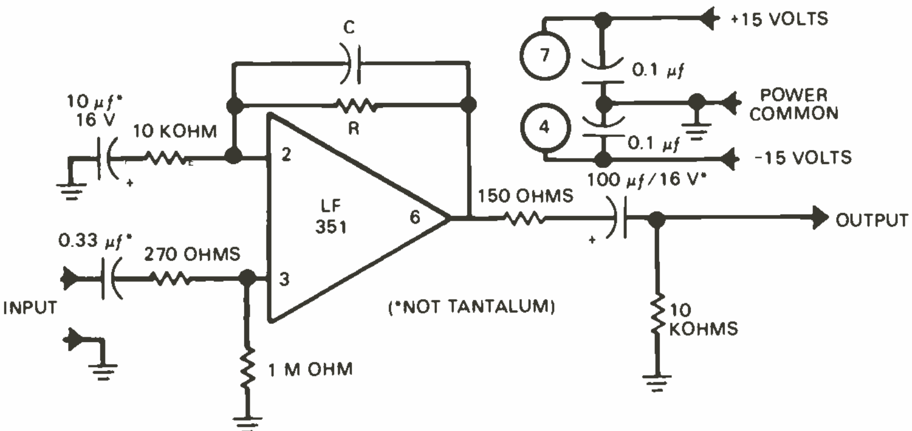
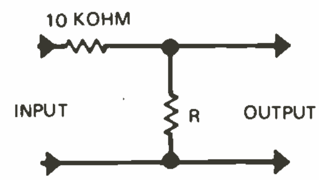


Figure 8

BOOSTER AMPLIFIER FOR "CONSUMER" DEVICES



GAIN DESIRED	R	C
6 dB	10 KΩ	47 pf
10 dB	20 KΩ	22 pf
16 dB	39 KΩ	12 pf
20 dB	100 KΩ	5 pf
26 dB	200 KΩ	-



LOSS DESIRED	R
6 dB	10 KΩ
10 dB	5 1 KΩ
12 dB	3.3 KΩ
16 dB	2 KΩ
20 dB	1.1 KΩ

INPUT PAD FOR "CONSUMER" DEVICES

Figure 9:

The Tascam 16-Track System.

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Subsonic Resonance

Subsonic resonance is caused by tank circuits resulting from the connection of a transformer winding to a coupling capacitor (see Figure 10).

This phenomenon is the least known and one of the most important problems encountered in professional audio today. It has caused multitudes of blown woofers and kissing grooves and rendered countless recordings useless.

The only way to totally eliminate this problem is to direct couple all active electronics (with no capacitors used at all) or to use no transformers.

Although current trends are toward the elimination of transformers, a majority of existing equipment includes them, and many engineers feel they are absolutely necessary (especially in broadcast, film and television).

If transformers must be used then one of two steps must be taken to minimize subsonic resonance:

1 - If there is no DC present, coupling capacitors should be eliminated.

2 - If DC is present then coupling capacitors should have values large enough to insure that their reactance is less than real impedances (resistive components) in the circuit at low frequencies.

The author has found subsonic resonance as an inherent design flaw in many commonly used devices, and recommends that all equipment be checked for this problem.

To determine if subsonic resonance exists in a DUT, an asymmetrical tone burst source and a DC coupled oscilloscope are needed for this procedure, devised by Deane Jensen (see Figures 11 and 12).

An alternate method for detecting subsonic resonance which is not as reliable is to check frequency response of the DUT in the .5 to 10 Hz range for a resonant peak.

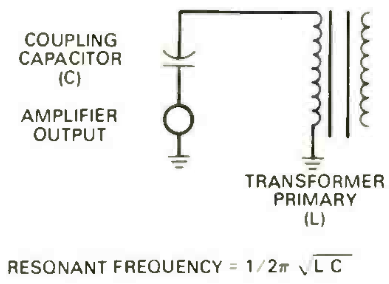

Figure 10

DC Induced Distortion

The presence of direct current on interface lines may be an indicator that unnecessary distortion is being generated. This is particularly true when transformers are a part of the circuit. If a transformer winding has DC on it and it is not designed for this condition, then its core is biased into a part of its hysteresis curve which will cause non-linear or distorted transfer of signals.

The DC voltage and current present may be very small and still cause a several

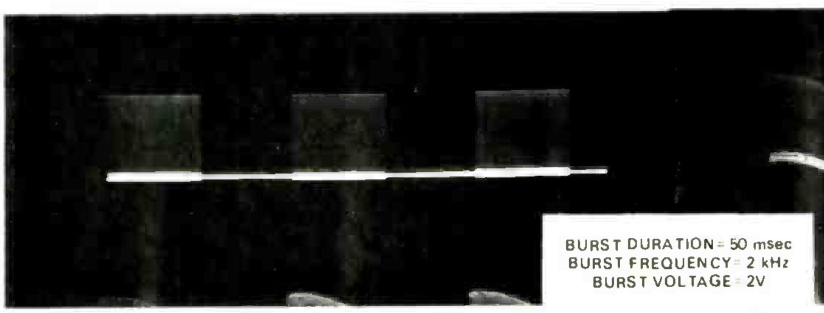

Figure 11: ASYMMETRICAL TONE BURST

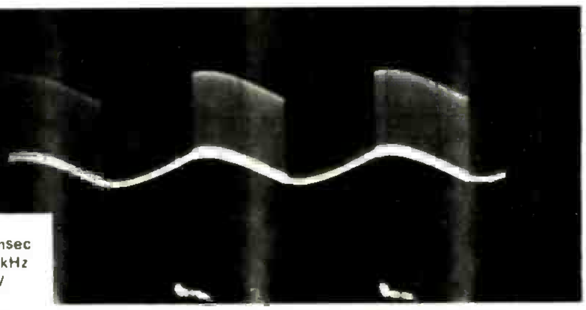

Figure 12: OSCILLOSCOPE TRACE INDICATING SUBSONIC RESONANCE

hundred per cent increase in distortion.

Typically, DC found in I/O lines is caused unintentionally by leaky coupling capacitors and intentionally by phantom supply circuits.

These problems may be identified by measurement across the windings for DC on capacitively coupled transformers and by measuring the system distortion with the phantom supply switched on and off.

Interfacing "Consumer" Or "Semi-Pro" Equipment

Although much of the lower cost audio equipment is excellent, the nominal intended signal levels are not intended for professional applications. To make these devices compatible with professional equipment, it is typically necesary to pad the inputs with 10 to 30 dB of loss and to add a 10 to 30 booster amplifier at the output (see Figure 9).

Interfacing Power Amplifiers and Loudspeakers

The same basic principles as discussed above apply to interfacing power amplifiers to loudspeakers. Special consideration should be given, however, to cable capacitance, cable conductance, and speaker impedance.

Because of the low impedances of loudspeakers, it is necessary to use minimum build out or insulation impedance at the amplifier output to maintain efficiency. This practice makes practically all power amplifiers extremely susceptible to cable capacitance. Also, because of the low load impedance involved, low cable resistance is essential for high efficiency.

These two considerations combined with the ability of a speaker to have positive, zero and NEGATIVE impedances at different frequencies points to the careful consideration that must be given in connecting amplifiers to speakers.

It is therefore recommended that:

1 - Cables be selected for minimum capacitance.

2 - Cables should be as short as possible.

3 - Adequate isolation inductance must be inserted at the amplifier output. (This is usually but not always done by the manufacturer.)

Conclusion

In conclusion the author recommends:

1 - That all studios and mixers invest in a good oscilloscope and a function generator.

2 - That everyone reading this not be afraid to use the best test equipment available around — ***ears.***

□□□

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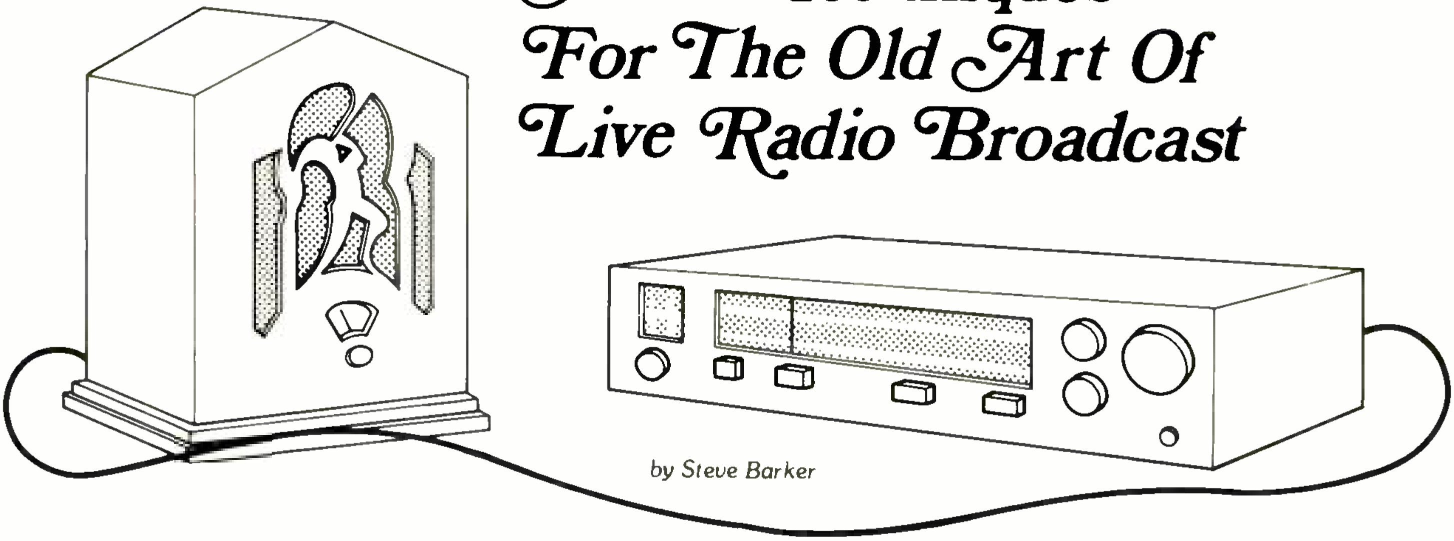
New Audio Techniques For The Old Art Of Live Radio Broadcast



by Steve Barker

The art of radio broadcast is hardly new. Starting in the late '20s, it reached the peak of its glory in the mid-'40s. However, with the influence of TV, live radio gradually regressed into obscurity and, excepting a few specials produced in recording studios, became a lost art.

Today, the art of the live radio broadcast is experiencing a re-birth. Due to the proliferation of wide-band satellite technology and the thrusts that public radio is making into the market place, the art of live broadcast is once again becoming a valid study.

The purpose of this article is to explore methods in which the live radio technology of the '40s can be combined with the modern recording studio technologies of the 1980s. In doing this, we must bear in mind that although the radio studios of the 1940s represented the state-of-the-art technically, the radio studio of today is often sadly limited in its live production capabilities compared to the advanced audio technologies that have been developed in the past 40 years.

The method of this exploration will be an analysis of the production techniques currently being developed by Barker Brothers Audio Technologies (B.B.A.T.), a company in which the author is the senior partner.

STAFF

The most essential ingredient of live radio is the staff, as without a trained staff, live, real-time radio will fall flat on its face.

Production Director

The person primarily responsible for the show from the time it enters the studio until the time the show ends is the production director. In the tradition of the old time directors, this person is, and of necessity must be, the absolute ruler of all activities in and around the studio during both rehearsals and air time. The production director's responsibilities include: arranging for the availability of the studio and other facilitites, "rounding up" the artistic and production personnel so that they are on time and ready for performances, supervising rote mechanical and engineering functions, and overseeing the timetable so that the show goes on and off air on schedule. In the conventional radio environment, the production direction and engineering functions would be separate; however, in most shows produced by B.B.A.T., both hats are worn by this author. (This immensely simplifies communication problems between the director and engineer!)

- the author -

***Steve Barker** began his involvement in radio production at KPFK, Los Angeles, during the mid-'60s at a time when the station was heavily committed to live music and drama, and continues to do consulting and engineering for them today. He was educated in telecommunications and theater arts at the University of Southern California where he did a great deal of sound engineering for their school of drama and all of the audio engineering for the concerts, speeches, and drama presentations of the student activities office. Mr. Barker currently is senior partner in Barker Brothers Audio Technologies an audio consulting firm, that specializes in symphonic sound reinforcement and recording as well as live theatrical audio design. He was awarded the 1978 Dramalogue Award for Excellence in the Theater for his sound design for Cages.*

Mr. Barker is currently under contract to the marketing divison of Home Savings and Loan, Los Angeles, is the Technical Director for the Ojai Music Festival, in Ojai, California, and the audio consultant to the Long Beach Symphony and the Long Beach Grand Opera.

His long-standing interest in radio and old college ties prompted his present involvement with KLON, Long Beach, California, where his experience and technical inventiveness have redefined the quality and art of live productions. According to his studio manager, "Steve never says it can't be done. He is always experimenting and developing new miking techniques and building new equipment. KLON is very fortunate to have him working with us."

Some of Steve Barker's miking and equipment designs are included in the article.

Studio Manager

Studio activities are controlled by the studio manager who is the liaison between those in the studio and the booth. This person makes sure musicians stay on assigned microphones and have whatever headphones or other peripheral gear they need available to them during the show. The studio manager is also responsible for fine tuning microphones for various artists who may share a common microphone circuit. In short, the studio manager is essentially the arms and legs of the producer and the engineer.

Engineer

The engineer is responsible for the basic microphone and the studio set-up, all mechanical operations in-booth, and for balancing (mixing) the program.

Host

The host's responsibilities during the show are to announce the numbers, interview guests, and provide background information and verbal continuity to the show.

Band Leader or Contractor

The band leader's responsibility is to make sure that charts are available for every member of the orchestra, to organize rehearsals, and to be the liason between the orchestra and the technical staff during the broadcast, informing the engineer of any technically important cues.

A word is in order regarding the communication system that has been devised for use during the broadcast. Due to the rigors of real time, live production, it stands to reason that efficient and immediate communication must be available among the staff and between the studio and the booth. The studio manager is in communication with the booth via a headphone communication system. The host/announcer is signalled

— continued overleaf . . .

when to talk by a pair of signal lights. In an emergency, the headphones worn by the host/announcer can be accessed to by the production director or engineer in the booth on a system separate from the musician cue system. The studio manager can talk to the booth via a "push-to-talk" system which accesses to a small paging speaker mounted in the console; and the booth can talk to the studio during rehearsals via a conventional talkback speaker system.

It should be noted that although this article refers to a control "booth," that luxury is seldom afforded the itinerant producer. In the case of B.B.A.T., the booth is a remote recording van specifically designed with live radio in mind. This van is connected to the studio via a 250-foot umbilical cord containing mike, communications, video, and power lines.

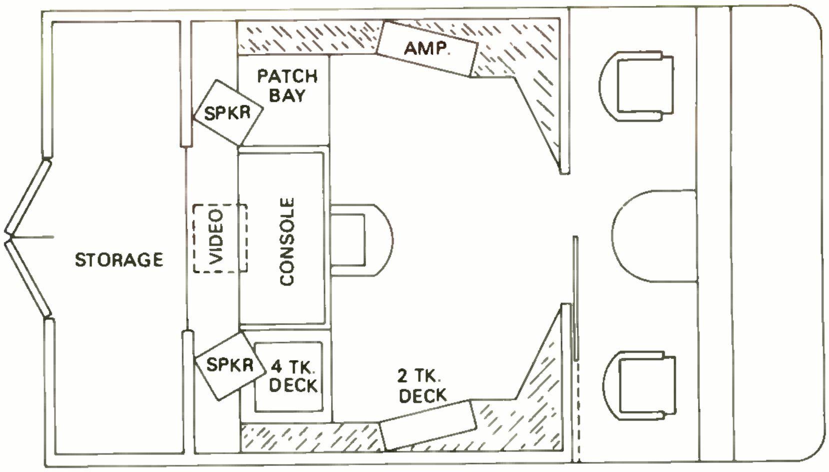


Figure 1: B.B.A.T. VAN

THE STUDIO

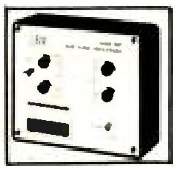
Most radio studios are not equipped for modern live music production work. In many cases, the available studio has two microphone inputs at most. For the purpose of this article, we will assume that a studio is available, but must be upgraded for live musical and dramatic production.

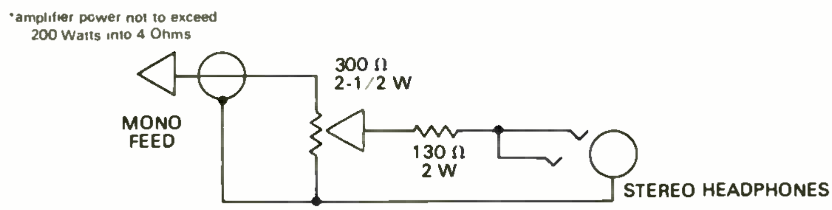
To begin with, the room should be reasonably well insulated from extraneous noises (traffic, office equipment, talking outside). Square rooms are generally to be avoided due to the standing waves and slap echoes normally attributed to that particular geometry. However, if the room available does prove to be an acoustic nightmare with the above characteristics and/or with predominant resonances at some frequencies often caused by room geometry and construction materials, there are several minor acoustic fixes that can be effected relatively inexpensively. The easiest and simplest is the use of a simple "gobo" baffle. These baffles are made by hinging two pieces of plywood together, painting one side, and then covering the other side with high density foam or rock wool and covering this with burlap (see Figure 1). Gobos range anywhere from 2' x 2' all the way to 8' x 8'. (A 6' x 6' is an optimal large size as it can be folded and brought through a standard doorway.) A couple of cabinet handles attached to the outside hard surface of larger baffles makes them easier to maneuver and carry in the folded position.

These simple baffles will allow the achievement of some degree of isolation from instrument to instrument and will allow you to give the appearance of a much more intimate space when you need a voice or an effect to sound enclosed.

In rooms where the primary acoustic problem is the tendency of the space to resonate at certain frequencies, the fix is a little more complicated. The simplist approach is the construction of a device called a membrane resonator, a large, narrow box covered with a compliant surface so that it will resonate at the offending frequency, thereby absorbing a good percentage of that energy which would otherwise be reflected back into the room contributing to the room's acoustic woes. As the specific size, dimensions, and materials could vary greatly depending on the needs of the device, we will not attempt to describe their construction here. However, there are several very fine books and articles available which detail the subject quite thoroughly.[1]

Another acoustic problem that may be encountered is noisy air conditioning. Normally, the easiest approach is to shut the unit down when the ambient noise requirements of the recording are softer than the air conditioner will permit. If the problem is simply rattles and squeaks from the grilles and deflectors in the room, this can often be remedied by tightening all nuts and bolts and putting duct tape over the pieces likley to rattle. The low frequency rumble produced by air movement in the room can often be effectively removed by the use of a subsonic filter in the output of the mixer. A unit which is inexpensive and highly recommended is the UREI Subsonic Processor #501 (below $100.00). This will greatly reduce air conditioner noise and the majority of thumps, bumps, and low frequency garbage (below 30 Hz) that contribute nothing to the program material and cause the limiters to work overtime. This device is an important supplement to the radio studio which has not been constructed, designed, and equipped as a sound studio.





The maximum number of units is determined by the lowest impedance the amp can tolerate. With 8-ohm headphones the worst case load the box represents is 92.6 ohms. Therefore, twelve units may be used with an 8-ohm amp or twenty-four units with a 4-ohm amp.

Figure 2: HEADPHONE CIRCUIT

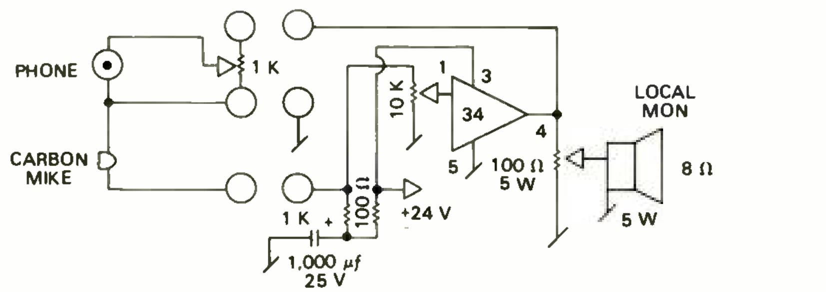


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Figure 3: SIMPLE COM SYSTEM A LA OP-AMP LABS

Limiters

As radio station limiters are generally designed and calibrated to prevent their transmitters from being overmodulated, they are oftentimes anything but musically unob-

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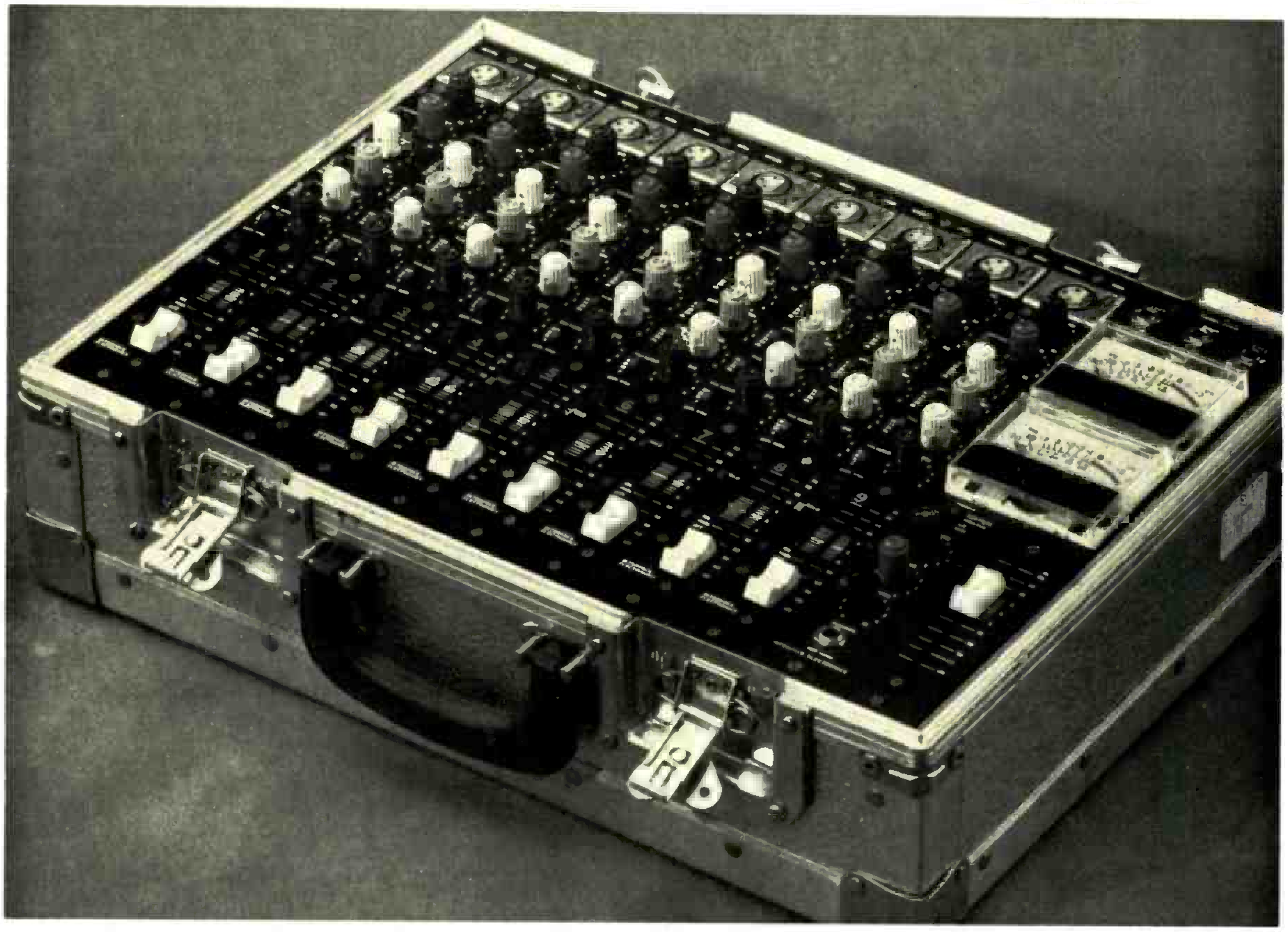

THE MODEL 200 MIXER is a special purpose compact "suitcase" portable mixer built in an ATA-style case with detachable lid and handle measuring 13 X 17.5 X 7 inches with eight or nine inputs and two stereo outputs plus echo and cue, and can be operated for 16 to 20 hours on an external rechargeable "Gel-Cel" battery (supplied, with charger). Built to the highest professional standards, it is recommended for the most exacting portable mono or stereo work.

THE SPECIAL MODEL 200 INPUT MODULES provide balanced XLR input connector (on panel) through four position input pad to transformer, to provide easy phantom powering and high rejection of RF interference, phantom power switch , phase reverse switch, four position input gain set switch plus a 20 db gain trim thumbwheel, panpot, three equalizers with four frequency mid frequency select switch, four position low frequency rolloff (12 db/octave), solo to phones monitor (which works even when module is muted), on/mute/off switch (module draws no current when off), and Duncan professional conductive plastic attenuator with dust seal.

THE SPECIAL MODEL 200 MASTER MODULE provides a stereo slider master and two standard VU meters which can also be switched to read cue/echo or battery level; monitor phones level and phones jack, phone plug outputs for left, right cue, and echo, echo return jack, and echo return trim pot. The stereo monitor phones output switches automatically to one input when the "solo" switch on that input is thrown. Peak phones output into 600 ohms is 160 milliwatts.

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Denver: Walt Stinson at Listen-Up, (303) 778-0780
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trusive. It is therefore recommended that any limiting be done at the production stage itself. The least expensive and yet unobtrusive limiter I have found to accomplish this is the dbx 119.* These units are stereo, require only two knobs to adjust, and once set can be left relatively unattended. These units do however require a line amp to bring them up to broadcast operating levels of +4 or +8 to allow them to drive a 600 Ohm line directly.

Although the dbx 119 is an excellent inexpensive limiter, in the B.B.A.T. recording van our current choice of limiter is the UREI LA-4. Two are required and can be coupled for stereo. These units do an excellent job and do not require any external equipment to interface with the broadcast equipment.

Other Ancillary Equipment Needed In The Radio Control Booth

As in the conventional recording studio, there must be some means of monitoring what is being done. A simple selector switch allows the engineer to monitor the local mix, the off-air sound (the signal actually being received by the radio audience and the only way of judging the final product), the tape being made (as even in the best of all worlds live presentations are sometimes pre-recorded for later broadcast), and a solo monitor which allows listening to each input on the console without having to inject it into the mix or disrupt that input if it is already in the mix.

Monitor Muting

The booth monitor should be capable of being muted whenever a microphone in the booth is active. This is accomplished through a simple relay in series with the booth monitor speaker lines.

Headphones

Like a conventional sound studio, the radio studio should have some provision for a headphone feed to the musicians and/or actors in the case of live radio drama. The easiest way to approach this is to sum the left and right output of the console through a single channel with amplification and out to the headphone distribution system (see Figure 2).

Other equipment needed for live radio broadcast is a simple communication system between the floor manager in the studio and the production engineer in the booth (see Figure 3). Also useful are signal lights to indicate to the musicians and host when they are on and off the air.

IN-STUDIO MICROPHONE TECHNIQUES

Miking and miking techniques are a highly personal and subjective area: therefore, this article will not attempt to set any hard and fast rules. It will, however, explore live radio miking techniques through a discussion of problems and solutions encountered when the author produced a Big Band program at KLON, Long Beach, California.

As live music at KLON began with the *Big Band* show, the first set-up explored is a typical live "big band." When there are a dozen or so musicians in the studio at once, leakage from instrument to instrument is inevitable. Rather than trying to avoid it, it is best to simply use it to advantage.

Figure 4 is representative of a big band set-up. The miking used is a mix of old and new techniques. The horn section is miked relatively "loosely." The reed section is miked a little more tightly, and each member of the rhythm section is miked individually. This set-up seems to work both for straight ahead ensemble charts and for modern charts with equal success. Utilizing this miking technique, a band could do anything from Glenn Miller to disco. A point of interest is that the piano, which could conceivably be brought into a single input, is miked in stereo. The reason for this is that on some songs with nothing but a piano and a vocal this permits the image to be "spread out" a little for stereo. Also, of interest is the miking used on the horn section. The B&O ribbon mikes have a figure eight pattern; and two trumpets and two trombones are miked with each unit. By having the musicians direct the bell of their horns at their assigned mikes, phase cancellation and cone filtering are kept to a minimum.

The mikes chosen for the reeds are Electro-Voice RE-10s. These have been chosen because they have excellent off-axis rejection of sounds thereby helping discriminate against the drum kit. Drum leakage in a live room is a problem in the best of all studios and is exacerbated when up to 12 mikes are open at a given time in a single room. The drums present the only serious leakage we've ever encountered. The drums themselves are mikes with a single PZM-150 (Pressure Zone Mike) taped to the inside of the drum baffle. The drum baffle itself is a simple structure made out of two 4'x 4' plywood sheets; one sheet used as the face and the other cut in half and hinged on either side of the face to act as sides. By utilizing a PZM on the baffle itself, first order reflections from the baffle and other attendant cone filtering and phase cancellation is avoided. Drum balance is determined entirely by the player. A plywood baffle is chosen so as not to destroy high frequency harmonic content of the various members of the drum kit. The overall effect is a live, "tight" drum sound.

Miking on the bass is determined by whether the instrument is an acoustic or an electric. In both cases, the instrument is fed into an amplifier to be used as a live monitor for the rest of the rhythm section. In the case of an electric bass, it is simply channelled through a direct box and into the mix. Leakage into the room negates the need to place a special mike on the amplifier speaker to round out the bass sound. In the case of an acoustic bass, although it is fed into an amplifier as a room monitor, the broadcast miking is accomplished by placing a PZM 150 tie tac mike through the "f" hole inside the instrument under the highest string. This not only affords a high degree of isolation from other instruments in the room, but provides a phenomenal bass response.

Electric guitars are miked in a rather conventional manner by placing either a Sennheiser 421 or an Electro-Voice 666 or any other high quality dynamic mike at the edge of their speaker. Dynamic mikes are used because they are difficult to overload and do not require tying up a phantom power service. For live broadcasts no appreciable increase in quality has been observed using a condenser mike on the electric guitar.

Another interesting way of miking an electric guitar is to put the hard side of a baffle facing the amplifier with a PZM plate taped to the baffle surface facing the speaker. Vocals are generally mikes on an AKG C-414-EB. Although virtually any high quality condenser or dynamic mike can be used, wind screens are always recommended for voice work, especially with condenser mikes which are prone to blasting and popping with close work. One problem we have noticed on *Live Music 1980* is that the venerable Neumann U47 tube mike tends to pick up a large amount of garbage through the body of the mike itself when there is a loud group in the studio.

Recording Rock

A rock set-up is a little more complicated as almost every instrument must be miked individually. Also, all effects which in a studio environment would be added later must be incorporated immediately into the live broadcast. In the diagram show (Figure 5), the miking is fairly straightforward. RE-10s were chosen for all vocal positions. Their

** The dbx model 119 has been superseded by the model 118, pictured.*

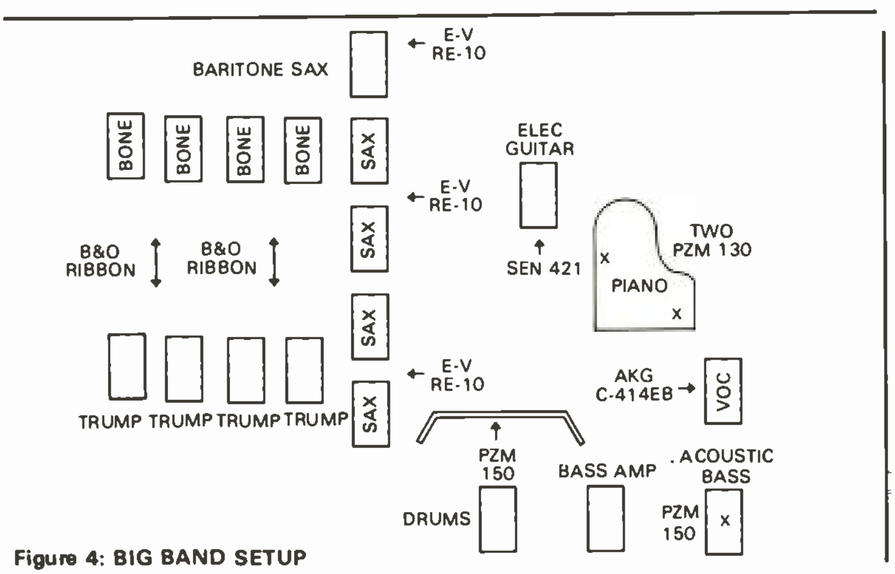

Figure 4: BIG BAND SETUP

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New Audio Techniques For The Old Art Of Live Radio Broadcast

REMOTE RADIO BROADCAST SET-UPS

Not all live radio originates from a studio. Many shows originate today, as they did in the 1940s, from hotel ballrooms, clubs, and outdoor summer facilities. This form of radio presents a number of problems that the technical producer never dreamed of in the studio. They can be grouped into three areas: acoustic, logistical, and electrical.

Acoustic

The acoustical problems in a remote broadcast could range from the Goodyear Blimp that floats overhead during the most crucial passage in a live opera performance out-of-doors to the heckling drunk uttering obscenities at the star of your favorite punk rock group. More often than not they lie between these extremes, restricting themselves to the clinking of glasses, the jingle of the cash register, and the incessant opening and closing of the restroom doors always located strategically next to the stage. The only way to eliminate acoustic noise is to mike the musical group as closely as possible; and, if ambience mikes are used, only bringing them up in the mix during the applause and sequences of audience participation. As in the studio, if low frequency garbage is a problem very often the use of the UREI subsonic processor #501 can save the day.

Out-of-doors, close miking techniques can be used for rock and jazz application but prove to be impractical for symphonic work. Other than bringing up mikes only when necessary, there is little that can be done out-of-doors in a symphonic setting to eliminate the extraneous noises other than rely on the good will of the gods. Another acoustic problem encountered in the live remote is the use of the live sound reinforcement system. The radio producer and director must work closely with the house sound man to insure that as much as possible the house sound reinforcement system does not interfere with the recording system. It should be stressed that any squeals, clicks, pops, or feedback that originates in the sound system will be very clearly audible in the broadcast system. A good rule of thumb to follow is that any mistakes made by the recording engineer will not generally be audible over the sound system, while any mistakes made the sound engineer will be audible on the recording.

Logistical Problems

One of the most critical logistical problems encountered in a remote broadcast is insuring access to the facility far enough in advance of the concert to insure that your set-up procedure will not be rushed and frenetic. The easiest way to eliminate this problem is to find who is responsible for the physical operation of the facililty and arrange to meet with that person several days before the broadcast. This will allow you the establishment of a relationship with this person ahead of time to view the facility and start to plan for the broadcast procedure. It is generally recommended that on the day of the broadcast the remote crew gain access to the facility a minimum of two hours before air time if there is no rehearsal, and a minimum of four to five hours if a sound check is anticipated. It should be stressed that these are minimum requirements. In a live broadcast situation, the more time for set-up and preparation the better.

As Barker Brothers Audio Technologies utilizes a remote van for its live work, we will discuss cable routing and logistics in that light. The methods utilized are equally applicable to anyone who must set-up, for example, in a spare dressing room or office. The only additional encumbrance for an individual who does not possess a van would be the need to locate a control area. If worse comes to worse, the individual mixing the remote can set up the console in the same area as the performance and mix the broadcast utilizing a pair of very high quality stereo headphones. This is not recommended as the perception of imaging and instrument placement with a headphone monitoring system does not generally correspond with what will be heard over a spaced pair of speakers in a reasonable monitoring environment.

When routing cables it is generally recommended that you attempt to route them around walls and over the tops of doorsills. If cables must be run across large areas where the public is likely to be, it is a good idea to purchase several rolls of indoor-outdoor carpet to cover cables lying across public traffic routes. Although indoor-outdoor carpeting costs several dollars per square yard, it is less expensive than rubber matting or the legal action that could be filed against your company by an individual tripping over your cables if the carpet were not used. It should also be noted that B.B.A.T. was able to effect a large savings in their liability insurance premiums due to use of this precaution.

After the main cables are run between the house and the remote van, the next problem the remote producer will need to confront is the actual stage set-up. On B.B.A.T. remote sessions, the sound reinforcement console in the house and the remote van split all mikes common to both operations. This is accomplished through the use of Jensen JE-MB-C bridging transformers with a direct feed being sent to the remote truck and the transformer isolated feed being sent to the sound reinforcement system. This system seems to work very effectively and the internal Faraday shields within the transformer provide excellent isolation of common mode noise generated by any voltage potential differences on a two-mixer chassis. Each input on the recording snake also contains a ground-lift switch as well as the isolating transformer to eliminate ground loops within the system (see Figure 10).

In an emergency, the two systems may be interconnected through a hard-wired "Y" cable, but this leaves the system prone to picking up common mode noise and ground loops.

The B.B.A.T. recording van is designed to operate from either 210 Volts or 110 Volts, drawing a maximum current of 40 amps. The maximum current consumption of the van was intentionally kept low in order that the van could be operated from any voltage source. The AC power is delivered to the van via a four conductor, 10 gauge cable. This allows two hot runs, a common return, and a neutral to be run totally independently of one another. At the end of the AC cable provisions have been made to either tie directly into the building's main system with bus-bar clamps or to be split out to individual 110 Volt, 20 amp circuits via a simple splitting cable. The main cable itself terminates into a four conductor, 50 amp, four pin plug.

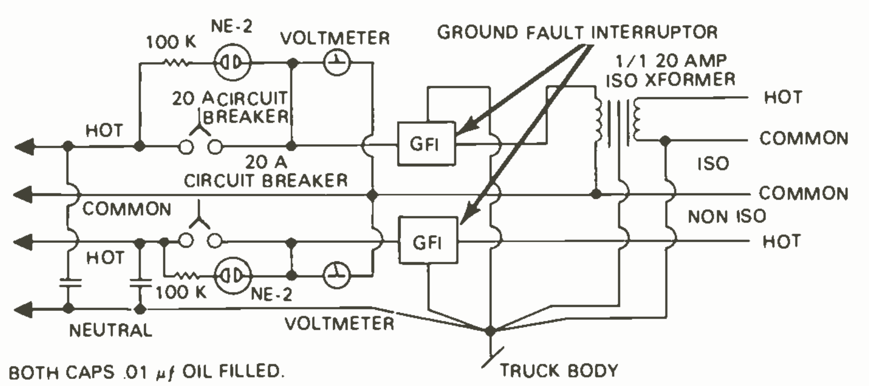
In the recording van, one leg of AC power is fed directly to the air conditioning, lighting, and non-audio electrical gear. A second 110 Volt feed is fed through a one-to-one isolation transformer with a built-in Faraday sheild to attenuate electrical noise and interference generated by machinery and equipment in the building. Both AC hot lines to the truck are shunted to their common lead by .01 1,000 Volt mf capacitors to reduce further any residual interference which might eke its way through the Faraday sheild in the transformer.

The truck is protected from overload by a pair of 20 amp circuit breakers and a ground fault eliminator on each hot lead entering the truck. This system has been found to work reasonably well and is also relatively inexpensive to produce (see Figure 5).

In remote work the communication system must also be expanded to facilitate communication not only to the stage crew, but also to the house sound reinforcement mixer. This normally consists of adding one additional headphone location to the previously discussed communication system utilized in the studio.

As the remote truck is normally not located within direct visual contact of the

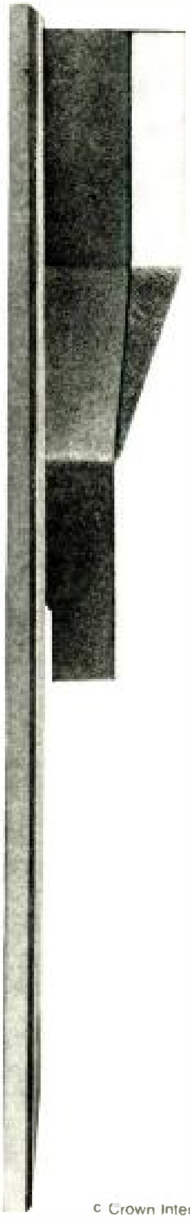
— continued overleaf . . .



FOR SINGLE PHASE 210 VOLT POWER SEND THE COMMON AND THE NEUTRAL ARE BOTH TIED TO THE NEUTRAL BUS AT THE MAIN POWER SEND POINT.

Figure 5: TRUCK ELECTRICAL

Introducing the Crown PZM,™ the second major advance in microphones in 100 years.



In 1876, Bell invented the first microphone.

Crown now announces the second microphone – the PZM.™

During the last century, microphones have been much improved, but they still employ Bell's basic concept: a movable diaphragm connected to a transducer, the whole assembly intended to be stuck out in the air somewhere near the sound source.

Comb filtering is a side effect of that design that cannot be eliminated. Every Bell-design microphone demonstrates frequency response anomalies because of an inability to satisfactorily combine direct and reflected signals. Phase-induced amplitude cancellation and reinforcement are the inevitable result.

Crown PZM microphones eliminate comb filtering from the primary boundary because they detect sound according to a new principle, the Pressure Recording Process.™ As a sound wave approaches a boundary (wall, table, floor) a pressure field four or five millimeters deep forms at the boundary, within which the direct signal and its reflection from the boundary add coherently and remain in phase.

The Crown PZM™ places a small pressure transducer into the primary boundary pressure zone, eliminating the possibility of phase-induced interference. The PZM concept thus provides a significant improvement in signal quality. Its small profile also improves microphone aesthetics.

The PZM pickup pattern is hemispheric, with no "off-axis" position.

Singers and speakers can move more freely around the PZM. Gain related to distance will change, but not tonal quality.

The PZM responds accurately to SPL up to 150dB. You can put it right inside a drum, a bass fiddle, or a piano.

The PZM hears whispered conversations in an ordinary room at thirty feet. In certain situations where undesired ambient noise can't be eliminated, or in halls with poor acoustics, the PZM probably should not be used – it will pick up everything.

Singers, orchestra conductors, pianists, percussionists, broadcasters have all tried – and praised – the PZM.

Recording engineers find that the PZM suggests new miking techniques. For small groups it now seems that the best place for a PZM is on the floor! Recording and reinforcement may well require fewer PZM mikes.

Several PZM models are now available, including a clip-on and recessed model for permanent installation.

The PZM is changing ideas about how a microphone ought to sound, look and be used. Find out for yourself how it might improve your own recording or reinforcement systems.

Write for information on the PZM. Or call us at 219/294-5571.

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for additional information circle no. 69

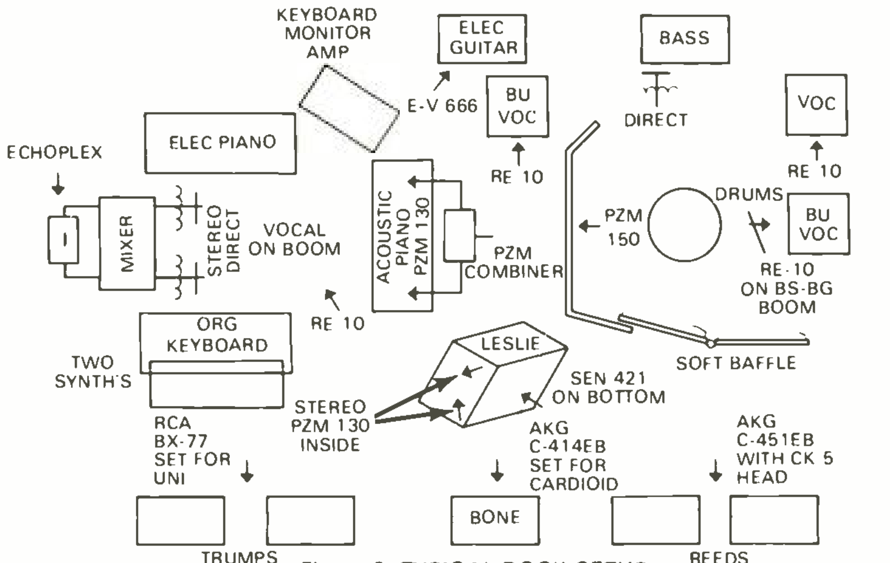


Figure 6: TYPICAL ROCK SETUP

hyper-cardioid pattern and smooth response independent of working distance makes them an ideal choice for live radio production. They are also, in the Electro-Voice tradition, nearly impossible to destroy. In addition, their hyper-cardioid response makes them less sensitive to leakage from other instruments which is a particular problem in a loud rock set up.

The trumpets were miked with an RCA BX-77 chosen for its smooth sound on brass and the fact that it could be set for a unidirectional pattern to help discriminate against leakage in the studio. The trombone was miked for an AKG C-414-EC chosen for its wide frequency response and its ability to flatter that instrument.

The reeds were miked with an AKG C-451-EB with a CK-5 capsule chosen for its ability to flatter the reed and to some degree discriminate against popping and blasting which sometimes occurs when a musician leans into the mike causing mechanical and breath noises.

The acoustic piano was miked with a pair of PZM 130s inside the piano. These were connected to a combiner and brought into one channel of the mix.

The Leslie organ speaker was stereo miked with a pair of PZM 130s placed on either side of the rotating horn inside the top of the cabinet. The low frequency rotor was miked with a Sennheiser 421, and the organ was then covered on three sides with a packing blanket to reduce leakage. One side of the organ facing the keyboard player was left open so that the performer could hear his instrument. The two synthesizers and the electric piano were mixed by the keyboard artist on a TAPCO 6200. The echo output of the 6200 was fed into an Echoplex and routed back into the mix. The stereo output of the keyboard mixer was fed to a pair of direct boxes and the monitor output to a guitar amp placed near the keyboard player as a personal monitor. The electric guitar was miked with an Electro-Voice 666 placed at the corner of the speaker. The bass guitar was taken direct; once again, leakage in the room precluded having to mike the bass to round it out. The drums were miked with a single PZM 150 taped to the inner face of the baffle.

It might be interesting to note by careful scrutiny of the diagram in Figure 6, that all of the vocal mikes are facing away from other instruments to minimize leakage.

Acappella Jazz Choir

An interesting but simple set-up utilizing a pair of PZMs is the miking for an acappella jazz choir. The two PZMs are affixed to each side of a 2' x 3' square plexiglass plate. The choir then simply surrounds the plate and when somebody wishes to do a solo they simply step forward and sing within a few inches of the plate.

The results were truly impressive and

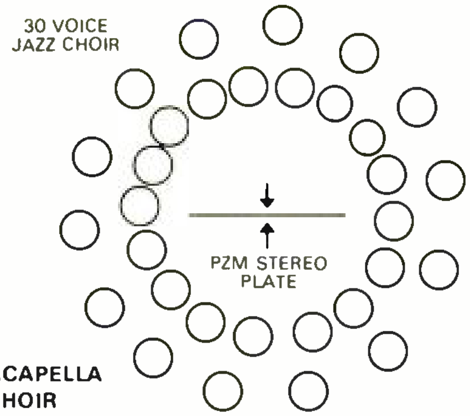


Figure 7: ACAPELLA JAZZ CHOIR

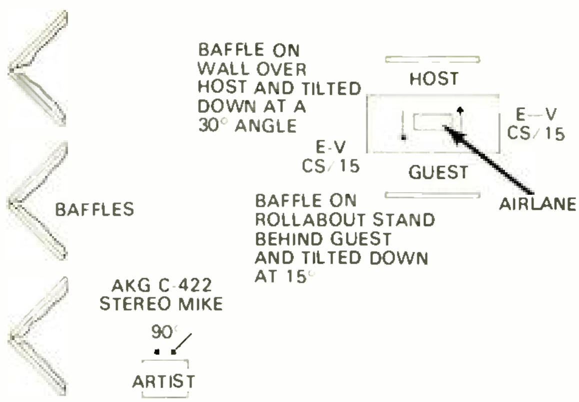


Figure 8: HOST POSITION AND SOLO ACOUSTIC ARTIST

produced a true 360 degree stereo pick up with no dead zones. This type of pick up is also applicable to stereo radio drama although this author has not yet had the opportunity to explore the possibilities of this set up.

The stereo PZM plate has also been used successfully on non-electrified blue grass and blues and should work on virtually any set-up in which a natural acoustic balance prevails (see Figure 8).

Acoustic Guitar — Solo Concert

The set-up shown in Figure 9 was used for a classical solo acoustic guitar concert. This set-up could use either a stereo mike set as a coincidental pair or a PZM stereo plate. As the PZM stereo plate has already been discussed, we have drawn the diagram with a conventional stereo pair. Both methods have proven to be successful. The PZM plate appears to have a slightly more transparent sound, but the conventional coincidental pair of cardioid mikes has a greater capability of discriminating against noise in the studio. The guitar was miked with an AKG C-422 stereo mike. The two capsules were rotated in 90 degrees opposition to one another, and each capsule was offset 45 degrees off-axis from the artist. The patterns for the two capsules were then set in the cardioid position. This produced very realistic stereo and at the same time a very intimate sound. With a single artist, large 6'x 8' baffles are sometimes placed near the artist to give a more intimate sound. This technique tends to hold valid for any single acoustic artist.

Host And Guest Interviewing Mikes

Also shown in diagram 9 is the table where the host sits and interviews the guest. This area is set up in a corner of the studio for virtually all emceed live music shows although it was not diagrammed in the other figures. The interview area is set in a corner of the studio to be out of the way. In order to produce a more intimate sound, baffles were placed above the host and behind the guest. These were tilted down at different angles so as not to introduce any parallel surfaces and their attendant standing waves. One problem encountered when the use of baffles was first introduced was that the mikes which had been intentionally wired out of polarity with one another to reduce low-frequency noise from the studio had to be re-wired in polarity due to the much earlier arrival of first order reflections from the baffles. The overall effect has been successful. The baffles produced a close, more intimate sound and eliminated the cavernous effect of the large studio.

stage, a video monitoring system is also recommended. B.B.A.T.'s facility is capable of accepting two camera signals displayed on a pair of 7" Conrac video monitors placed over the console of the remote recording van. This allows the use of one camera to monitor the stage and a second camera to monitor the host or band leader/conductor depending on the needs of the particular remote in question. The cameras utilized for this are simple Panasonic black and white surveillance cameras with attached viewfinders for focusing and aiming. The use of electronic viewfinders on cameras allows one person to set up the video system without the need for assistance.

In a rock or jazz remote, mikes and mike stands are generally acceptable by both the show producer and the audience and present no special problems. However, in a symphonic or operatic remote, a stray mike stand on stage is likely to drive the artistic director absolutely berserk. B.B.A.T. has overcome most of these problems by either flying mikes from the ceiling of the facility by their cables or by utilizing PZMs on the floor of the stage.

The communications problems and interface problems for symphonic and operatic work are essentially the same as those encountered in rock or jazz work.

After the concert has ended, provisions should be made with the facility to insure that lighting and power are available to take down. Also, all equipment used on a remote should be clearly labeled to eliminate confusion during the take-down, which is often hectic.

At this point we will discuss a couple of remote set-ups, the first being a typical big band set-up which includes a full 17 piece orchestra, a vocal soloist, a table for a host-announcer and a guest, and ambience mikes for the audience. It should be noticed that the set-up does not differ dramatically from the set-ups in the studio. The only major difference being in the layout of the musicians which makes it necessary to use an additional pair of ribbon mikes on the horn section. As in the studio set-up for *Big Bands*, B&O 53 ribbon mikes were used for the trumpets; the trombones were then miked with a pair of Beyer M-360C ribbon mikes. The vocal mike used on a remote is generally of a dynamic and not a condenser type; and, in the case of the remote, instead of a single PZM being used on the entire drum kit, the PZM 150 has now been placed inside the kick drum and the upper portion of the kit is miked with a single AKG 422 stereo mike. You will also notice that a pair of PZM 130s on 2' x 2' plexiglass plates have been hung on the left and right side of the hall facing the audience as ambience mikes. These were chosen because PZMs exhibit virtually the same response both on and off axis and produce a very realistic sonic impression of the hall. Other than these changes, the miking of a remote *Big Band* is not very different from the studio.

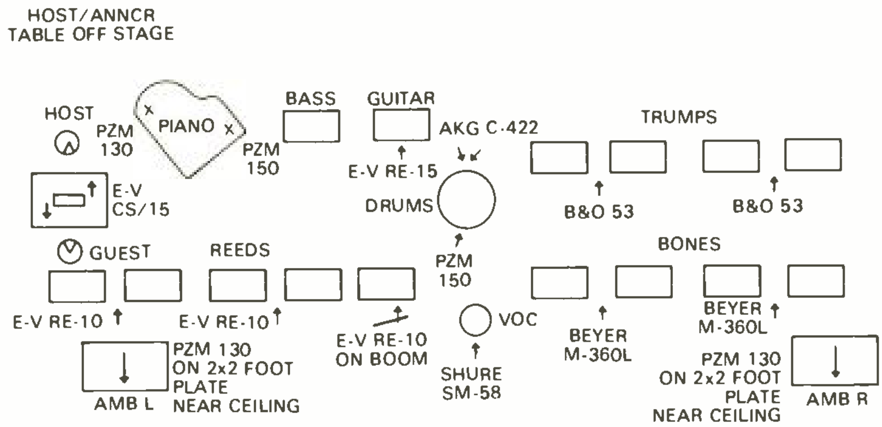
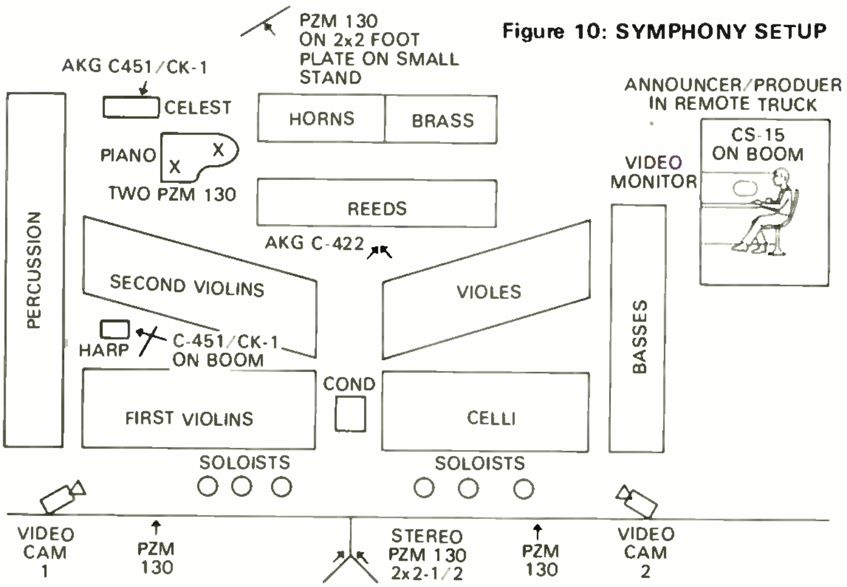


Figure 9: TYPICAL BIG BAND SETUP FOR LIVE REMOTE

Next, is a discussion of a typical set up for out-of-doors symphonic and partially staged opera performance. This set up is chosen to represent one of the more complex symphonic set-ups. The main mike for the system actually consists of a pair of PZM 2½s placed back-to-back. This forms a nearly coincidental stereo floor mike. The orchestra is then flanked on the left and right with PZM 130s placed approximately three-quarters of the way out from the center line of the orchestra. This arrangement of PZMs give reasonably accurate coverage of the orchestra and requires only a minimum of supporting mikes to touch up the inner balances. The reeds were sweetened with an AKG C-422 single point stereo mike. The harp and celeste were each sweetened with AKG C-451 condenser mikes with CK-1 capsule, and the piano was miked internally with a pair of PZM 130s taped to the lid of the piano. A PZM 130 on a 2' x 2' plexiglass plate on a small mike stand was used to sweeten the horns. Its use was only necessary during horn passages and was not used at any other time during the performance. Normally, an AKG C-422 stereo mike would have been flown over the middle of the stage approximately 15' in the air and 5' out from the orchestra for the main pick up instead of using the stereo PZM 130s on the stage floor. However, visual considerations precluded the flying of any mikes. No audience mikes were used as the open-backed PZM 130s on stage left and right provided more than ample ambience. A Shure SM-57 hand held mike was used for announcements over the sound reinforcement system, but not connected to the recording chain and is, therefore, not shown here.

All mikes shown in the diagram were split between the house console and the recording console. Phantom power for the condenser mikes was provided by the recording truck. The announcer for the opera sat in the rear of

Figure 10: SYMPHONY SETUP



the van and was able to observe the action on stage on a pair of video monitors being fed by a pair of cameras left and right. The announce mike was an Electro-Voice CS-15, chosen for its crisp sound, and when used with its wind screen is relatively immune to popping.

Indoor symphonies utilizing the same orchestra layout is shown in Figure 10, but generally the only miking required would be an AKG C-422 stereo mike flown over the center line of the orchestra 15' up and 5' to 10' out over the audience. Hall ambience can be controlled by the pattern setting of the mike with a dual cardioid setting giving the driest sound and the dual figure eight setting giving the most reverberant sound. Unlike the outdoor set up, indoor orchestral work normally requires no sweetening mikes within the orchestra.

Video monitoring for the announcer is handled in the same manner as the outdoor performance. Another form of miking indoors is the utilization of the mid/side system as proposed by Blumlein.[2] Although the Blumlein system tends to require either more equipment or more inputs on the console, it does offer the advantage of reasonably accurate control over stereo image width. Either the Blumlein system or the coincidental pair are both perfectly satisfactory; and the decision as to which to use will have to be made as a function of the hall acoustics.

In conclusion, this article has explored methods of live radio production being devised by Barker Brothers Audio Technology. The methods utilized are equally applicable to the small studio or video shoot, and all miking methods shown provide excellent mono compatibility.

This author believes that in the decade of the '80s live radio production will again come into its own, opening wide a new technical and artistic area that will be the legitimate stage of the trained audio technician and artist. □□□

References:

1 · Everest, F. Alton, *How To Build A Small Budget Recording Studio From Scratch*, Tab Books #1166, Blue Ridge Summit, PA 17219. 1979.

2 · Blumlein, Alan Dower, *"British Patent Specification 394,325,"* Jour. AES, April 1958, Vol. 6, No. 2, page 91.

Bibliography

1 · Blumlein, Alan Dower, *"British Patent Specification 394,325,"* Jour. AES, April 1958, Vol. 6, No. 2, page 91.

2 · Everest, F. Alton, *How To Build A Small Budget Recording Studio From Scratch*, Tab Books, #1166, Blue Ridge Summit, PA 17219. 1979.

3 · Everest, F. Alton, *Acoustic Techniques For Home and Studio*, Tab Books, #646, Blue Ridge Summit, PA 17219. 1973.

4 · Mosely, John, *"Eliminating The Stereo Seat,"* Jour. AES, Vol. 8, No. 1, January 1960.

SOUND MAN'S GUIDE TO VENUES

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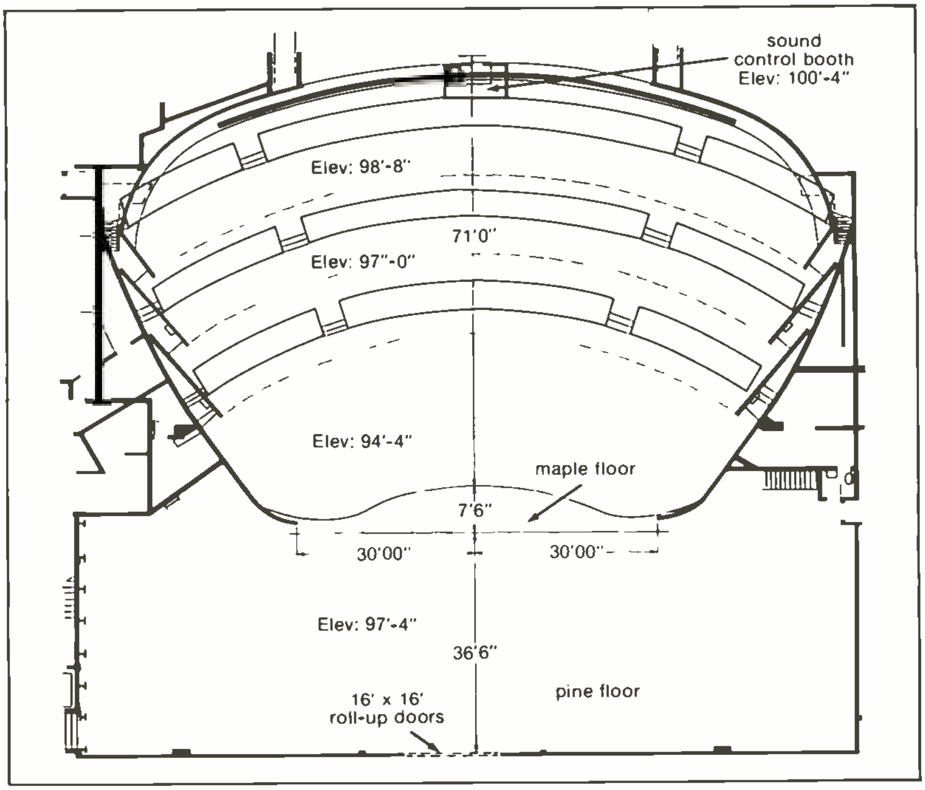
- number 12 in the series -



Driving Directions From Airport

MGM Grand is located approximately two miles north of McCarren Field. Take Paradise Road north to Flamingo Road. Turn left (west). MGM is ½-mile down on the left at South Las Vegas Boulevard (The Strip).

Interstate 15 is located ¼-mile west of the hotel.

Facility

Entertainment showroom seats 1,450; no balcony. Room is open for inspection 24-hours daily. Call for appointment to see backstage area. No specific ending time for shows. No orchestra pit. Level prescenium stage is 60' 5" W x 40' D and is 32" above audience floor. Seating begins right at edge of stage. Grid is about 30' above stage. Proscenium arch is approximately 20' above stage.

Acoustics

Fairly dry and even in the house. Tends to sound louder and brighter with a small crowd than it does with a full house. Stage acoustics are fair to good. No RT_{60} measurements available.

Loading

A 16' x 16' roll-up door at back of stage opens onto a hallway directly across from a second 16' x 16' door which opens onto the truck level loading dock on the south side of the building. This dock area is for loading and unloading only — trucks must park in the MGM lot at rear of building.

Setup

Permission to install a mixing console in the house must be obtained in advance from the Entertainment Department. Mixer location is about 70' from stage in center of house next to house console in section 14 or 19 and measures approximately 7' x 7'. A mult cable at least 150' in length must be run through a trap door at the house mix position under the floor of the showroom and then up through a 5" diameter conduit in the floor upstage against the back wall. A grounded AC circuit is available at the house mix position or anywhere backstage.

Permission to setup speaker risers in the audience area next to the stage must also be obtained in advance. Two metal risers approximately 8' W x 5' D x 6' H are available and their use blocks about ten seats.

Sound System

- house speakers -

Two-way bi-amped system located in a single permanently mounted central overhead cluster. Twelve bass speakers consist of Altec 421 and 421 LFs; some of which are housed in Altec 9816 cabinets. High frequency horns consist of 8 Altec 203s and 8 Altec 803s, all driven by Altec 288-8G drivers. Six Altec 1590B amplifiers power the woofers and another 4 Altec 1590Bs drive the horns. Overall EQ is set via an Altec 1650 1/3-octave equalizer.

- house console -

Quad-Eight Electronics board has a total of 40 mike or line inputs, 4 submasters, and 4 outputs. Sixteen of the inputs have a 2 band EQ section — the remaining 24 inputs have no EQ available other than overall EQ on whichever submaster they are assigned to. Console can accept a balanced line level signal on a male 3-pin XLR connector.

- monitor system -

Monitor speakers consist of 6 Altec 1219 bi-amped speakers and 4 Shure SR-112s. House monitor mix is done from the main house PA board utilizing three of the echo send busses. EQ is the same as whatever is going out over the house system.

- microphones and stands -

All Shure mircophones: (12) SM-58; (28) SM-53; (2) SM-7; (1) SM-57, and (12) SM-17 lavalier mikes.

All Atlas mike stands: (30) MS-12C; (4) MS-12s; (8) short, plus assortment of boom arms and goosenecks.

Altec 1650 1/3-octave equalizers used in house and monitor systems. AKG BX-10 reverb chamber available.

Electrical

Three phase, 600 amps total. Separate power panel for sound feed. Main breaker box is 10' from stage and requires pigtail connections. Actual voltage measures 120 VAC. SCR lighting equipment used.

Personnel

Union house; departmentalized. Separate crew call to load trucks not necessary.

Tech Director: Don Nelson, (702) 739-4695.

Stage Manager: Kelvin McWhinny, (702) 739-4111.

Head Sound: Gary Nellis, (702) 729-4111.

Chief Electrician & Lights: Len Raden: (702) 739-4694.

Piano Tuner: Ron Hammond, (702) 870-7440.

Traveling Soundman Reaction

"Coverage of house PA is pretty good — improved recently with the addition of extra bass speakers. Relatively dry sounding room is easy to control and overall sound is pretty comfortable for the average Las Vegas type show. I would like a little more extreme top end, however — say from 10 kHz — and on up. Head soundman Gary Nellis is very knowledgeable and helpful. Very easy to load in and out.

"Don't care much for the house mixing console as it only has EQ on 16 channels. Additional side speakers have been brought in by other Stanal engineers on occasion and have been used mainly as support for the center house cluster. They also helped lower the apparent sound source down to stage level." ***Jim Anderson, Stanal Sound.***

FEEDBACK ISN'T ALWAYS A DIRTY WORD!

The information contained in these surveys is as accurate as possible at the time of printing. However, there will always be changes and improvements made to in-house sound equipment and acoustics as more and more venues become conscious of high quality sound. Many of these halls rely on the comments and reactions of visiting engineers such as yourselves and make changes accordingly. So, if you should come across a situation that you feel is contrary to what is printed, please drop me a note and I'll print an update. Also, if you have any reactions to a venue (pro or con) that you'd like to see surveyed in an upcoming issue, please address them to:

PAT MALONEY
RECORDING engineer/producer
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You can tri ours with a lot less... than you can bi theirs.

Bi-amping and tri-amping (electronically dividing the frequency spectrum into low-high/low-mid-high bands and amplifying them separately) are quickly becoming the excepted methods for insuring the low levels of intermodulation and harmonic distortion so necessary for clean, accurate reproduction of today's music. Unfortunately, this involves the use of multiple power amplifiers and electronic crossover networks — usually a substantial financial burden for the average musician or sound engineer.

To help ease that burden, we have incorporated circuitry for **two** separate and independent electronic crossover networks into our CS Series stereo power amplifiers. Bi-amp operation is enabled by merely selecting and plugging in the appropriate crossover module. Utilizing an additional power amp makes tri-amp operation possible. In addition, we have included our unique DDT® compression circuitry to virtually eliminate amplifier clipping and the associated headroom problems, transformer balanced input capability, and two-speed forced air cooling.

There are many good power amplifiers on the market today, but not one can match the quality, versatility, performance, and, most importantly, the

incredible dollar value offered by our CS Series. We invite you to compare specs, features **and** prices. You'll see why so many are choosing Peavey.

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SOUND MAN'S GUIDE TO VENUES

- number 13 in the series -

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Driving Directions From Airport
Exit from airport onto 82nd Avenue. Drive south on 82nd to Banfield Freeway (I-84, previously called I-80). West on I-84 across Morrison Street Bridge to Broadway. Left to Main Street. Allow 45-minutes drive time.

Facility
Indoor auditorium (old vaudeville house). 1,606 seats in orchestra, 1,354 in large balcony. Not normally open during the week; call for appointment. No enforced ending time for shows. Permanent proscenium stage has level floor and measures 52' W x 30' D. Stage is 4' above audience level. Orchestra pit area can be used for additional seating or for a small dance floor and measures 52' W x 20' D. First row of seats is about 25' from stage if pit is used as dance floor; otherwise portable seats start 4' from stage. Grid is 82' above stage. Proscenium arch is about 30' above stage level. 50 pipes are available over stage for hanging mikes, speakers, lights, or backdrops.

Single balcony is fairly large and steep and begins about 40' from edge of stage above row N on the main floor.

Acoustics
Typical vaudeville/movie house sound: live and reverberant with a tendency toward bass heaviness. No RT_{60} measurements available. Recommend flying in side wings and teaser curtains above stage to help contain stage sound. Light fixture shells under balcony create standing waves in sections of seating on main floor.

Seats are all padded; some more than others.

Loading
Loading door at top of short, steep ramp off stage right measures 10' x 10' and opens onto Park Street at rear of theatre. Theatre exit doors on Park Street connect to a hallway inside the theatre which facilitates bringing the mixing console into the orchestra seating area. Truck parking available on Park Street behind theatre.

SOUNDMAN'S GUIDE to VENUES
*is a series being compiled by **R-e/p's** sound reinforcement consulting editor, **Pat Maloney,** whose full-time profession is as an internationally recognized sound reinforcement engineer/mixer. The series is the result of a questionnaire Pat developed to be sent to performance venues in anticipation of the start of a concert tour. The information returned by the venue is considered vital to pre-planning the tour. Periodically **R-e/p** will offer an updated collection of the reports published. — ed.*

Setup
Area reserved for mixing console is in Section B, Rows N-O-P, and is located about 50' from stage. 100' cable is necessary to reach center stage via Fire Marshall approved routing. Closest grounded AC outlet is on stage. Boards are placed over seats to provide table for mixing console. This console area is just under the front edge of the balcony but is not subject to the typical "under balcony" compressed sounds. An alternate mix position is from the back of the house, although the sound does get compressed somewhat.

Sound wings are built out from stage at extreme sides in front of fire curtain and measure approximately 12' W x 8' D and are at stage level. About 40 seats are blocked if these wings are used. Five risers measuring 1' H x 4' W x 8' D are also available and are usually used for drum risers.

Theatre can supply equipment and personnel to hang speakers in front of proscenium on each side of the stage if sufficient advance notice is given.

Sound System
No in-house sound system. Theatre recommends calling either Sundown Sound, (503) 232-2298, or Northwest Sound, (503) 286-9411. Both companies have done numerous shows at this venue.

Electrical
Two separate 3 phase circuits at 600 amps per leg are located in a room directly beneath the stage. 100' electrical cable with pigtail or lug connectors is necessary to reach breaker boxes from the normal amp rack position off stage right. A separate single phase 200 amp service is also available for a total of 3,800 amps into the building. Nine separate 20 or 30 amp circuits are also available in stage pockets to power musical instruments, amps, or other sound equipment if necessary. Actual AC voltage is at least 120 VAC, and often around 127 VAC. This is higher than most halls, and I'd recommend re-setting the voltage switches on equipment so equipped. The high voltage is probably due to the fact that the actual power company transformers are located inside the building next to the breaker boxes.

Personnel
House usually employs two union men (chief electrician and a lighting tech), and calls for a non-union stagehand crew for loading and general stage work.

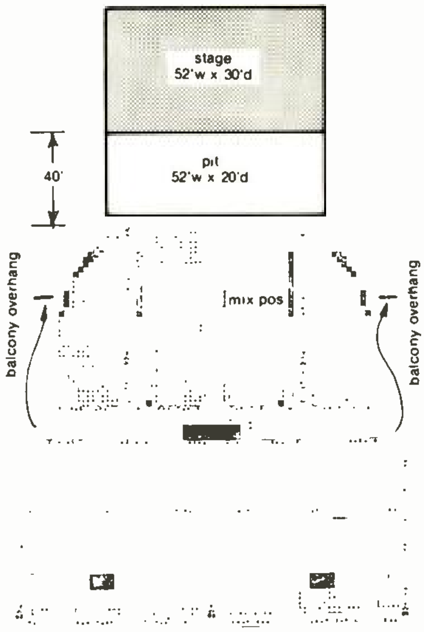
Building Manager: Michael McManus, (503) 226-3619.

Stage Manager: Bruce Chaddack, (503) 232-4937.

Chief Electrician: Willis Holland, (503) 226-0034.

A piano tuner is available (Don Flemming), and can be contacted through the theatre.

Traveling Soundman Reaction:
"Not a very flat room — reverberant and boomy in the low end. I usually use quite a bit of overall EQ here. The areas around 160 Hz, and 315 - 800 Hz, are especially hot and need to be brought down 3 to 5 dB. It gets a lot better with a full house though. Don't mix from under one of the light coves as standing waves are created here. The sound varies thoughout the house so I like to set up the console next to an aisle so I can step out into it every now and then during the show to get a different perspective on the sound.



"On jazz or blues shows where the level doesn't have to be extremely loud I'll often mix from the back of the house. I don't take up as many prime reserved seats this way and the sound is still pretty representative of the rest of the house at these lower levels — although a bit compressed. Access to the stage from the balcony is difficult so I wouldn't recommend setting up the console there.

"Load in and out is not real bad, but you must remember that it takes three or four stagehands to safely move anything up or down the ramp instead of the usual one or two. Also, the non-union crew members tend to be young and inexperienced for the most part. Calling in a few extra men is always helpful." ***Dave Cutter,** Sundown Sound.*

"Acoustically not bad, but the walls are all hard plaster and the hall tends to get boomy and very reverberant if the level is too loud. Jazz, blues, light rock, etc., sound best in here — you can get a good sound if the level is held down a little. Somebody like Ted Nugent would not work out too well here, sound wise! A mix position just back from the edge of the balcony seemed to be about the best location — far enough back so no one could drop anything on you and not so far back that you'd get too much of the "under balcony" sound. The crowd soaks up a lot of the sound here even though the seats are padded, so I set a basic EQ during the day but save the fine tuning until the place fills up at night. If the crowd is small the sound will lack distinction and you'll have to keep the level down to minimize excessive reverb.

"The crew is used to load in here, but it's still a bit difficult due to the steep ramp. Crew tends to be young and inexperienced, but getting better all the time. Recommend adding two extra men to the crew." ***Ron Sarver,** Independent.*

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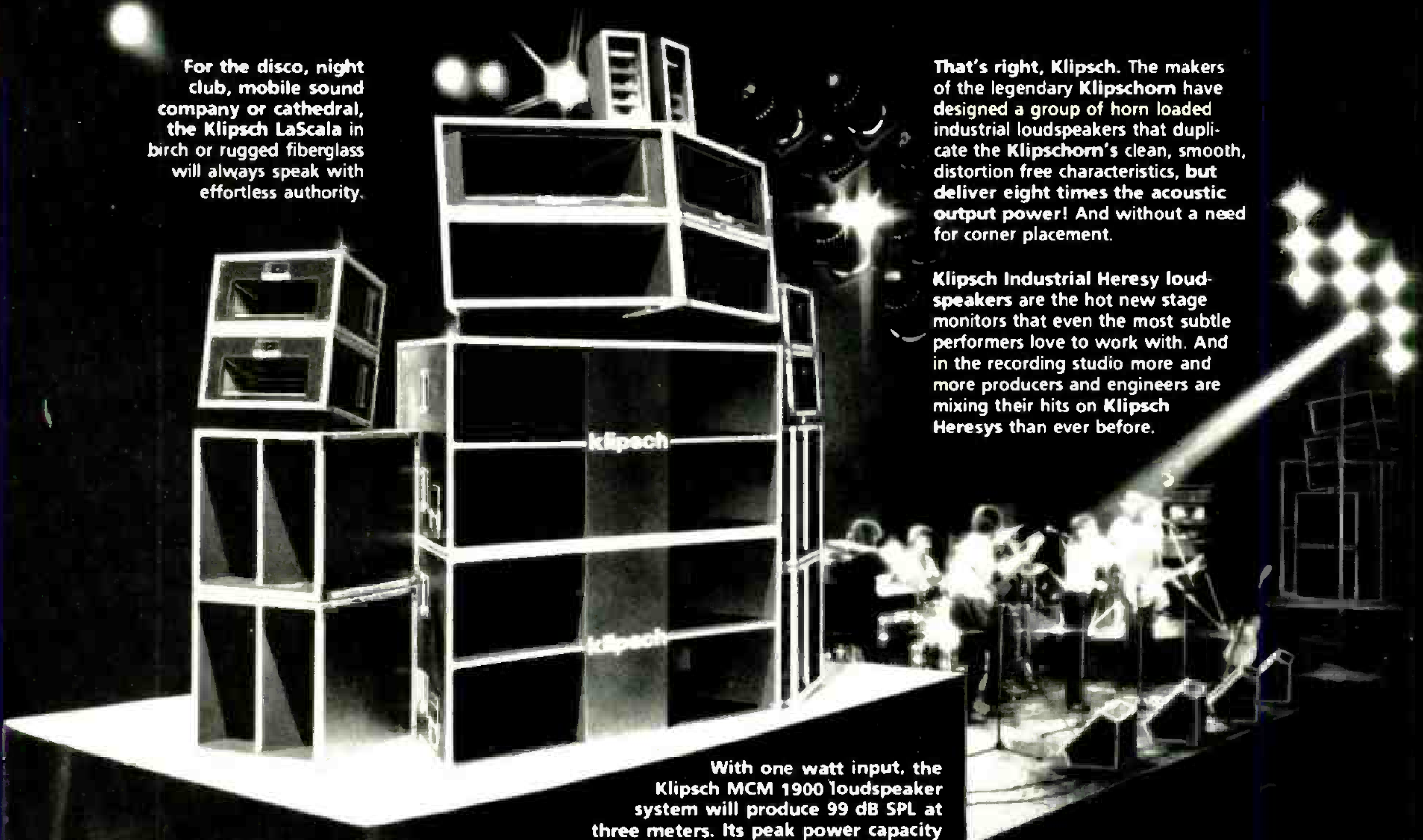

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a standardized mobile snake splitter system

by Chris Michie

When JACK CRYMES became chief engineer for the Wally Heider Mobile in 1972, one of the most difficult problems involved with remote recording had to do with successfully splitting the microphone feeds to three separate mixers; the house, monitor, and recording consoles had to have access to any or all of the microphone lines. PA companies and touring groups commonly used custom stageboxes and snakes which were essentially hard-wired for their systems only, and the splits were not usually transformer-isolated. The onus was (and still is) on the remote truck to derive their feeds with the minimum disruption to the existing setup.

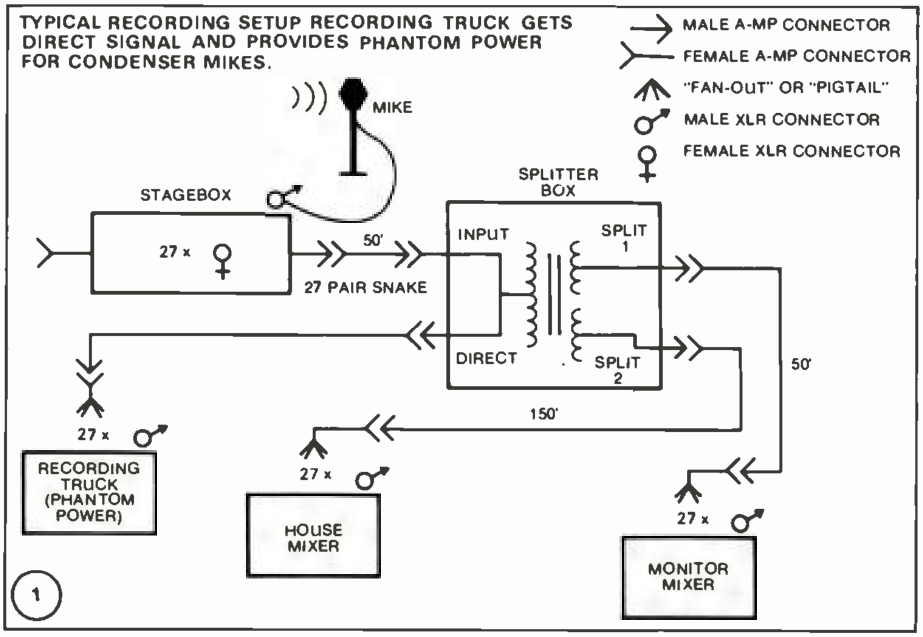
Other than double-miking, which has obvious drawbacks, the usual method was to use large numbers of splitter boxes, usually situated next to the PA stageboxes. As Jack Crymes recalls: "Every show was a nightmare. It was not uncommon to have piles of cable three feet deep around the stageboxes and it took two men to trace a particular cable." Jack decided to design a snake splitter system that would be quick to install, extremely flexible in terms of its configuration, and would provide the necessary number of splits with provision for lifting ground (pin 1) on individual mike lines to eliminate the ground loops that plague such set-ups.

> · the author ·
> *Chris Michie's first involvement with live audio was as House Sound Mixer for Pink Floyd in 1972. Since then he has toured in 18 countries and has mixed sound for acts as diverse as Blondie, Sarah Vaughn, Fripp and Eno, and the San Francisco Symphony. Recent jobs include the Playboy Jazz Festival and the live radio mix for the Monterey Jazz Festival, both with McCune Sound, and using the snake system described in this article.*

Designing The System

An essential part of the projected system, the muti-pin connector, was already installed on the Heider truck. TOM SCOTT, Jack's predecessor, had found a 150-pin connector from A-MP, and, having worked out the pin

— *continued overleaf . . .*



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- Assignable voice modes (normal, single, double, alternate)
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- Pitch bend and modulation wheels
- Polyphonic modulation section
- Voice defeat system
- Two assignable & programmable control voltage pedals which can act on each manual independently
- Three-band programmable equalization
- Program increment footswitch
- Programmable volume control and a master volume control
- Octave transposition switches
- Upper & lower manual balance control
- A-440 reference tone

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configuration for 50 pairs, had constructed a snake and stagebox using twenty-seven pair Belden cable. The splitter box design required a 1:1:1 turns-ratio transformer, which at the time was unavailable. DEANE JENSEN, who was at Heider's at the time, wound the prototypes. At the suggestion of Jack Crymes, multiple Faraday shields were included, one for each winding. These dramatically reduced the buzz being picked up from SCR lighting dimmers, a problem all too familiar to Jack and Deane.

The completed system fulfilled all the requirements, and when Jack left Heider/Filmways both the trucks and Filmways Audio were completely equipped with the snake systems.

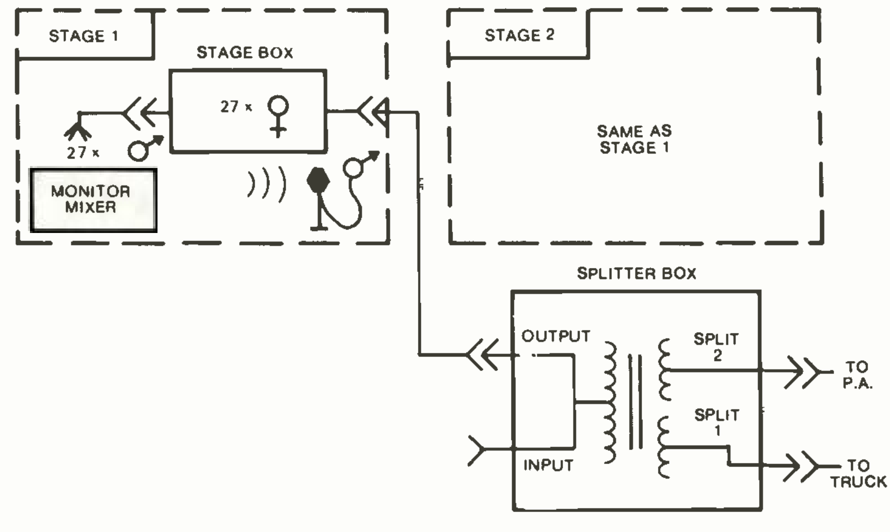
Typical Mobile Recording System

Since 1975 Jack has been chief engineer for the Record Plant Remote Recording Division, which he set up with TERRY STARK, and each of the three trucks (one more is under construction) is equipped as follows:

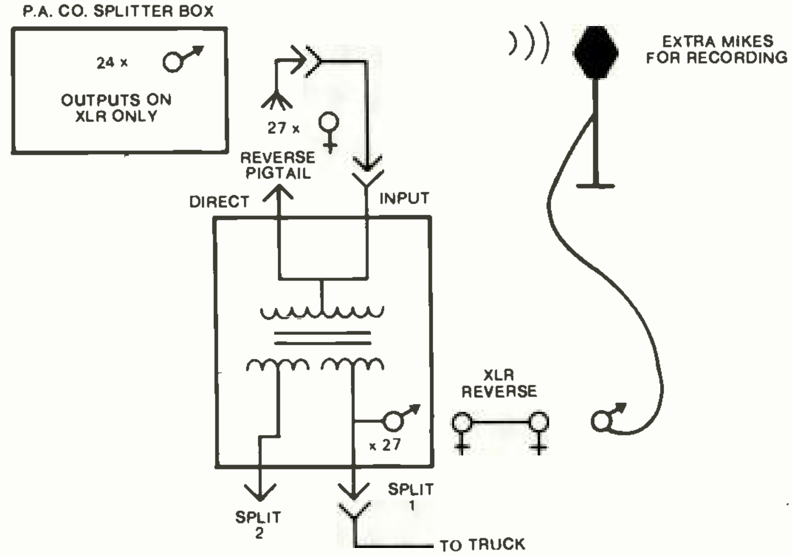
—Four 150 foot 27 pair snakes (Female A-MP connector to Male A-MP connector)
—Two 50 foot 27 pair snakes (Female A-MP connector to Male A-MP connector)
—Two 10 foot 27 pair snakes (Female A-MP connector to Male A-MP connector)
—Two Male and two Female "Pigtails" or "Fan-Outs" (A-MP to 27 XLR connectors on six-foot mic cables)
—Two Stage Boxes (termination boxes) (27 Female XLR connectors to one Male, one Female A-MP)
—Two Splitter Boxes (one Female A-MP, three Male A-MP plus 27 ground lift switches)

All snakes and pigtails have A-MP *plugs* which will mate with plugs or receptacles of the opposite sex. All stageboxes and splitter boxes have A-MP *receptacles* which mate only with plugs of the opposite sex. Snakes can thus be extended with ease and stageboxes may be "daisy-chained" together around the stage. The A-MP plugs can be connected or disconnected in under five seconds, making it simple to use one mixer for several stages, invaluable for festivals or TV shows.

— *continued overleaf . . .*



2 SETUP FOR MULTIPLE STAGES WITH INDIVIDUAL MONITOR MIXERS. NOTE "BACKWARDS" SNAKE FROM STAGE BOX TO SPLITTER BOX.



3 SETUP SHOWING USE OF EXTRA XLR CONNECTORS ON RECORD PLANT SPLITTER BOX.

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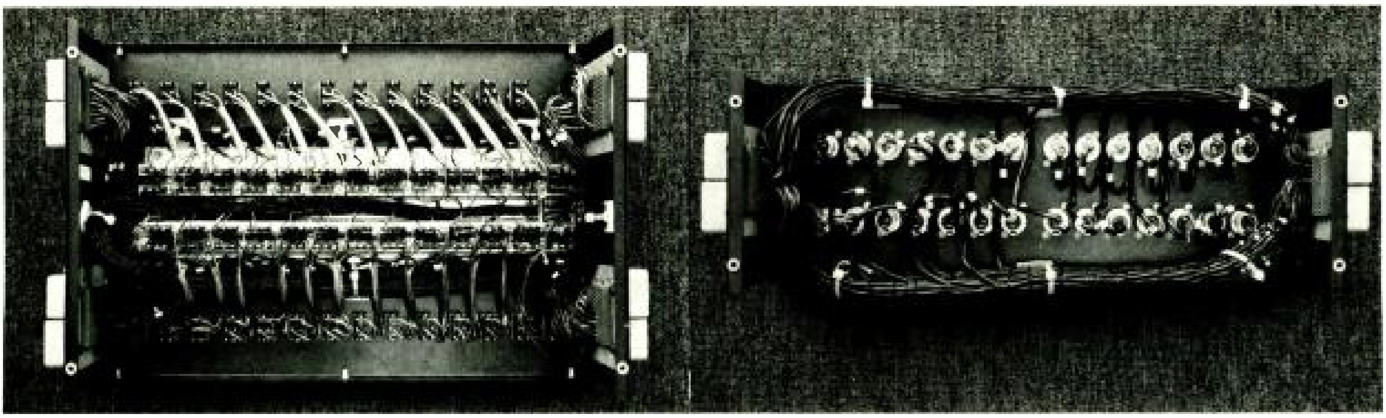

(Left) 27-pair Splitter Box built by McCune Sound. Note ground lift switches and transformer mounting rails. *(Right)* 27-pair Stage Box with "loopthrough" connections. Photographs courtesy of McCune Sound, San Francisco, California

For flexibility Jack has included 27 Female XLRs on the splitter box paralleled to Split I, and pigtails come in both sexes so that feeds may be taken direct from male XLRs rather than by patching to a stagebox with short mike cables. He has also developed a number of tricks as a result of dealing with TV shows, where stage and set changes are made at alarming speed. Some sample setups are shown above.

Standardization

The title of this article implies that the system described is an industry standard, which is not strictly true. But it is a tribute to Jack's design that it has been adopted by so many others. In addition to the Record Plant and Heider/Filmways, the system is used by Stanal Sound, McCune Sound, A-1 Audio, and has been installed in several concert facilities, notably Concord Pavillion. (See *Sound Man's Guide to Venues #11, R-e/p, August 1980*). The A-MP connector was originally chosen with 50-pair snakes in mind and in fact the Doobie Brothers' system uses all 150 pins. McCune uses a 50-pair house and truck feed on large shows by combining the outputs of two complete 27-pair systems through a 2 by 27 to 50 adapter. (The last four lines on the second 27-pair are not used.)

A-MP CONNECTOR WIRING CONVENTION

H Hot; C Common; S Shield

Figures Moulded On Contact Housing Block	Block #1		Block #2		Block #3	
A	H		H		H	
B	C	Pair 1	C	Pair 17	C	Pair 33
C	S		S		S	
D	H		H		H	
E	C	Pair 2	C	Pair 18	C	Pair 34
F	S		S		S	
H	H		H		H	
J	C	Pair 3	C	Pair 19	C	Pair 35
K	S		S		S	
L	H		H		H	
M	C	Pair 4	C	Pair 20	C	Pair 36
N	S		S		S	
P	H		H		H	
R	C	Pair 5	C	Pair 21	C	Pair 37
S	S		S		S	
T	H		H		H	
U	C	Pair 6	C	Pair 22	C	Pair 38
V	S		S		S	
W	H		H		H	
X	C	Pair 7	C	Pair 23	C	Pair 39
Y	S		S		S	
Z	H		H		H	
a	C	Pair 8	C	Pair 24	C	Pair 40
b	S		S		S	
c	H		H		H	
d	C	Pair 9	C	Pair 25	C	Pair 41
e	S		S		S	
f	H		H		H	
h	C	Pair 10	C	Pair 26	C	Pair 42
j	S		S		S	
k	H		H		H	
m	C	Pair 11	C	Pair 27	C	Pair 43
n	S		S		S	
p	H		H		H	
r	C	Pair 12	C	Pair 28	C	Pair 44
s	S		S		S	
t	H		H		H	
u	C	Pair 13	C	Pair 29	C	Pair 45
v	S		S		S	
w	H		H		H	
x	C	Pair 14	C	Pair 30	C	Pair 46
y	S		S		S	
z	H		H		H	
AA	C	Pair 15	C	Pair 31	C	Pair 47
BB	S		S		S	
CC	H		H		H	
DD	C	Pair 16	C	Pair 32	C	Pair 48
EE	S		S		S	
FF	H	Pair 49	S	Pair 49	C	Pair 50
HH	C	Pair 49	H	Pair 50	S	Pair 50

How Much?

A system similar to that shown in Figure 1 represents an investment of close to $2,500.00 in A-MP parts and transformers alone, but the flexibility of the system and ready availability of compatible replacements for any component gives it a long working life. As an example, in 1973 Jethro Tull took delivery of a house sound console which could only accept mike inputs through a single multipin connector. The mating connector on the snake made a poor fit and a week into the "Passion Play" tour the cast-alloy connector housing shattered, putting the entire production in jeopardy until the next day off, when the connector was replaced. By contrast, the last two Playboy Jazz Festivals have been staged on a turntable, and set changes necessitate complete disconnection of the mike lines while the turntable revolves, and rapid reconnection before the first note! A McCune Sound snake/splitter system is used with duplicate stageboxes on both sides of the turntable and, despite as many as 60 disconnects/reconnects per show, not one audio line has been lost.

Similarly a recent Big Band Show at the L.A. Forum featuring three bands playing in rotation was handled with four snakes, three stageboxes, and a pigtail. Disconnects between sets were made at the side of the stage while the M.C. used a "hard-wired" mike to introduce the next act.

Conclusion

Any audio system is only as good as its weakest part, and the expense of the systems detailed above is more than justified by their reliability and the way in which they can be reconfigured for different requirements, a feature often overlooked by newcomers to live audio. For an equipment rental company there are obvious advantages to a system that is compatible with both the client's state-of-the-art custom mixer and that "old tube mixer that sounds great on strings." Renting your back-up mixer to the support act becomes feasible with the addition of a pigtail and stagebox, and you can promise your client trouble-free remote recording.

This year's Monterey Jazz Festival featured split feeds to the house mixer, the Phil Edwards Remote (recording for NPR) and a McCune truck providing mono and

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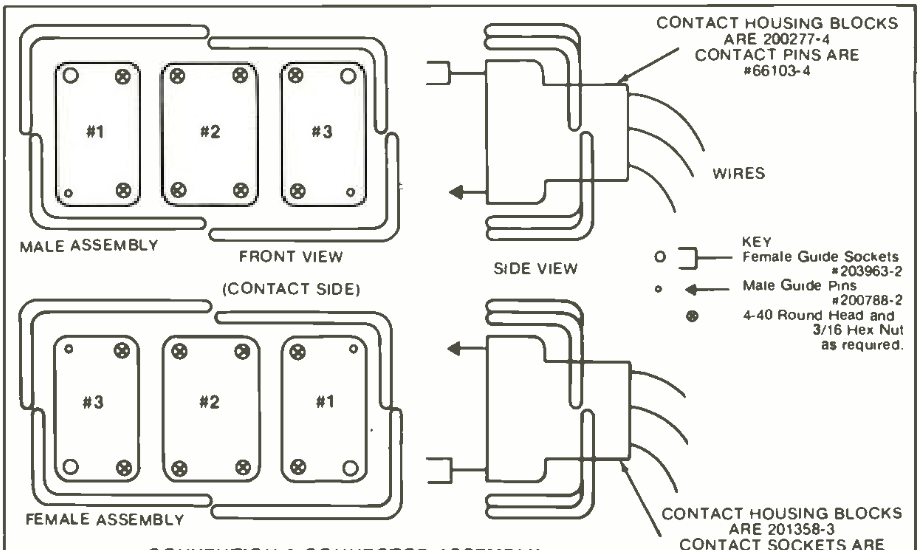

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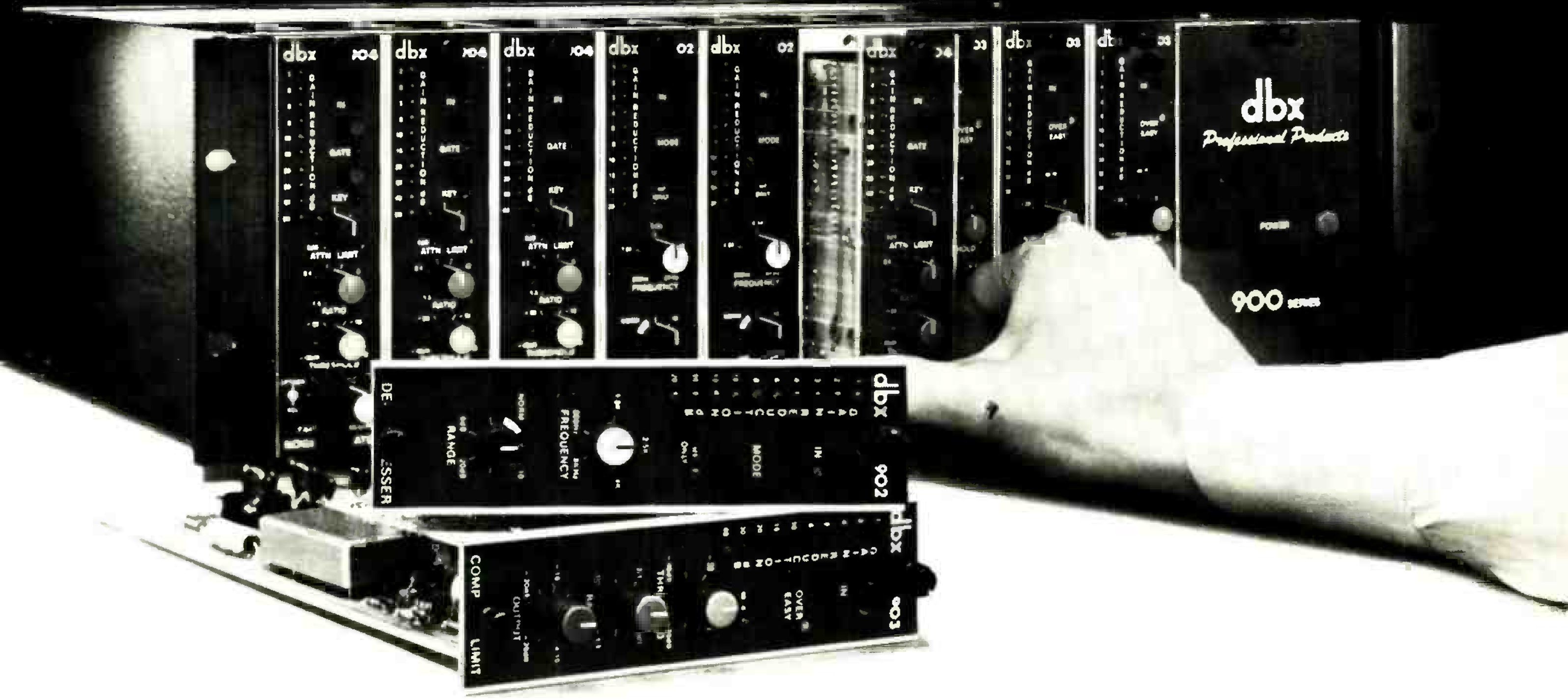

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PARTS LIST

Does not include metalwork or cable. See note on construction.

Part Description	Part Number	Splitter Box Qty	Snake Qty	Stage Box Qty	Pigtail Qty	Reverse Pigtail QTY
Connector Body (Plug)	225004-1	-	2	-	1	1
Panel Mount Connector (Receptacle)	225005-1	4	-	2	-	-
Male Contact Pin Housing Block	200277-4	9	3	3	-	3
Female Contact Socket Housing Block	201358-3	3	3	3	3	-
Male Guide Pin	200788-2	8	4	4	2	2
Female Guide Socket	203963-2	8	4	4	2	2
Strain Relief Bushing	202038-1	-	4	-	1	1
Male Contact Pin	66103-4	243	81	81	-	81
Female Contact Socket	66105-4	81	81	81	81	-
1/2" (half-inch) 4-40 thread round-head machine screws	-	32	16	16	8	8
4-40 thread 1/4" (quarter-inch) Hex Nut	-	16	-	8	-	-
4-40 thread 3/16 inch Hex Nut ("Radio Nut")	-	32	-	16	-	-
Male XLR Connector	-	-	-	-	27	-
Female XLR Connector	-	-	-	-	-	27
Female Panel Mount XLR Connector	-	(27)	-	27	-	-
JE-MB—D Mic Bridging Transformer (or equiv.)	JB-MB-D	27	-	-	-	-

Splitter Box — Accepts mic-level inputs on snake and provides one direct output and two isolated splits to snake or pigtails for connection to mixer inputs.
Stage Box — Accepts mic inputs on XLR connectors and connects to 27-pair snake.
Pigtail (Fan-Out) — Connects snake or splitter box output to XLR inputs (Female).

PRICE LIST: A-MP PARTS
(As of September 26, 1980)

Part Number	Description	Price
225004-1	Connector Body (Plug)	$ 50.70
225005-1	Panel Mount Connector (Receptacle)	$ 7.32
200277-4	Male Contact Pin Housing	$ 2.76
201358-3	Female Contact Socket Housing	$ 2.76
200788-2	Male Guide Pin	$ 1.17
203963-2	Female Guide Socket	$ 1.34
202038-1	Strain Relief Bushing	$ 1.24
66103-4	Male Contact Pin	$ 169.32 per thousand
66105-4	Female Contact Socket	$ 192.00/1000*

Price subject to Gold surcharge.

PRICE LIST: JENSEN TRANSFORMERS

Part No.	Description	Price Schedule 1-19	20-39	40-59	60-79
JE-MB-D	3 Windings 3 Faraday Shields	$52.64	$48.96	$45.53	$42.35
JE-MB-C	2 Windings 2 Faraday Shields	$30.13	$28.03	$26.06	$24.24
JE-MB-E	4 Windings 4 Faraday Shields	$75.18	$69.92	$65.03	$60.48

stereo mixes for backstage monitoring and radio transmission respectively. Despite the apparent complexity of the system (and the state of some of the players' amplifiers), buzz and hum were marked by their absence throughout.

Notes On Construction

The A-MP Connector assembly is described in Information Sheet IS 2118, available from the manufacturer. The 4-40 thread HEX nuts in the accompanying parts list are substitutes for identical A-MP parts. The 4-40 round head machine screws are also substitutes for A-MP guide pins, as it was felt that four guide pins per connector were sufficient (see Convention and Connector Assembly). When connecting to Belden 8773 two Strain Relief Bushings are used, one slit lengthwise and enclosing the other. (A Strain Relief Bushing for cables of diameter 0.84" to 0.94" is A-MP Part #2020371.)

Everyone I have spoken to agrees that the A-MP Connector is very easy to work with and should take two to four hours to wire and assemble. A high quality crimping tool and a pin-extraction tool are essential. (Both are available from A-MP.)

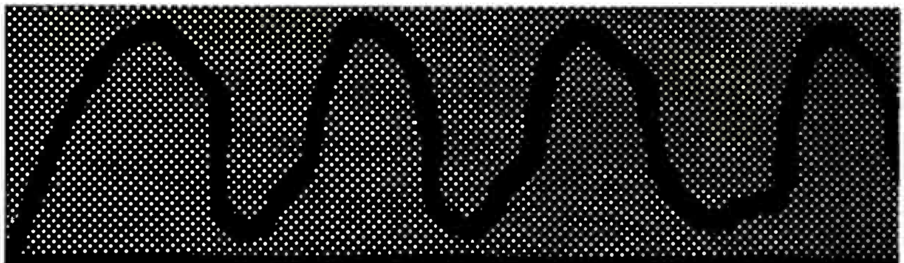
The Splitter Box is the most time-consuming (and expensive) item to build. Metalwork should be designed to protect A-MP receptacles and ground-lift switches. Transformers must be mounted securely. Non-locking Female Panel Mount XLR connectors might be considered for Stageboxes, as some lock-release tags bend and break easily. On the other hand, one of the major TV networks insists on locking connectors for audio lines.

Cable manufacturers are highly competitive and comparisons are outside the scope of this article, but the author's experience with Mohawk W&C #22 twenty-seven pair leads him to recommend it for handling and weight, important considerations when dealing with 150-foot lengths. Neumann mike cable (three conductors and a shield) is also recommended for pigtails, and lengths of at least six feet should be used so that split mixing consoles are easy to reach. Sensible use of Panduit makes a potential rat's nest into a manageable and useful tool.

Ground lifts are provided on the splitter box, but each user has developed a different philosophy. Filmways/Audio has 81 miniature toggle switches on each box, one for each mike line on each output. Jack Crymes favors ground lifts on Split 1 only, leaving Split 2 grounded through its destination, the PA or monitor mixer. McCune has only one common ground line, normally number one, so that input must be used on all mixers.

An important safeguard implemented by BOB CAVIN at McCune is a one-amp fuse across the ground line to prevent damage to the snake and mixer in the event of, say, a guitar amp shorting to the PA system. Another modification is to provide bypass circuitry for one of the transformers, so that a line-level signal may be returned to the stage on the house snake. This is not recommended practice, but could prove invaluable in an emergency.

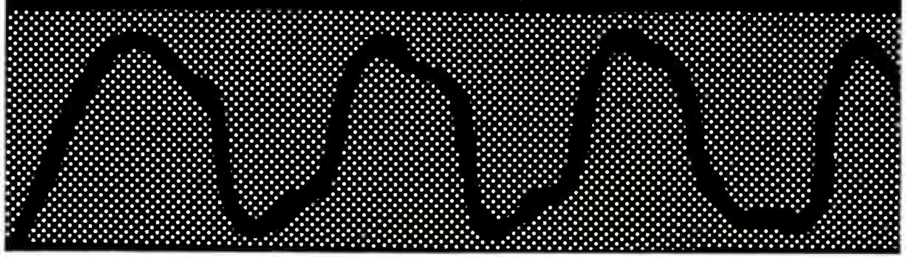
Metalwork will probably vary with the application, so drawings have not been included. Any questions should be directed to the author, c/o *Recording Engineer/Producer*. Further input on the subject of mike splitter snakes is most welcome. □□□



. . . the psychoacoustics of wave form complexity —

the LIMITING factor
or
factors of LIMITING

by Paul C. Buff



(and, from those friendly folk who brought you Kepex and Gain Brain . . . a new product)

Audio compressors and limiters have been with us for some time. There are as many philosophies of design of this sort of equipment as there are philosophies of how they should be used . . . to what avail.

My purpose is to conduct an overview of how and why these varied design and use categories have come to be, as well as to try to define a set of parameters which best suit the music makers' moods for the '80s.

The key word here, perhaps, is my choice of the word "moods," for this is what determines the rules of subjectivity which will be applied to this sort of equipment (or other sorts for that matter), at the specific point in time in question.

Now these statements may be met with a barrage of rebuttal from the purists, who will claim that the purpose of audio equipment, always, is to capture the recorded program in a form as nearly identical to the original performance as possible. While I will not disagree that equipment should be capable of doing this, when called upon, it is obvious that the wants and desires of commercial producers, engineers, and the buying public often depart from these goals. They generally want a sound which constitutes a departure, in some form, from that which was originally performed in the studio. Hopefully, this departure offers something in the way of an "improvement" over the original.

As with most things associated with human desire, what we perceive as in "improvement," in the area of recorded sound, is subjective, and changes periodically in a general cyclic pattern. Thus, the equipment designer cannot speak in terms of absolutes, rather, he is required to speak in terms of desirability for the current ear of users.

Early Limiters and Compressors

In the decades preceding the mid-fifties (a generalization) the recording industry was primarily engaged in the activity of "capturing" the performance, with the highest level of "fidelity" obtainable. It was perceived that noise, from whatever source it came, was a deterrent to those objectives. So, on the other hand, was distortion. As today, the obvious method of dealing with these degrading elements was for the engineer to carefully place the signal at a level just below the saturation point of the processing equipment, for optimum noise performance. However, woe be to the engineer who failed to keep the signal below the saturation point, lest the dreaded distortion set in.

Now, in these days, the onset of distortion often came about rapidly and with catastrophic results . . . overcutting the disc, clipping the optical track, over-modulating the transmitter. These forms of overload are associated with *peak* excursions of the waveform, thus the *peak limiter* came about. The principle here was to electrically "look" at the waveform peak excursion and, through an electronic gain control mechanism, to control the gain of the equipment in such a fashion as to assure that the output signal never rose above that point which would result in clipping.

With this "overload insurance" the engineer could safely strive for program levels consistently above the noise floor, with the assurance that any *unexpected* high level excursions would be taken care of by the peak limiter. Obviously, there remain today, many applications for such equipment, as we are still cutting records and transmitting material via radio and TV.

However, it was, and is, apparent that the use of such peak limiters, if the goal of signal purity is to be accomplished, must be sparing, lest a whole new set of undesirables come about . . . the sound of limiting . . . pumping and squashing. Why do these effects occur in a peak limiter? Universally, they come from the attack and release

— continued overleaf . . .

structures, which we shall review independently.

In looking at the attack structure, it is evident that most uses of such limiters require a rather rapid assessment of a potential overload condition, and reaction quickly enough to prevent the undesired over-excurison. Now, it is the nature of peak detection to do just what the name implies . . . to detect peaks . . . electrical waveform peaks. Electrical waveform peaks have little to do with audible peaks, due to the complex nature of music and speech waveforms. This may be made more clear by use of example . . . comparing a sine wave to a complex music or speech waveform:

Let us analyze a 0 dBm level sine wave. It has a power content of 1 milliwatt (which is how we judge its audible loudness), while exhibiting an electrical peak value of 1.095 volts peak. If we were to set up a peak limiter for a 1.095 Volt limiting threshold, a sine wave, when inputted to the limiter, would come out at a maximum level of 1.095 volts peak, and would produce 1 milliwatt of audible power into a 600-ohm load. It can be said that this happens because the sine wave has a peak-to-RMS ratio of 3 dB.

Now let us input a typical complex music waveform, say, from a saxophone. It, too, will exit the limiter at a maximum level of 1.095 volts peak. However, such a waveform does not exhibit a peak-to-RMS ratio of 3 dB, rather, the ratio is much higher and is a function of the specific instrument which produced it. For a saxophone, a more typical ratio might be 9 dB. If this were the case, even though the limiter would output the sax waveform at the same peak level as it outputted the sine wave (1.095 volts peak), the sax signal would only produce about ¼-milliwatt, thus *sounding* some 6 dB softer than the sine waves 1 milliwatt.

The net result of this behavior, on steady state signals, boils down to a rather simple observation: The audible loudness of various signals passed through a peak limiter is inversely proportional to the complexity (or peak-to-RMS radio) of the waveforms involved. Stated another way, the more complex the waveform, the lower the apparent output loudness. Thus, high complexity signals such as brass and reeds are discriminated against, while simple waveforms such as flutes and falsetto voices are accentuated. More on this subject later.

So far, we have discussed only steady state signals . . . which rarely exist in music. Much of a musical program is made up of waveforms which exhibit an abrupt attack to a high level, then a quick decay to a lower, more sustained level. Struck instruments fall into this category. Often, the attack of such waveforms contains little audible energy, and serves primarily to identify the waveform as being transient. This waveform topography tends to make the effective complexity even greater than in the steady state condition. When passed through the fast attack peak limiter, such waveforms are sensed as "very powerful" and cause an inordinately high amount of gain reduction . . . thus serious discrimination . . . a squashing of transients.

Now, since the original purpose of the peak limiter was to protect the signal chain from either instantaneous or sustained over-excursions of the electrical waveform, little could be done about these discriminations, other than sparing using to minimize the effects. Otherwise, the very purpose of the equipment would be undermined.

Once the limiter had reduced its gain, in response to an over-excursion, there remained the question of how rapidly the gain would recover, or release, once the over-excursion had ceased. If the release time were made long, gapping holes would appear in the program following an attack, while the overall output program level would be lower than desired. On the other hand, the use of fast release times led to the gain actually being modulated by lower frequencies, thus giving rise to serious distortion products, as well as to audible pumping and a higher tendency to produce very squashed transients. Again, sparing use was the only real answer.

THE SOUND OF THE SIXTIES

Sheer Volume

The sound of the sixties could probably best be described as the sound of maximum volume. It was discovered by "hip" producers that sales were directly proportional to volume. If the record jumped out on the airwaves and juke boxes at a level higher than that of the competition, it sold better. So every possible dB was put on the record, with a damn the torpedoes view of the pumping, squashing and distorting which was caused in the process. We had left the era of high fidelity and entered the era of purposeful level modification. The fast attack, fast release peak limiter was ideally suited for the purpose, as it offered the maximum possible volume, within the confines of keeping the needle in the groove (sometimes) and keeping the FCC from shutting down the station for overmodulation.

During the same era, it was discovered that by limiting the components of the mix individually, still higher average levels could be obtained. Thus, the term "leveling amplifier" was born. The voice was "leveled" so that it could be placed a scant fraction of a dB above the music, and still be intelligible (sometimes). The solo instruments were "leveled" for the same reasons. Even the rhythm instruments were "leveled" so that they would not get too loud, and upset the whole critical formula. Thus, the sterilized formula sound of the sixties was merely layer-after-layer of signal sources whose dynamic range had been reduced to as near zero as the equipment would permit.

THE SEVENTIES

Effects

Slowly, it began to sink in that this "leveling" process was not really working. Transient instruments, when limited, actually appeared at a lower audible level than if they were left alone. This was primarily due to the fast attack circuitry which shoved down the gain every time a sharp transient appeared. So slower attack and adjustable attack peak limiters came into vogue. Even though they did a lesser job of controlling the absolute peak output level (since fast transients escaped the threshold of limiting), they did a better job of unifying the apparent levels. By now, also, the

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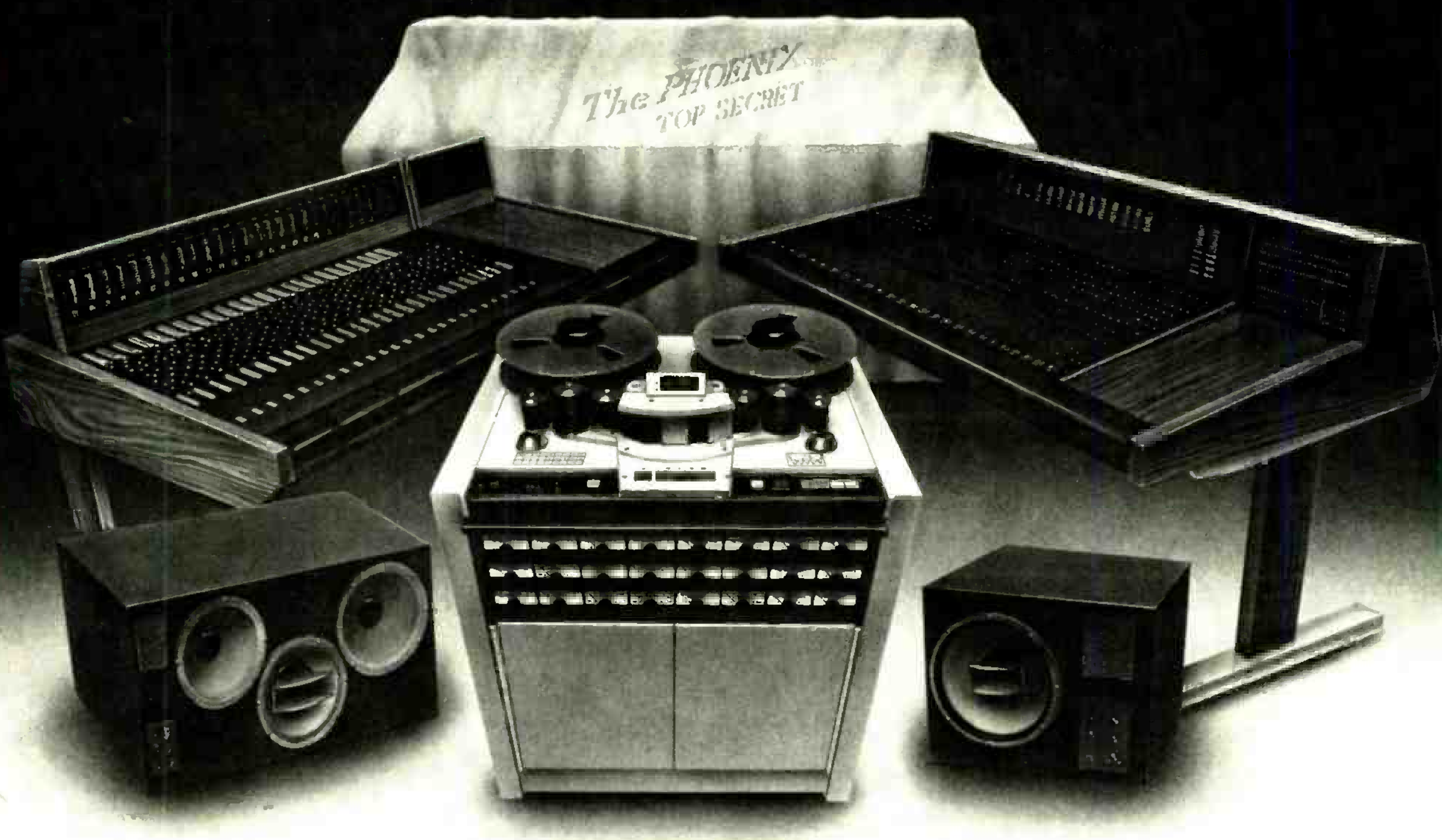

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... the psychoacoustics of wave form complexity —

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requirements for absolute control of peak excursions had laxed, since most limiters fed tape tracks, rather than disc chains and transmitters. The primary purpose was no longer to prevent overload distortion, rather, it was purely to control the aesthetics of the mix. The dependence on sheer volume subsided, and gave way to the creative processing and manipulation of signals. Transients were back, and music began to come back to life.

If one were to try to put a handle on the "Sound of the Seventies," it would have to be the sound of processing and manipulation. Just as the groups on stage would do absolutely anything to get attention, so would the producer in the studio. It was an age of gadgetry. In both camps, there was a near total disregard for the cost of achieving the insane . . . the end justified the means.

THE PROFESSIONAL EIGHTIES

And what of the eighties? This one, of course, will involve crystal ball gazing. In my globe, I see a return of sanity to the recording industry. From an economic point, this will be necessitated by the unavailability of funds for non-productive gadgetry and $200 per hour fees for using them. Producers will discover that saleable product can be made in less than 200 hours of studio time, and on less than 46 tracks. They will also discover that sonic manipulation for the sake of oddity will no longer make their product stand out . . . for the world has heard it all, in a manner of speaking. I, for one, would just as soon bank my money on a Joan Baez and her guitar, cut in clean stereo (two tracks) in 20 minutes, than I would on another super group whose magnificent performance takes three months of time in a "world class studio." Now, I am not suggesting that the industry will go back to mono, or that all progress will cease. Surely, the "world class studio" will exist in the future, in greater glory than we know today. It will not, however, exist on every street corner, and be the backbone of the recording industry. As far as the mainstay studio of the eighties, I see a much less elaborate approach. More real signal quality, fewer "lights and whistles," greater functionality in a smaller package, a "living room mixing approach" . . . you name it. I also see, as entrepreneurs of these establishments, a much more enlightened group of people . . . people who understand what is happening in their studio, both electronically and musically, and who know how to get the best out of it. These same people, armed with this knowledge, will dictate to the manufacturers that they receive good value for their dollars. They will not accept equipment which fails to keep pace with the level of performance allowed by the current electronics technology. They will not be turned by hype level promotion and gadgetry. These people will act like, no, will be, professionals themselves, for if they are not, they shall not survive.

And from what ranks shall this new breed come? Today, they are called "semi-pro" . . . people not able to afford to spend $800,000 on a studio. But what if they were to spend, say, $100,000 on a studio and put their heart and soul into running it? Would they still be "semi-pros?" From where I sit, a person who spends this kind of money is not in a "semi" position . . . he is serious . . . he's either a pro or a fool, and there's not many fools running around with a hundred grand. (Or $40,000, if you like that number better.) A "MacDonald's" studio run by hype artists, regardless of the investment, will not show very well against serious proprietors with a lessor investment but greater abilities. Certainly not in this sort of economic climate and in an artistic medium.

To sum up the Sound of the Eighties, as I see it, we will have come around full circle, to a certain degree. An emphasis on simple sonic beauty and quality. Will it be pure and un-modified . . . as performed? Doubtfully. The effects will still be there. The difference will be that records will not be three minutes of effects plus a little music, it will be the other way around. The effects will enhance the performance, not fight with it.

Limiter/Compressors Redefined

O.K., what requirements will these objectives place on the equipment of the eighties, or more specifically, on the limiter/compressors, since that is what this article is supposed to deal with? We must start by realizing that we have two conflicting objectives for this sort of equipment. This first objective is the same as it was back in the fifties . . . prevent program peaks from overloading successive stages such as cutterheads and transmitters. Here, the requirements have not changed. We still must look at electrical peaks, as these are what can cause the overloads. We must still use these devices sparingly, for the same reasons given earlier, although we can gain some measure of improvement through preview methods and computational circuitry, in order to minimize the undesirable side effects.

This sort of limiter usage constitutes a very small percentage of what limiters are used for. In fact, the average recording studio, unless it is engaged in disc work, has little need for such equipment. However, what the studio sends out for mastering will have great bearing on how unobtrusively the peak limiter can do its job of getting the material on record and on the air.

In the recording studio, we are dealing with a whole new set of rules. We are not limiting or compressing in order to prevent overload. Our equipment has sufficient headroom, and soft clipping characteristics, particularly if we have noise reduction. We need not strive for needle bending levels, on individual tracks of the multitrack thanks, in part again to noise reduction, and in part to post noise clean-up devices such as KEPEX,® and KEPEX II,® and automation.

Then just what do we use limiter/compressors for in the studio? Effects. Manipulating the levels of the individual tracks and instruments, as indicated by what we desire to hear. Sometimes we want to "level" the sound so that it can be placed in a particular region of loudness, and so that it will be dynamic to whatever degree we desire.

At other times, we might desire to actually modify the envelope of a program source . . . diminish the percussiveness . . . or accentuate the percussiveness. We might want interactive dynamics control . . . where the envelope of one instrument modifies the envelope of another. For instance, every time the singer sings, we might wish to slightly lower the level of the lead guitar track . . . or wipe it out completely. Who knows what we might want to use a limiter for. That decision is the job of the creative engineer/producer.

In all the potential uses for this sort of equipment, one factor becomes quite clear: In a recording studio production limiter/compressor, we are not concerned with the electrical excursions of the waveform . . . we are concerned with *audibility factors*. The precepts of peak limiting are fully at odds with these objectives, and cannot serve the user very well in obtaining the results he might desire.

For a production limiter, we want a device that "hears" what we hear, and reacts as we would react.

Detection Methods

At what level, and to what degree, a limiter/compressor reacts to input signals is a function of the type of detection employed. We have already discussed peak detection, and ruled it out as being very discriminatory and not representative of how we hear.

How do we hear? Very simple . . . we hear power levels, not voltage levels. And by what mechanism can we detect power levels, in voltage processing equipment? True RMS detection, of course. For the true RMS value of a voltage waveform is a direct measure of its real energy potential. Well, then, that ought to settle it. If we use true RMS detection, our limiter will hear exactly what we hear, and will perform flawlessly. Right? *Wrong!*

We must delve a bit further into the world of hearing and waveform complexity, if we are to make any real sense of what we are trying to achieve. First, we should look at the nature of various waveforms of the type we wish to process. Why are certain waveforms more complex than others, from a music and speech standpoint? One particular pattern of general waveform behavior should give us a clue in answering this question: In general, when an instrument or voice wants to be heard above the ambience surrounding it . . . to stand out and be noticed, it tends to produce a more complex waveform than when it desires to blend in with the ambience. A scream, or an accentuated vocal line, is always more complex in waveform than is normal conversation. Solo lines, on almost any instrument, produce higher complexity than do background lines. This even applies to the instruments themselves. Those instruments which are designed to predominate, almost invariably produce higher peak-to-RMS ratios (complexity), than do those instruments which are primarily designed to provide blending background foundations.

Now, in addition to producing a higher waveform complexity when it wants to predominate, any signal source tends to increase in absolute power amplitude, at times dictating its predominance. Thus, we are given a "loudness code" which carries information relating to the desired power level of a waveform, this code being contained in the degree of waveform complexity itself. As was stated earlier, the conventional peak limiter makes a complete mess of this inherent waveform coding . . . when a signal increases its waveform complexity, saying, "I want to be heard, make me louder," the limiter does the exact opposite and makes it softer. As one might guess, this action literally destroys the intent of the instrument and performer, and

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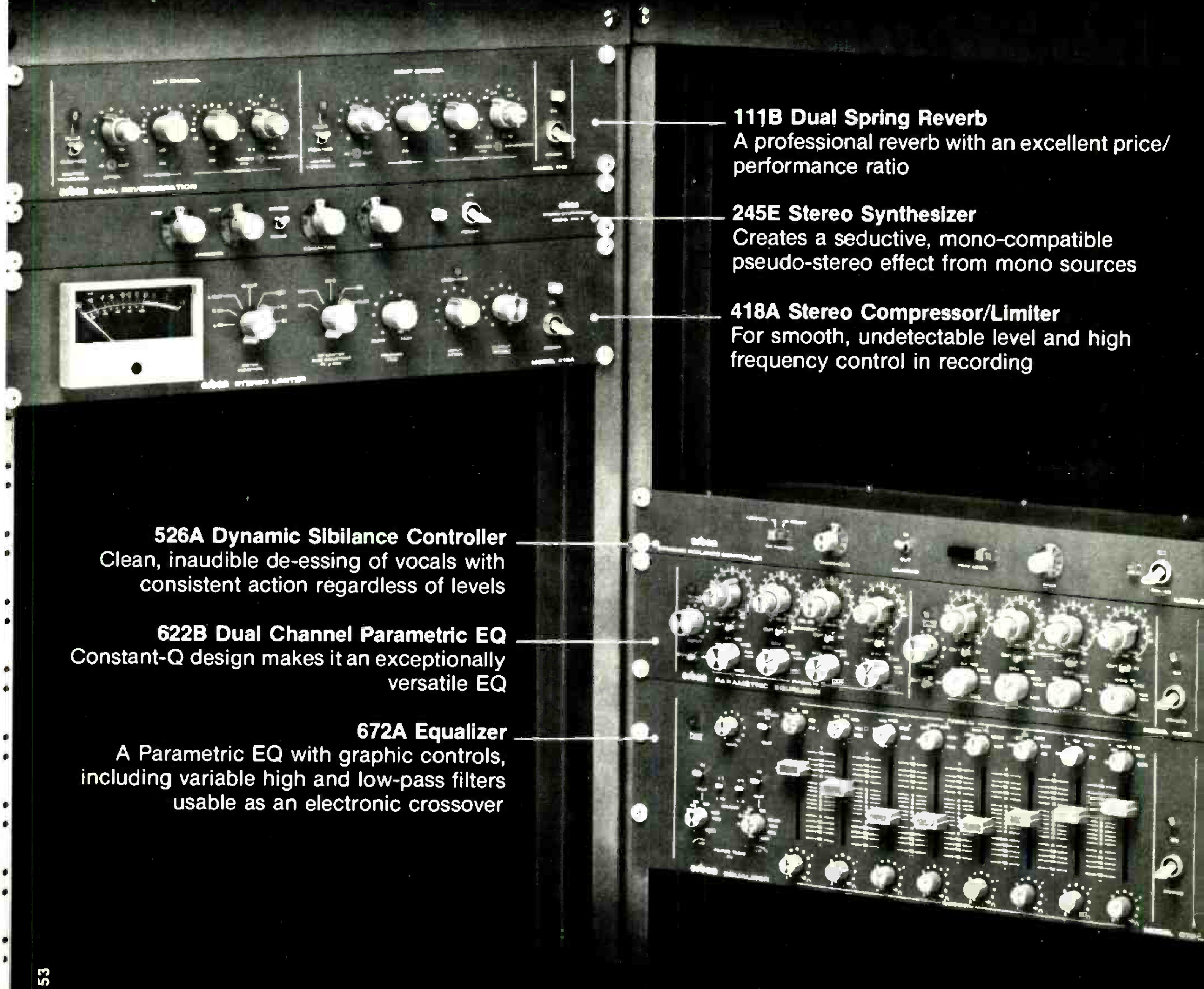

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renders the program very unnatural.

If an RMS detecting limiter is employed, the situation is considerably better. Instead of doing the reverse of what the waveform code requests, the RMS limiter simply ignores the request entirely, making all waveforms come out at the same power level, regardless of what is requested in the waveform's "loudness code" . . . well, almost. In actuality, the RMS limiter does a little discrimination of its own, in that it fails to cope with the inherent high frequency roll off which is a part of the normal hearing process. When a solo instrument or voice produces complex waveforms, the natural result is the production of high order harmonics . . . for these harmonics are, in fact, the very mechanism which contributes to the complexity. Often, the frequency range of these harmonics extends into the region where the human ear suffers a sensitivity loss. Thus, the ear does not hear these overtones at as high a loudness level as is indicated by their measurement on an RMS basis. Thus, the RMS detector "over-reads" the effective power content, thereby reducing the gain more than what a human would have done, based on his perception of loudness. This sort of problem could be taken care of by de-emphasis in the detection loop, to give the RMS detector the same approximate frequency response as the human ear.

Even if the RMS detector were made to hear exactly like the ear does, there is still the situation of ignorance to the coded requests for certain complex waveforms to be outputted at a somewhat higher level, in order to convey the intent of the performer.

Now, of course, we can do an excellent job of obeying these requests by the simple expedient of unplugging the limiter entirely. Indeed, this is done with regularity by those engineer/producers who won't stand for this sort of waveform discrimination.

But, on the other hand, we bought the limiter in the first place, in order to be able to exercise some degree of control over the dynamics of the program material.

Ideally, what we would like to see is a detection method which allows that desired control, yet listens to the requests made by the waveforms themselves for dominance or sub-dominance.

After doing considerable work in this direction, using exotic detection schemes such as "SMR" (reverse RMS), and variable exponent averaging, the author has concluded that a very simple detection scheme, specifically "linear integration" does an excellent job of fitting the requirements. What linear integration does is simply to slightly under-read the RMS value of a given waveform, to an extent proportional to the complexity of the waveform itself. Thus, if the performer "codes" a particular note, or passage, by giving it an increased waveform complexity, the linear integration detector allows that note or passage to go through at a slightly higher level than it would have, had the waveform been more pure (less complex). Thus, the intent of the performer has been recognized, rather than being ignored (as in the RMS limiter), or reversed (as in the peak limiter). The degree of this purposeful reading error is not severe . . . on the order of 2 or 3 dB on most music and speech waveforms, yet is enough to give that "edge" or "life" to the processed signals. Thus, the operator is allowed to take advantage of the dynamics modifying advantages of limiter/compressor equipment, within a framework of maintaining program integrity.

Attack Times

Next, we must address the question of "attack time" or, more correctly, in the case of either integrating or RMS detectors, "integration time." In order to perform an integrating operation (which is also a necessary step in the detection of RMS values) the signal must be measured over some period of time. The time required to perform this operation is a function of the lowest frequency whose average value is to be determined. In order to achieve an accurate average value, this integration time needs to extend to perhaps three times the cyclic rate of this specified lowest frequency. If we were to assume that we desired to obtain the integrated value of, say, a 20 Hz waveform (whose cyclic period is 50 msec.) we would have to employ an integration time on the order of 150 msec. Yet, if we were to feed typical transient waveforms into such a slow responding circuit, these fast rise transients would escape detection entirely, as the sound would come and go before the integrater had a chance to respond.

Thus, a limiter fitted with such a long integration time, while doing a splendid job on 20 Hz material, would fail to be a limiter at all, with respect to transient material. When applied to steady state material, such as organ notes, limiting would occur, but only after a fairly long (and audible) period of serious overshoot.

If, on the other hand, we opted for a much shorter integration time, say 500 usec., the limiter would capture the transients respectably, and would accept the onset of steady state signals without audible overshoot. Such a limiter, however, would act as an integrating limiter only for input signals whose cyclic period was shorter than the stated 500 usec. integration time (i.e., frequencies above 3 or 4 kHz). For frequencies below this point, the limiter would simply revert to becoming a peak limiter, together with all the disadvantages of same. Additionally, the use of such a fast integration time would result in the over-detection of certain highly transient signals, such as drums, since much of the energy on these signals is often contained in the first millisecond or so, yet this energy has a low factor of audibility. Overly fast attack times, when applied to such fast rise signals, causes the same undesirable effects experienced with the fast attack peak limiter . . . namely the squashing of transients and lower than desired output levels.

Unfortunately, there is no single integration time, or attack time, which is optimum for all possible varieties of signal processing. While integration times on the order of 5 milliseconds appear to be a good median, there is much to be gained by the inclusion of variable integration time mechanisms in limiting equipment. These benefits become immediately apparent when one processes highly transient material such as percussion instruments. As a variable integration time control is tuned through the range of from "very fast" to "very slow," the degree of impact intensity increases dramatically, as does the apparent level of the overall instrument. At the fast settings, the program sounds very much like it has been limited . . . the impact transients are squashed, while the after-ring of the instrument is accentuated. It does not at all sound like what you hear in front of the microphone.

As the integration time is increased, the sound of the instrument begins to open up . . . to sound more like what you hear in the studio. Still, you do have control over the level. This point is perhaps the optimum point of attack time for processing that

Representative Music Waveforms	Limiter Output	Peak Limiter "A"	RMS Limiter "B"	GAIN BRAIN II "C"
	VU Reading	0VU	0VU	0VU
	RMS Power	0dB	0dB	0dB
Median Complexity Music Wave	Comments	Reference Waveform, vocal "a"		
	VU Reading	-8VU	-2VU	0VU
	RMS Power	-6dB	0dB	+2dB
High Complexity Music Waveform	Comments	ie reeds, brass, vocal "rrr"		
	VU Reading	+5VU	+2VU	0VU
	RMS Power	+3dB	0dB	-2dB
Low Complexity Music Waveform	Comments	ie flute, harp, falsetto voice		
	VU Reading	+2VU	-8VU	+1VU
	RMS Power	+1dB	-7dB	0dB
Typical Bass Instrument Waveform	Comments	ie Fender Bass, etc.		

Figure 1 — Representative illustrations of various steady state music waveforms, and the expected output levels from these waveforms after limiting by:
"A" — a conventional Peak Detection Limiter. ***"B"*** — a conventional True RMS Limiter without Peak Reversion Correction. ***"C"*** — Gain Brain II (Linear Integration Detection and Peak Reversion Correction).
Notice, strong discrimination against complex waveforms in ***"A"***. In ***"B"***, at mid and high frequencies the RMS power remains constant, while the VU meter response varies. At low frequencies, strong discrimination results. In ***"C"***, the VU meter response remains constant, except for a slight rise at low frequencies of medium complexity. The RMS power varies about ±2 dB with various waveforms, in a musically correct fashion.

particular instrument, if you are interested in preserving the natural sound of the instrument. But what happens if you make the integration time still slower than this optimum signal preservation point? Through controlled overshoot of the limiter, the audibility of the ***impact signal*** begins to become accentuated, with respect to the after-ring of the instrument. The limiter begins to sound not like a limiter at all, but like an expander. Still, you are limiting, thus have control over the absolute level. If you want "punch" from your percussion tracks, this is the way to get it. These sorts of effects can only be gotten with a limiter which has a variable, or selectable, attack or integration time.

Now, let us return for a moment to the earlier statement that for frequencies whose cyclic period is slower than the integration time, the detector becomes effectively a peak detector. What are the implications of this phenomenon, and how can we deal with it? Let us setup a typical working example to see what occurs: First, though, let us gather a bit of background material to give us some information relating to the expected range of waveform complexities to be found in the various frequency ranges, with typical program sources.

Generally speaking, most music or speech waveforms will exhibit a peak value which is between 5 and 15 dB higher than the average value of the waveform. Advantageously, it is generally found that the majority of waveforms normally found in conventional program material will show a peak/average ratio quite near 10 dB. Toward the lower part of the frequency spectrum, waveforms tend to become less complex, yet somewhat higher in absolute level. The general increase in absolute level is due to the fact that these waves are produced by humans . . . creatures whose hearing response to low frequencies is diminished. The ramifications here are that we can expect the lower frequencies to be simpler . . . having peak/average ratios more like 7 or 8 dB as a median, yet we must recognize that, due to our hearing deficiencies, these lower frequencies must be passed at a somewhat higher absolute level in order to sound proper.

Now, armed with these scraps of information, let us setup our hypothetical limiter, using a linear integration detector (or true RMS detector for that matter), set for a 5 msec. integration time.

Peak Reversion Effect

From previous statements, we can expect this limiter to do a good job on frequencies above 500 Hz or so, reacting relative to the integrated value of the input waveforms. For frequencies below this figure, the device will begin to cease measuring the integrated value of the waveforms, and will begin to revert to the measurement of the peak value of the waveforms. Let's say we calibrate the unit to put out a 0 VU level when fed a large (above limiting threshold) sine wave signal at 1 kHz. Remember, the VU meter itself is an average responding device. Now, if we put in music whose frequency range is above 500 Hz or so, we will find a nice even VU meter reading of 0 VU, and a nice level sound, with some audible emphasis on the more complex waveforms, as desired. Now, let us add a bass line, in the 50 to 100 Hz region. Since these frequencies are below the detector's ability to integrate, the limiter will respond according to the peak value of the bass line. Since we had stated an anticipated peak/average ratio from this sort of signal source of around 7 to 8 dB, the limiter will over-read by this much, and the VU meter output will drop to -7 to -8 VU. The device has seriously discriminated against the very sort of program material which our ears say should be louder than anything else. Additionally, if we were feeding our hypothetical limiter from a full mix of music, it is highly likely that the bass line will have a somewhat higher input level to the limiter than do the other frequencies. This is the way it would have been mixed for a pleasing tonal balance. So what happens? In spite of all our efforts to produce a pleasant sounding limiter, we find that in fact, all of the limiting is occurring from the bass line, and the program is pumping up and down with each note of the bass. Not at all what we had intended . . . not very effective.

Since we have previously shown cause as to why we cannot increase the integration time to a length suitable to perform waveform integration of the bass frequencies, we have but one alternative if we are to obtain our desired goals. That alternative is to second guess the expected waveform complexity of frequencies below the detector's integrating abilities.

At the point where the detector begins to deviate from accurate detection of the integrated value, we must begin to modify the input to the detector such that the errors resulting from the reversion to peak detection are taken care of, and the bass frequencies maintain the same effective limiting threshold (or a couple of dB higher) than do the higher frequencies. From our analysis of the predicted behavior of these low frequency waveforms, it is seen that a correction which ultimately ends up around 9 dB will do the job very nicely. For the average anticipated low frequency material, this will allow for an effective threshold some 1 to 2 dB higher than for the higher frequencies, to compensate for hearing losses at the low end. Should a more complex low frequency waveform show up, the relative thresholds will become more equal. This correlates well, since the more complex lower frequencies have, of necessity, harmonics outside of the very low frequency range, and these harmonics add to the audibility. Should an exceedingly simple bass line show up, the output level will rise to perhaps 3 dB relative to the higher frequencies. This also correlates well, since the presence of very simple bass frequencies spells out few harmonics and a lower audibility factor. Thus, the ear naturally wants these pure tone low frequencies to appear at a somewhat higher absolute level, so that they can be heard.

Having taken care of the detector's reversion to peak response, one small problem remains. We stated that a variable integration time was a must, for optimum performance and effect. If we are to have a variable integration time, the peak reversion point also become variable. This dictates that the correction network which compensates for this reversion must also be variable, and must effectively "track" the integration time control. This whole situation can be taken

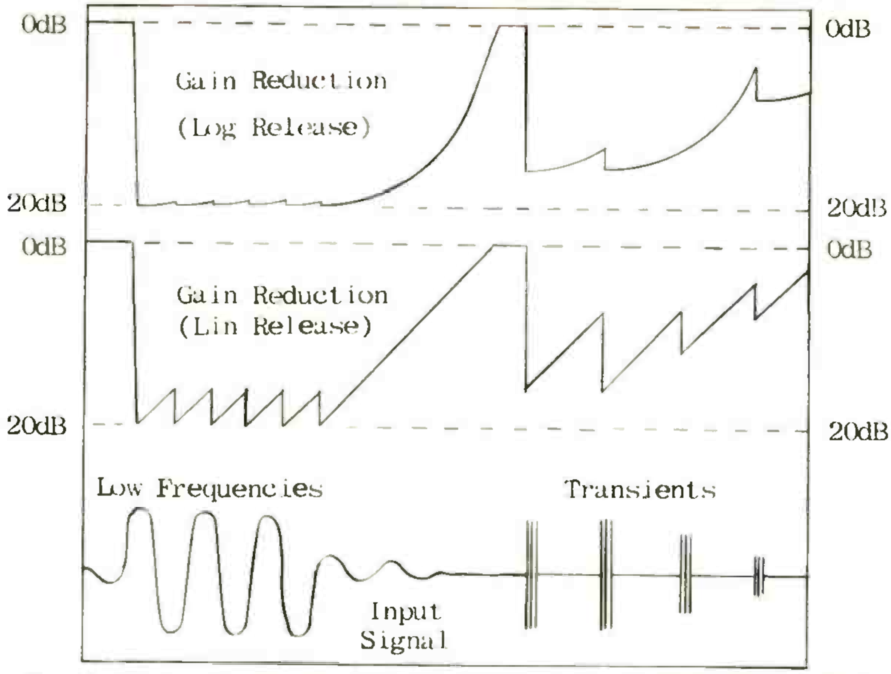


Figure 2 — Pictographic representation of Log Response Shape vs. conventional Linear Release. Note that, although the overall release rates are similar, the Log shape produces significantly less gain modulation on low frequency material (lower distortion), and on recurring transients (less pumping and envelope distortion).

care of with a suitably configured tracking anti-reversion network, ganged to the integration time control.

Release Time

As with the attack time of a limiter, there is no optimum release time which will suit all sorts of program material. To restate the reasons; the use of a long release time, while preventing low frequency modulation effects, tends to lead to excessive recovery times, and holes in the continuity of the material. On the other hand, fast release times encourage severe intermodulation distortion, as well as to add to the squashed, limited sound in general. Many schemes have been put forth as mechanisms to counter these problems. Among them are circuits which sense limiting from a low frequency signal source and to increase the release time when this happens. While this methodology can reduce the distortion effects, it tends to even accentuate the audible program holes, following a low frequency attack. Another method, which is pretty much the opposite of the former, is to introduce circuitry to allow for an extremely fast recovery, following attacks from transient bursts. To me, this method is all wrong, since, while it reduces pumping effects from transients, it effectively removes, or squashes the transient itself, which leads to what I call the "Sixties Sound" . . . no punch. Also, with either of the above methods, the resulting "two speed release" can become very annoying. I believe this is true because a listener is able to accept a certain amount of a release effect, as long as that effect remains resonably constant. However, if such an effect audibly changes periodically, as with a two step release, the listener may find himself very conscious of the fact that the device is doing something strange to the material.

A methodology which I personally prefer, is to setup a non-linear, or logarithmic release shape, which behaves as follows: Each time the input signal causes the limiter to attack, the *initial* release time is relatively long. If the attack is caused by a single, isolated input excursion, the release will begin to shorten, getting faster and faster as normal gain is restored. However, if the input signal causes a repetitive series of attacks, as might be caused by a steady state low frequency input, or by a series of repetitive transients, the release time will not have time to ever shorten itself, and will remain rather long. Thus, low frequency modulation effects are greatly reduced, as are pumping effects from repetitive transients. However, as soon as the barrage of attacks ceases, the release will exponentially shorten itself, thus quickly recovering. When I speak of "long" and "short" with respect to this log release pattern, these terms are relative to the setting of an actual release time control. In practice, the release time control might be variable from, say, 50 msec. to 5 sec. This is the time during which a full release from 20 dB of limiting would occur. Within this overall release period, the initial release might be around 10 times what the ultimate release rate might be. Since nature is always smarter than Man, there will be instances where a straight linear release of so many dB per second might sound better than such methods as I have described. Thus, provisions should be made to defeat such clever circuitry, and revert to a conventional variable release characteristic, when desired.

Interactive Gain Modification

There are sometimes instances in the recording studio, where it is desirable for excursions of one signal source above a given threshold, to cause a pre-determined gain reduction in another signal. Examples of this sort of usage might be where it is desirable to "duck" the lead guitar every time the lead singer comes in. Thus, the guitar can be screeching away, at maximum volume, yet can be attenuated to leave room for the singer, anytime he utters a word. Another obvious example is the practice of a disc jockey "talking over a record."

In order to best accomplish this, a "range" control would be incorporated. The singer's or announcer's signal would be routed to a side chain input of the limiter, while the program to be "ducked" would be fed to the normal audio input. The circuit would be setup such that whenver the side chain signal exceeded the pre-determined threshold, the audio gain of the signal fed to the normal input would be attenuated, to whatever degree was set on the range control. The attack and release circuitry would still function, and would determine the attack and release characteristics of the "ducking" action.

Another obvious benefit of the accessability of a side chain input lies in that, with the addition of outboard equalization, the limiter may become frequency selective, on any basis the operator might choose.

Compression

So far, we have only concerned ourselves with the limiting function. What about compression? Compression can be equated to "limiting to a lessor degree." In a limiter, whenever the input signal rises to the threshold, a specified output level is established. Further increases in input level cause no (or extremely little) further increase in output level. If the compression ratio (or limiting ratio) is, say, 50 to 1, this means that, above the threshold level, an increase of input level of 50 dB will result in an output level increase of only 1 dB. In a typical compressor design, this ratio is much less, and often adjustable. Typical ratios might range from, say, 1.5 to 1, on up to around 8 to 1. Ratios above 8 to 1 are generally considered to be limiters, rather than compressors.

My personal preference is toward limiting, as the gain is reduced only during the loudest portions of the program . . . thus the apparatus is not active for as great a portion of time as would be the case with compression. Hence, there is less pumping and breathing, plus a more defined control of the output signal.

In some instances, the engineer might simply want to compress the dynamic range of the material, as in office music systems or for other reasons. If so, the lower

compression ratios might prove effective, coupled with long, and thus less obtrusive, release times. This is purely a matter of personal preference, and provisions should be made for these preferances, on compressor/limiter equipment, via selectable or variable compress/limit ratios.

Feedback vs. Feedforward Limiters

A "feedback" limiter/compressor is characterized as having the detector mechanism *following* the electronic gain control element. Such topography is generally to be preferred in equipment where the gain control device is not highly linear, as is the case with FET devices and photocell gain cells. In these examples, the feedback loop tends to straighten out these control non-linearities, and can result in uniform and

high limiting ratios. This configuration has the disadvantage that the signal must have already passed through the controlling element, before control can be exercised. This can lead to problems such as ringing, over-limiting, etc. It is also difficult to structure a feedback limiter for side chain operations, or for accurate compression ratios other than infinity to one.

With the advent of very high linearity VCAs, such as the Valley People (Allison) EGC 101 series, there is much to be gained by using "feed forward" techniques in a limiter/compressor. Hence, the detector operates independently of the VCA, and is normally fed the signal in parallel with the VCA input. Thus, the VCA control signal can be made to arrive at the VCA at the same time as does the audio signal. For interactive gain control situations, as well as for side chain equalizing, the signal fed to the detector need not, in fact, be the same signal which feeds the VCA. Thanks to the extremely linear control response of the EGC 101, very precise ratios may be easily established, while the available range of gain modification is enormous . . . as much as 150 dB if so structured.

Then, of course, there is the all important consideration of the signal handling characteristics of the VCA itself. The EGC 101 VCA is well known for its extremely low distortion production (typically below .003%), very wide dynamic range and headroom, and excellent transient response and slew rate (13 v/usec.).

THE VALLEY PEOPLE GAIN BRAIN II

The forecast has been made that the eighties will be marked by a return to serious and cost effective recording techniques. In order to meet these objectives, it is felt that the equipment offered must be functional and straightforward, while being capable of the highest possible level of audio performance obtainable with today's electronic technology. In the past couple of years, this attainable level of performance has undergone marked advances, thanks in part to a number of inexpensive yet very high performance Op-Amps, and in part to the availability of very high quality VCAs.

These ingredients, together with the new detection methods outlined in this paper have gone into the development of a new device for the dynamic manipulation of audio signal sources, that device being the Valley People Gain Brain II.

While most of the theory of the Gain Brain II has already been discussed herein, the following will be a specific discussion of the new design:

The Valley People Gain Brain II is a small modular device, intended to mount in the Valley People TR 804 processing rack. It is designed to offer the functions of a limiter, compressor, and interactive gain control device (for "ducking" and related processes). The device features a linear integration detector, which is adjustable, by a front panel control, for integration times (attack) of from 200 usec. to 200 msec. Overall release time is adjustable from 50 msec. to 5 sec., for 20 dB of release, while either a linear or logarithmic release shape is switch selectable. Compress/limit ratio is continuously variable from 1.3:1, to Infinity:1. A switch stop on the ratio control converts the ratio to a "duck" position with a gating ratio of 1:-50 (50 dB available gain reduction for a "ducking" signal 1 dB over threshold). In the "duck" position, provisions are made to adjust the depth of attenuation anywhere between 0 dB and 48 dB. The threshold is variable from -40 dBv to +20 dBv. Gain reduction is monitored on a 13 element LED display, which indicates up to 50 dB of gain reduction. Provisions are made for both side chain operation, as well as for stereo intercoupling.

As for audio performance, the Gain Brain II offers a maximum signal-to-noise ratio of 112 dB, with the output noise at a 0 dBv threshold setting being -91 dBv re .775 v RMS (20 Hz to 20 kHz). Static distortion runs a typical of .003% THD, while dynamic distortion, as with all limiter/compressors, is a function of the selected release parameters. With respect to conventional linear release limiters, the dynamic distortion production of the Gain Brain II, in log release position, is on the order of 1/10 for a given release time. Full power bandwith is 150 kHz.

Summation

It is felt that the recording industry of the eighties will demand a higher level of sophistication in the audio performance of equipment which is used. Concurrently, it is felt that this equipment must offer increased functional simplicity, on an improved basis of cost effectiveness, with less gadgetry. It is with these objectives in mind that Valley People enters this new decade. □ □ □

Note: During August, 1980, Allison Research, Inc., together with Valley Audio, officially merged into VALLEY PEOPLE, INC. Information on the products of the merged operations may be obtained from:

VALLEY PEOPLE, INC.
P.O. Box 40306
2820 Erica Place
Nashville, TN 37204
(615) 383-4737
Telex: 558610 VAL PEOPLE NAS

New Products

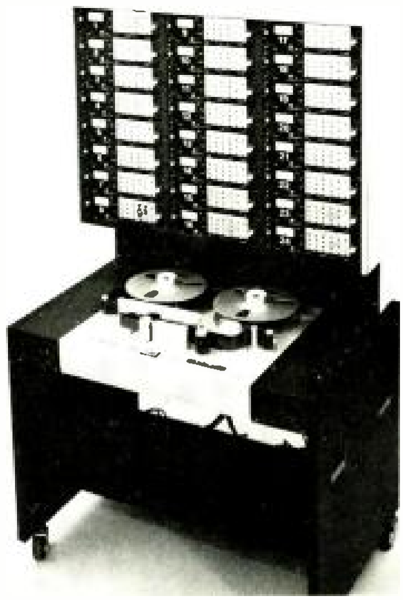
STUDER INTRODUCES MARK III EDITION OF THE A80 VU 24-TRACK RECORDER; ANNOUNCES PRICE REDUCTION

The new A80VU Mark III has been outfitted with a completely redesigned, "close proximity" headblock, with an in-line erase head, for virtually undectable drop-in/drop-out operations and totally reliable performance in the most demanding situations.

This latest addition to the A80 family features a solid diecast aluminum chassis which tilts up for complete serviceability, while remaining totally operational.

The tape drive system is electronically controlled, with fully integrated switching and sensing logic. Tape sensors monitor tape motion and direction as well as detecting the end of tape. The servo-controlled spooling motors keep tape tension constant at any speed and in any mode. The capstan motor itself is servo-controlled, and its speed can be varied over a range of ±7 half-tones.

Each I/O channel module is a self-contained unit with its own voltage regulator, record and playback electronics, and VU meter with buffer amplifier. The A80 VU Mark III is also equipped with a new control logic circuit which has been especially developed to meet the needs of American Recording Engineers.



The new A80 VU Mark III system includes a 20-address memory, microprocessor controlled, programmable autolocator with channel remote which enables all tape deck functions to be operated within easy reach of the engineer at console position. The auto-

locator remote control is packaged as a fully adjustable, freestanding unit.

In addition to the increased performance efficiency of the new Mark III, Studer announces a reduction of nearly 18% from the price of Mark II. Similar price reductions are announced for the 4-, 8-, and 16-track versions of the A80 VU.

Headblock conversion kits and installation will be available shortly for updating the Mark II A80 VU 24-track to the new head arrangement of the Mark III.

Suggested retail prices: A80 VU-24-2"C Package, including 20 position autolocator, channel remote and stand — $46,000.00.

A80 VU-24-2"W Package, including 20 position autolocator, channel remote and stand — $46,500.00.

For additional information, contact:

STUDER/REVOX AMERICA, INC.
1425 ELM HILL PIKE
NASHVILLE, TN 37210
(615) 254-5651

for additional information circle no. 77

M1000 MARSHALL MINIMODULATOR

This new high speed programmed multi-tapped delay line/special effects processor provides studio quality Time Modulation effects, with new ease and speed of use.

Single input, true stereo output configuration allows exotic effects such as simultantaneous positive and negative stereo flanging with cross-fed feedback, stereo multiple tracking, and much more, in addition to the usual delay processor effects. Phantom center circuitry allows full field dynamic image movement (uses the full 70:1 sweep to position image by modified Hass routine) or three isolated sound sources to be generated.

Information needed to set up effect parameters no longer has to be learned and then worked out on the front panel by carefully adjusting several controls, as it is kept in non-volatile internal digital memories. For example, selection of the proper sweep range, positive and negative summing modes, proper delay taps, stereo output assigns, feedback polarity (and EQ), and more can be done with one key stroke.

You simply touch a key pad and the Minimodulator programs itself correctly for the desired effect, instantly and reliably. Extemely wide range of high performance special effects plus echo.

Marshall also is introducing a new system of remote controllers for these units. Several formats are offered to meet any situation. The RC1 remotes all of the front panel key pad controls and adds selection of new modulation waveforms, RSV (random slow voltage) for truly natural automatic double tracking, four different types of envelope following, and right/left output transpose;

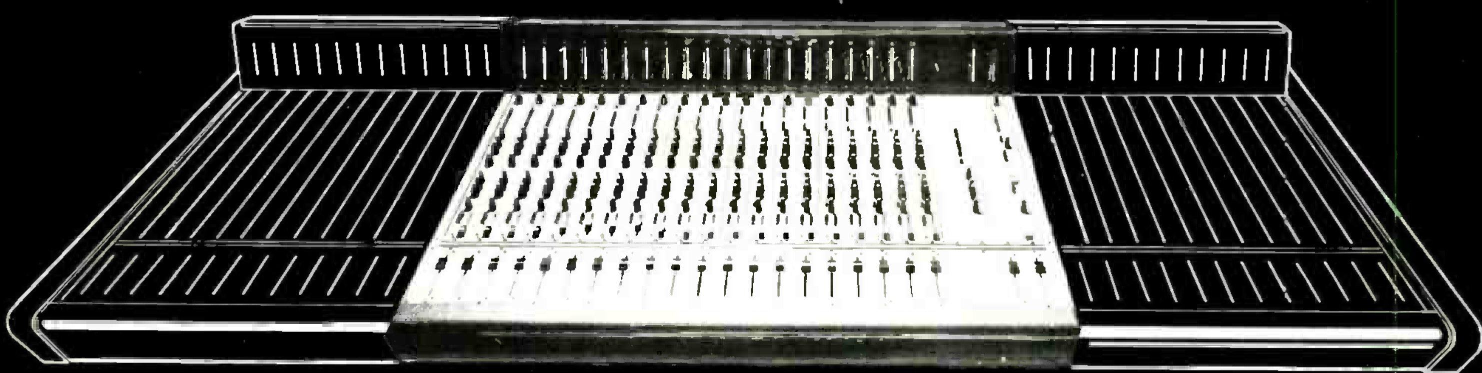
Affordable Technology

SYNCON series A

Syncon Series A has all the creative features of consoles twice its price.

Series A has been designed to give maximum flexibility within a standard frame format. Sophisticated PCB design has virtually eliminated hardwiring making Series A an incredibly reliable and serviceable console; an important factor for studios without resident electronic 'Whiz Kids'. Add to this full parametric Eq and a superb status routing and grouping system, which enables 28 tracks or effects to be mixed through 14 stereo subgroups. The result is one of the most flexible and cost effective consoles to have ever been produced.

SYNCON series B



AHB explodes the myth that 'state of the art' technology cannot be designed into a low price mixing console— The Series B proves it can.

Now through our philosophy of 'affordable technology' we have brought out a console that is both exceptionally advanced and adaptable within the reach of studios of all sizes. In its most basic format Series B is ideal for small 8 and 16 track studios and yet with no factory modification it can be expanded to a 44/24 fully automated console with full function patchbay. The Series B is the most important new audio product this decade. Check it out!

Made in England by:

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ALLEN AND HEATH BRENELL LTD.
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652 Glenbrook Road
Stamford
Connecticut 06906
Tel: Tollfree (800) 243-2598

WEST COAST:
ACI/FILMWAYS Pro Audio Sales
7138 Santa Monica Boulevard
Hollywood
California 90046
Tel: (213) 851-7172

CANADA:
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Ontario L4V 1H7
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all from a 16 key system.

The RC2 remotes the entire front panel, adds the controls that the RC1 adds, and will control two units in various lock or free run relationships. It also adds several more functions. The RC3 is a stage footswitch remote.

Full 70 to 1 continuous sweep range. Programmable LFO instigation. External C.V. input. Low level (quarter inch) and line level balanced (XLR) in and out. All functions including bypass noiselessly remoted by logic to ground. Full 95 dB T.D.R. at any delay with delays to 250 ms. or 450 ms. with optional memory. Internal power supply. Simplified stereo Time Modulation from $995.00.

MARSHALL ELECTRONIC
1205 YORK ROAD, SUITE 14
LUTHERVILLE, MD 21093
(301) 484-2220

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NEW DIRECT BOX CUE SYSTEM BOX FROM SYE MITCHELL SOUND

Designed for his own use and refined over years of studio use, these Sye Mitchell boxes are now available for purchase. They are completely hand-assembled with all American parts and extensively tested before delivery.

Direct Box: At the heart of this passive direct box is a hand-selected heavy-core Triad A-11-J transformer to assure the most faithful signal transmission — even with low range, hhigh level sources. Switches include input padding (0, 10 dB, 20 dB), ground lift, output matching (50, 250, or 600 ohms), and phase reversal. Heavy duty padding is available on special order. The Direct Box measures 4¾" x 3¾" x 3½".

Cue System Box: This well-thought-out box incorporates a protective resistive network which prevents headphone problems from shorting out the entire box and cue system amplifier. Controls are precision, hermetically sealed, 2 two audio taper pots. There are three level controls and six headphone jacks. A panel switch allows selecting A, B, or A & B (stereo). Selector switch and volume controls do not affect box output. This permits boxes to be coupled together and still maintain individual control. Connectors are XL type. The Cue System Box measures 7½" x 4¾" x 3½".

Both boxes are made of solid heavy-gauge aluminum and are fully engraved. Purchaser's name or studio name is also engraved at no charge. The finish is scuff-resistant hammertone with baked-on enamel. Custom colors are available.

List price for either box is $150.00, until November 15, 1980. Both are covered by a full 90-day guarantee.

SYE MITCHELL SOUND CO.
22301 CASS AVENUE
WOODLAND HILLS, CA 91364
(213) 438-4977

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THE ACCUSONIC TUNER FROM SONIC RAINBOW LABS

The Accusonic Tuner is said to be the simple solution for tape recorder alignment. Until this new product was introduced, according to the manufacturer, the professional recordist wishing to

electronically align his recorder had the choice of hauling it to a maintenance shop or investing a sizable sum in test equipment which requires a broad knowledge of electronics to operate. The Accusonic Tuner is the first of a new breed of test equipment that is human engineered. It provides ease of operation and high accuracy. Instead of gazing at meters as the tape machine trim pots are adjusted, the operator simply tunes an audible tone to a reference tone which emanates from a built-in speaker. Just like tuning a guitar. A selector panel provides all the necessary functions for tape recorder alignment at three different reference levels. A 20 Hz to 20 kHz sweet frequency oscillator is provided to adjust the record side and head azimuth. A non-technically oriented musician, it is said, can accurately align his tape recorder with a proper test tape, the Accusoninc Tuner, and his trained ear.

SONIC RAINBOW LABS
112 SPACE PARK DRIVE
NASHVILLE, TN 37211
(615) 832-2646

for additional information circle no. 82

SONY INTRODUCES NEW GENERATION DIGITAL RECORDING PROCESSOR and DIGITAL EDITOR

Sony unveils the PCM-1610 Digital Audio Processor at the 67th Audio Engineering Society Convention. As with the PCM-1600, the two-channel, 16-bit PCM-1610 provides better than 90 dB dynamic range, unmeasureable wow and flutter, and also incorporates such new features as a SMPTE

Sexless.

Whether you realize it or not, every cosmetic feature . . . in fact every feature on an amp costs you money. Sometimes, these features cost you performance and reliability, because the manufacturer is trying to be price competitive. Lately there's a trend to make every amplifier a little "sexier" than the competitions.'

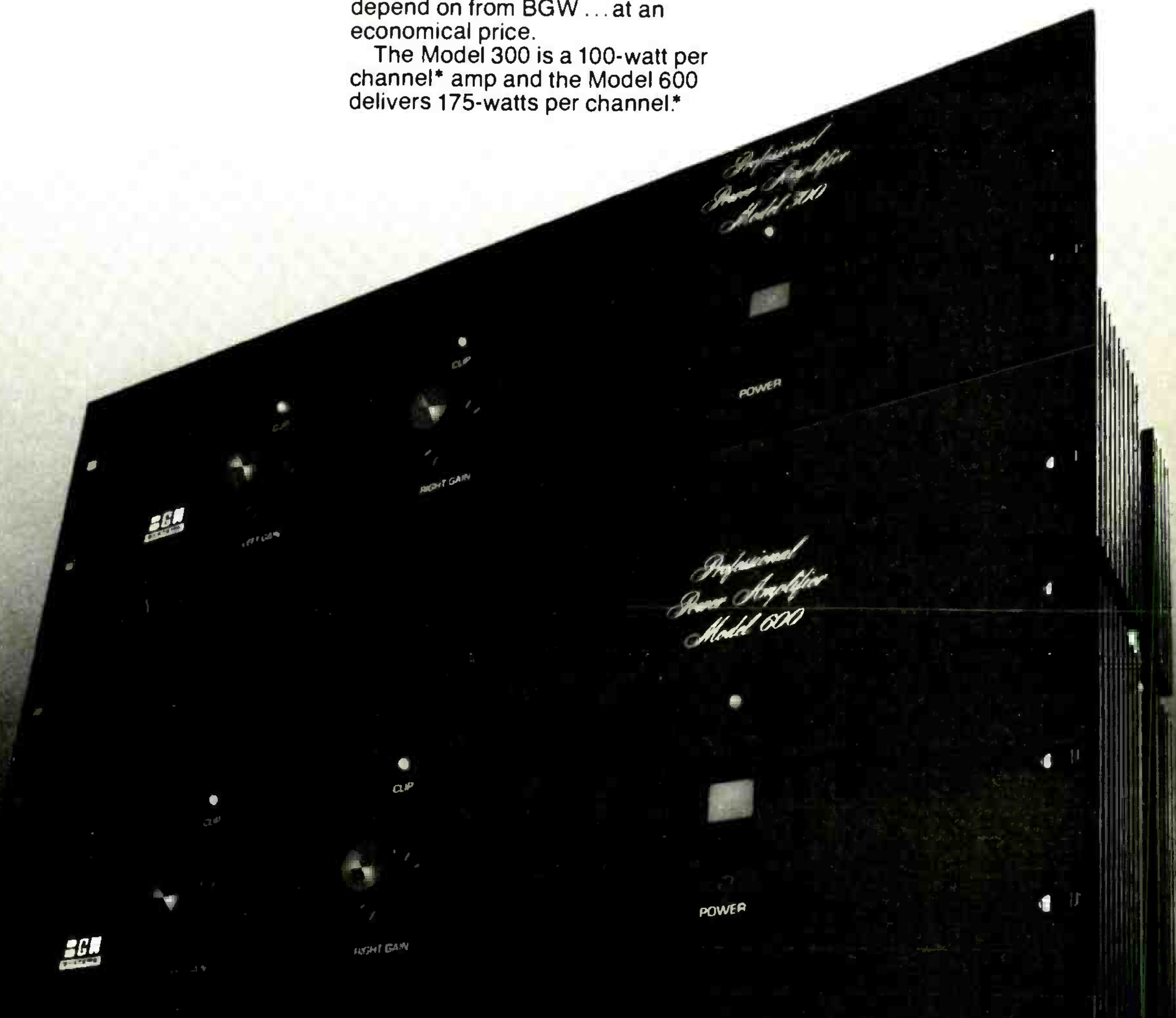
At BGW we know there are applications where all you want and need is performance and reliability. No bells and whistles. That's why we developed the BGW Models 300 and 600. These are true "basic" amplifiers, but they are packed with all the performance and reliability you depend on from BGW . . . at an economical price.

The Model 300 is a 100-watt per channel* amp and the Model 600 delivers 175-watts per channel.*

Both feature full complementary output stages . . . modular construction . . . a high-speed 15-MHz op-amp front-end . . . individual front panel gain controls . . . stereo/mono switch . . . sophisticated loss-of-feedback clipping indicators . . . separate signal and chassis grounds . . . and rugged all steel construction. Both are also available with built-in 70/25 volt autotransformers.

Check out the BGW 300 and 600 today.

*Minimum average continuous power output at 8 ohms over the full 20 Hz-20kHz band. 4 ohm rating significantly higher.

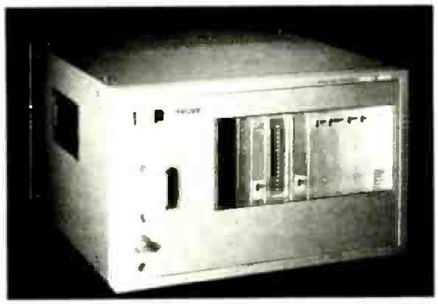


Depend On Us. BGW SYSTEMS

BGW Systems, Inc., 13130 S. Yukon Ave., Hawthorne, CA 90250 (213) 973-8090 In Canada: Omnimedia Corp., 9653 Cote de Liesse, Dorval, Quebec H9P 1A3 time code generator, a switchable transformer/transformerless function on inputs and outputs, and a variable mute time selector for protection against bit errors.

After achieving leadership in the field of digital recording with the PCM (Pulse Code Modualtion) system, the PCM-1610 marks another step forward. The new processor further establishes the advantages of digital over analog recording. Wow and flutter are eliminated by the absence of any mechanical functions in the actual recording, noise reduction systems are unnecessary because of absolutely no tape hiss or print-through, and the much greater dyanmic range makes limiting and compression a matter of choice, not necessity.

Since the PCM-1610 utilizes Sony broadcast U-matic videocassette recorders for the storage of information, the unit incorporates an advanced error-correcting code. Called the Cyclic Redundancy Check Code, it protects against any drop-out, analyzes and replaced missing information, and eliminates any pulse noise or crossword error. With the addition of the mute selector, the system protects against the hazards of bit errors.



The automatic SMPTE time code generator allows immediate editing and assemblage of recorded material, utilizing the new Sony DAE-1100 Digital Editor, or with a video editing console. Since digital tape-to-tape dubbing reproduces clone-perfect copies, there is no generation loss in re-recording, no degradation of signal in the editing process, and there is uniform mastering of unlimited pressed dsics. With the flick of a switch, the PCM-1610 incorporates transformers at the input or output level or allows for transformerless operation, providing flexibility in all situations.

The PCM-1610 provides a choice of either the 44.056 or 44.1 kHz sampling frequency, making it compatible with either the NTSC or the European PAL video formats, and uses 16-bit linear quantization, the accepted professional standard. Tapes that have been made on the existing PCM-1600 Digital Processor are totally compatible with the new PCM-1610.

The PCM-1610 weighs only 77 pounds and is the same size as the PCM-1600: 17 x 10½ x 18¾ inches. The unit is rack mountable, and is available for immediate purchase. Price: $25,000.

DAE-1100 DIGITAL EDITOR

Among several breakthroughs in digital recording, Sony has introduced the DAE-1100 Digital Editor. The compact, easy-to-operate unit features a "search dial" which can locate an edit point with greater precision than the width of a razor blade . . .

Studio quality microphones that don't need a studio to survive.

The CS15P condenser cardioid microphone is equally at home in a recording environment or broadcast studio. When hand-held it puts sex appeal in a voice with its bass-boosting proximity effect. With shaped high-frequency response and its ability to handle high sound pressure levels (140 dB with 1% THD at 1kHz), the CS15P is ideal for close-up vocal or solo instrument miking applications.

When boom mounted, the CS15P has better gain-before-feedback and a better signal-to-noise ratio than most shotguns. It's phantom powered and it's rugged.

The CO15P condenser omni extends frequency response to the very limits of audibility, 20 to 20,000 Hz. Unlike other "omni's," the CO15P maintains its omnidirectional polar pattern at the very highest frequencies. Perfect for the distant miking of an entire orchestra as well as up close on individual instruments. And like the CS15P, it's phantom powered and it's rugged.

The Electro-Voice warranty
Electro-Voice backs up these two microphones with the only unconditional warranty in the business: for two years we will replace or repair your CS15P or CO15P microphone, when returned to Electro-Voice for service, at no charge—no matter what caused the damage!

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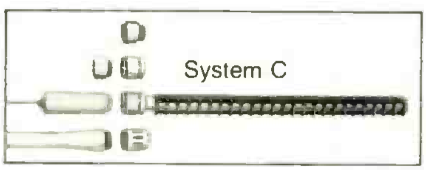
We can do this because we build these microphones to meet our standards for performance, ruggedness and durability. We accept nothing less, and if you're a professional, buying a professional quality microphone, you shouldn't either.



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

New Products

an accuracy of 362 microseconds. The editor provides a widely variable cross-fade edit capability as well as a convenient preview feature.

Because of the clone-perfect aspect of digital recording, as opposed to the inherent degradation and generation loss of analog recording, digital tape-to-tape editing maintains the quality of the original tape throughout the editing process. Utilizing

Sony broadcast U-matic videocassette recorders to store the digital information, the editor allows the sequencing and combination of varied material for a finished product with extremely accurate, undetectable edits. The preview function of the DAE-1100 makes editing fool-proof, and lighted buttons which flash in sequence show the order of operation.

The DAE-1100 incorporates a SMPTE time code genrator reader allowing any chosen point to be memorized and quickly located at a later time by the automatic locator. Digital LED display counters for both recorder and player(s) are capable of accurately displaying up to 23 hours, 59 minutes, 59 seconds, and 29 frames. The exact edit point is arrived at by a choice of playback speeds, a six second working area is stored in the memory, and fine-trimming to the precise edit point is found with the "search dial," making the operation as simple as open-reel analog editing.

The DAE-1100 insures smooth edits by the use of a gain offset fader to equalize the edit-in and edit-out points. A range of cross-fade times from 1 millisecond to 99 milliseconds assures audio continuity, and provides highly flexible fade-in and fade-out effects. With the use of the preview feature the edit can be monitored, then adjusted for varying volumes, cross-fade characteristics, and exact edit points. Once the perfect edit is chosen, the entire operation is completed automatically. The editor easily accommodates the range of needs from the precise editing necessary for the typical classical recording to the subtle requirements of advanced rock 'n' roll.

The Sony DAE-1100 Digital Editor is designed for use with the Sony digital audio processors: the PCM-1600, the PCM-1610, and the PCM-100. The editor is comprised of two components, a rack-mounted unit housing the electronics (16¾" x 10½" x 21") and a compact remote-control keyboard which can be conveniently placed where needed in the studio. Available immediately, the DAE-1100 costs $45,000.

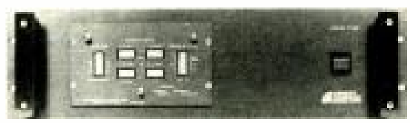
SONY CORP. of AMERICA
9 WEST 57 STREET
NEW YORK, NY 10019
(212) 371-5800

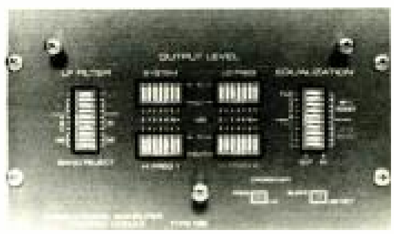
for additional information circle no. 87

THREE OUTPUT BIAMPLIFIER POWER SYSTEM FROM ADVANCED TECHNOLOGY DESIGN

Advanced Technology Design Corp., has introduced the industry's first three-output biamplifier, according to a company spokesman. Designated model 7132, the power system delivers 450 watts low frequency and 150 watts from each of two high frequency channels into 4 ohm loads. The two separate high frequency channels allow independent level and equalization adjustment for near throw and far throw high frequency horns. As in other Advanced Technology Design products, all controls are precision dip-switch controls, with no potentiometers used.

The dip-switch control system eliminates inaccuracies and uncertainties in the control of amplifier gains, crossover settings, filters, and equalizers. Each control function is performed with step-switch accuracy, and may be repeated, altered or eliminated with absolute precision. All controls are grouped in a compact, easy-to-understand, field replaceable control module. Precise adjustments to system levels and equalization are easily performed and repeated or adjusted without the aid of complex instruments. A security cover conceals the control module during normal use.





The 7132 includes selectable low frequency cut-off to restrict the operating frequency range of the amplifier to that of the low frequency loudspeakers, and equalization to compensate for rising response characteristics in the region of 100 Hz or 200 Hz. High-pass filters with 18 dB per octave slope at either 40 Hz or 80 Hz may be selected, and up to 7 dB of band reject, in one dB steps, centered at either 100 Hz or 200 Hz is available to tailor the signal to the performance capabilities of the low frequency loudspeaker system.

High frequency equalization, providing up to 15 dB boost at 20 kHz may be independently selected for each of two high frequency channels, to compensate for driver roll-off characteristics.

Each output amplifier is independently

for additional information circle no. 86

Introducing the world's finest digital delay system

. . . again.

In 1974, we introduced the Lexicon Model 102 Digital Time Delay System. It became the standard of the industry — the system specified by users seeking the ultimate in performance and reliability.

Now we're introducing the world's finest again — the Lexicon Model 122 Series, successor to the 102.

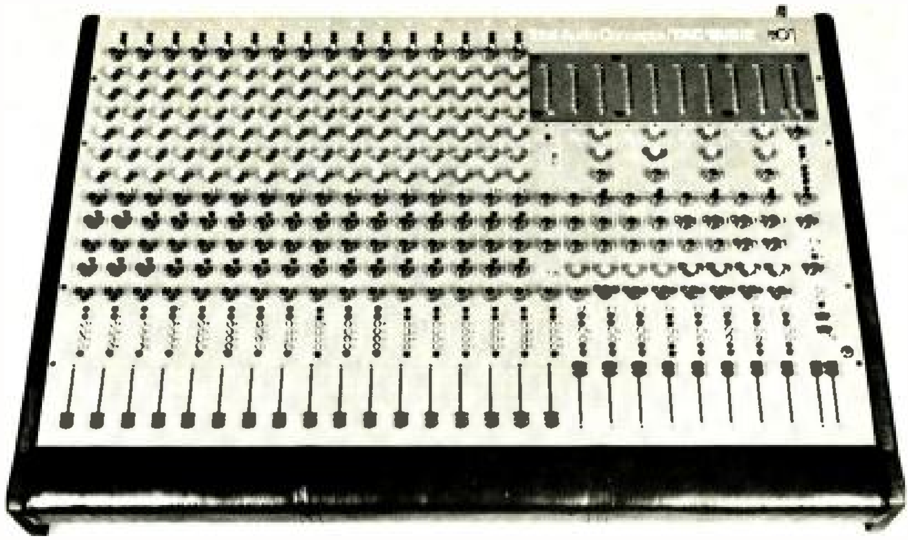
You get a lot to like. 14-bit floating-point encoding with 6 dB gain steps for 95 dB dynamic range free from noise pumping and artifacts. 9-pole Butterworth filters on inputs and outputs to ensure absolutely flat frequency response. PCM digital encoding for superior performance free from slew rate and high frequency/ amplitude distortion of delta modulation systems. Mono or stereo models, both fully modular. Advanced A/D/A and Sample and Hold technology to keep distortion typically less than 0.03%. Limiter stage on input for soft limiting at digital overload.

Look into the new Model 122 Series from Lexicon, maker of the world's finest digital delay system . . . again.

lexicon

Lexicon, Inc., 60 Turner Street, Waltham, MA 02154 • (617) 891-6790/Telex 923468
Export: Gotham Export Corporation, New York, NY 10014

for additional information circle no. 88



adjustable over a 31.5 dB range in half-dB steps, plus off. A system level control is also provided, which is adjustable over a similar range.

Deliveries of the 7132 began in May, the first ten being shipped to Stanal Sound, for installation at the Poplar Creek outdoor concert theater. The 7132s were used to power the delayed and distributed lawn clusters at ten locations within the theater. Each cluster consists of a pair of fifteen-inch LF units and two constant directivity horns. The HF horns were used as near throw and far throw devices, and the separate HF outputs and HF equalizers on the 7132 made the task of balancing the system a simple one.

The 7132 is said to replace as many as six separate chassis in a typical application, costing as little as one-third that of conventional products.

ADVANCED TECHNOLOGY
DESIGN CORP.
16123 VALERIO STREET
VAN NUYS, CA 91406
(213) 989-7577

for additional information circle no. 90

T.A.C. 1682 CONSOLE

The T.A.C. 1682 has a standard format of 16/8/2 with optional auto-direct assigns from channels 9 - 16. The console is supplied with Ruido linear element faders, and light emitting diode meters will be used throughout.

The mike/amp transformer is balanced at 1 Kohm for 200 - 600-ohm mikes, with gain variable +20 to +60 dB. Line amplifier differential is balanced, normally unity gain. -20 dB constant impedance pad; channel mute button.

The equalizer is a swept mid-band. Four-band equalization section is fitted: HF - ±14 dB, 10 kHz shelving; MF-1 - ±14 dB, sweep range 500 Hz - 16 kHz; MF-2 - ±14 dB, sweep range 80 Hz - 2 kHz; LF - ±14 dB, 80 Hz shelving.

Also fitted is a swept high pass filter at 120 Hz, 12 dB/octave.

Four mono auxiliary send busses each with pre/post button.

Four bus assign button with LED indication; panpot panning odd/even; for a total of eight busses as well as a separate stereo bus.

Monitor: Separate eight track monitor section.

The speaker monitor input may be selected from: a) any of two external sources; b) stereo tape one; c) stereo tape two; d) the stereo mix.

Level control with mute and dim (-20 dB) buttons. Solo lamp. Solo trim control, and auxiliary monitor output level control.

Talkback system with microphone input XLR.

Eleven LED meters are fitted, 8-track reading; stereo mix, and PFL/solo.

Specifications include: Bus output noise, -85 dB; Stereo bus output noise, -78 dB; Microphone ampifier noise, -123 dBv; Headroom, +23.0 dBv (0 VU = +4 dBm).

The unit is available through:

EVERYTHING AUDIO
SUITE 1001
16055 VENTURA BOULEVARD
ENCINO, CA 91436
(213) 995-4175

for additional information circle no. 91

STL ANNOUNCES LINE OF NEW TEST TAPES

Standard Tape Laboratory is now offering a complete line of test tapes made specifically for use with the Sound Technology Tape Recorder Test System. These tapes facilitate the measurement of the required parameters of any tape reproducer when used with this test system. The tapes include all formats including cassette and cartridge. An introductory set of tapes, for measurement of level, azimuth and wide band response is available at a special package price, when ordered directly from STL.

STANDARD TAPE LABORATORY
26120 EDEN LANDING ROAD — #5
HAYWARD, CA 94545

for additional information circle no. 92

AHB SYNCON SERIES B

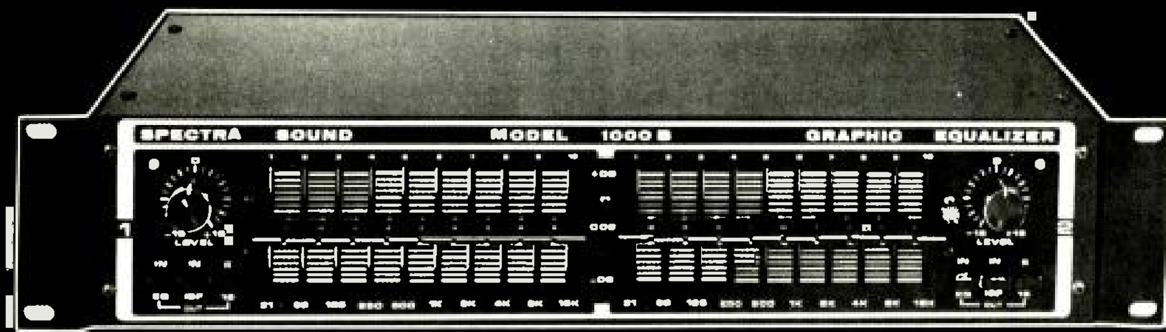
To meet the ever increasing quality demands of the professional audio industry and in an effort to bridge the gap between advanced technology and limited budgets AHB has created the Syncon Series B.

The console is free-standing and attractively styled with tinted acrylic meter hood, padded arm rest, and solid hardwood side trims.

Compact in size, Series B is of the "in-line"

NEW

SPECTRA SOUND



MODEL 1000B STEREO GRAPHIC EQUALIZER

- ☐ HIGH RELIABILITY
- ☐ EXTREMELY LOW NOISE
- ☐ SWITCHABLE RANGE
- ☐ UNMEASURABLE DISTORTION
- ☐ SYSTEM BY-PASS SWITCH
- ☐ LED OVERLOAD INDICATORS
- ☐ BOTH BALANCED AND UNBALANCED INPUTS AND OUTPUTS

Now, from **SPECTRA SOUND,** comes a new, high quality, ten band, two channel, graphic equalizer designed exclusively for the discriminating PROFESSIONAL. Wherever equalization requirements exist, that dictate high reliability with extremely low noise and distortion, the **SPECTRA SOUND** Model 1000B is the only choice. For further information contact:

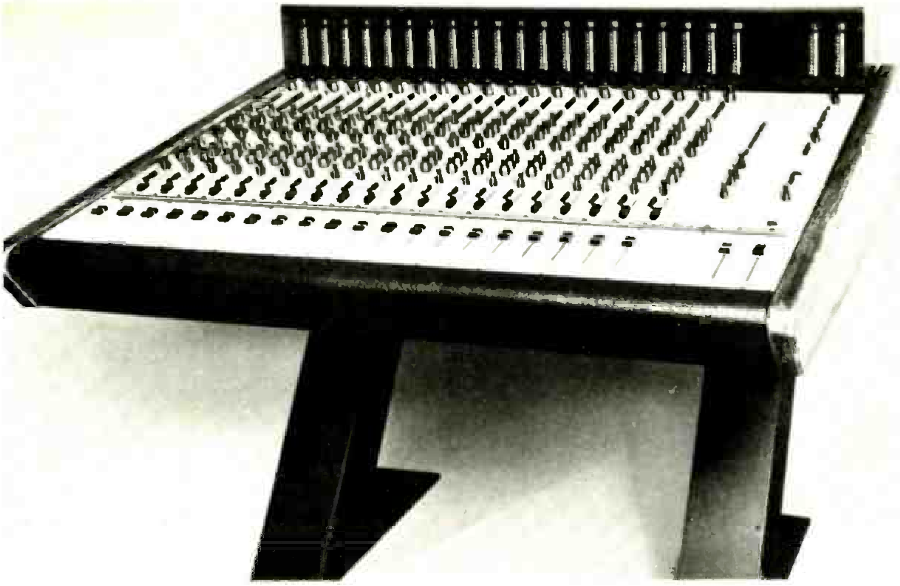
SPECTRA SOUND

A WHOLLY OWNED SUBSIDIARY OF SPECTRA SONICS

3750 AIRPORT ROAD   OGDEN, UTAH 84403   (801) 392-7531

for additional information circle no. 93

FOR A DEMONSTRATION VISIT ROOM 656 AT THE AES CONVENTION



concept and totally modular in construction. The standard main frame has a maximum capacity of 20 in/output modules and is complete with all the necessary master and monitor functions required by the latest multitrack recording techniques.

It is ideal fo 8-, 16-, or 24-track operation and with absolutely no factory modifications can be expanded to a 44/24 fully automated console, with comprehensive integral patchbay facilities.

The Standard Series B features: Twenty-four track grouping and stereo re-mix/monitor group; Each in/output module incorporates the newly designed AHB mike input, fully balanced with 48 V phantom power and phase reverse switch; a separate line input with continuously variable gain; a tape monitor input with its own fader, pan, and PFL, plus an effects input; a record/re-mix group with or without master fader; in-place LED 12-segment VU meter; a comprehensive EQ section, including two overlapping sweet equalizers an separate low cut filter; independent fader mute and "in-place" solo functions utilizing solid state techniques; LED control function indicators; four auxiliary mixes with versatile matrix output system; auto PFL which overrides monitor audio and meters; and a choice of three fader options (including P&G 105 mm conductive plastic faders).

Syncon Series B comes complete with detachable stands and a 19" rack mounting heavy duty regulated DC power supply. All frame formats are automation ready and can accept any retrofit automation package that is available.

Considerable time and design effort has been put into Series B to produce a high quality sophisticated purpose built console for multitrack recording, overdubbing, and master mixing.

ALLEN & HEATH BRENELL, LTD.
PEMBROKE HOUSE
HORNSEY, LONDON N8, ENGLAND
TELEPHONE: 01-340-3291
TELEX: 267727 BATGRP G

for additional information circle no. 95

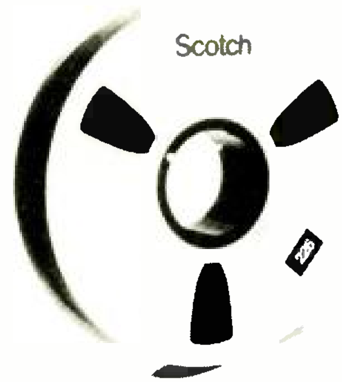
FAX INTRODUCES AUTOMATED FADER

Fax Audio, Inc., introduces the Series One programmable fader featuring solid state, touch sensitive membrane switches, comprehensive visual indicators that include simultaneous LED monitoring of precise Read and Write levels, and the most complete array of functions ever offered in a modular automated fader. The Read and Write level LEDs, located adjacent to the fader scale, indicate the precise levels of data to and from the computer with a 0.25 dB resolution over a 30 dB range (0 dB to 30 dB), with a single mute LED located at the infinity mark. The environmentally sealed, gold plated touch switches (rated at over 5 million cycles) offer the benefit of completely noiseless module operation and are accompanied by backlit legends which provide constant visual indication of fader functions. The fader's low profile panel, manufactured with an exceptionally wear resistant material, will retain its new look for many years.

In addition to all the Read and Write functions inherent in automated faders, the FAC Series One features all of the following: A safe mode which locks out all functions except Group Master; a true update function which enables addition or subtraction of previously recorded group or channel levels; a Read/Write auto switchover which, when used in conjunction with the Read and Write level LEDs, provides precise visual electronic nulling (overlapping the Read and Write LEDs) which triggers the automated switching of the fader from Read to Write or vice-versa; a Trim switch which activates the Trim knox as either the fader trim (providing an additional 12 dB of gain) or as the channel level when the fader is designated as a group master; a completely noiseless channel or group mute which can be remotely disabled; a channel or group mixdown Solo function, capable of soloing any fader within a soloed group; a Group select switch, adjacent to a seven segment digital display, which allows the fader to be assigned to any of nine Group

Suntronics

"FOR THE NEW GENERATION" of Recording Studios the TASCAM 85-16, one inch 16 track Recorder and Scotch 226 Studio Mastering Tape will lead the way in the 80's



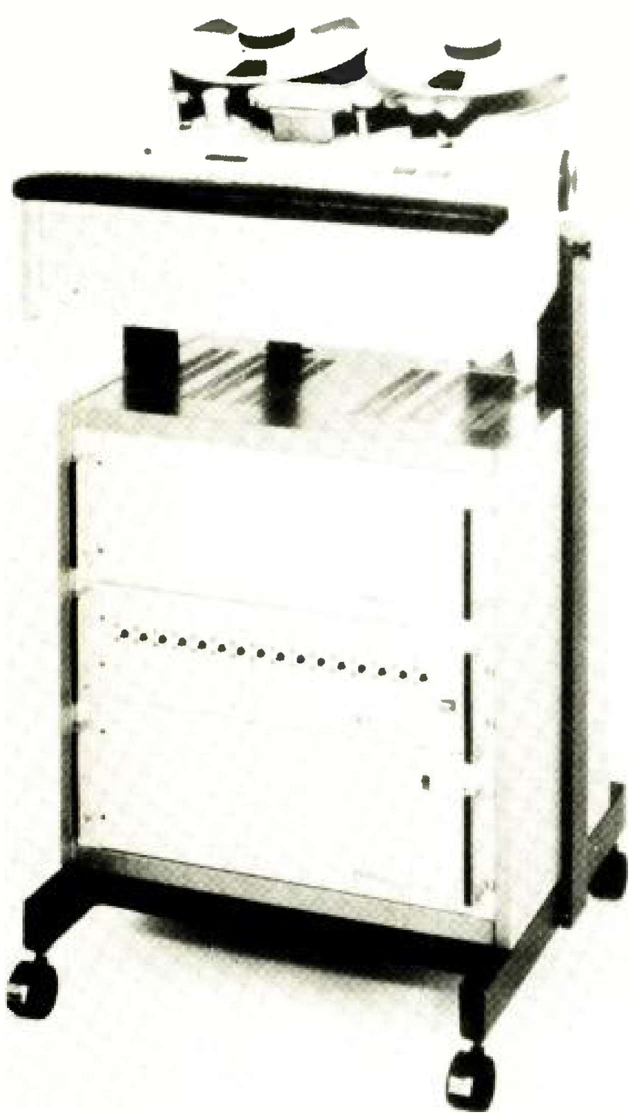
3M Scotch® 226 Studio Mastering Tape

TASCAM SERIES

TEAC Professional Products Group

85-16

- ☐ 16 tracks on 1" tape
- ☐ 15 inches per second, and ±10% record/play speed control
- ☐ 4 digit display for tape speed (% of 15 ips) or elapsed time
- ☐ Accurate zero-search function
- ☐ Plug-in front accessible PC cards for record/play amps and dbx encode/decode processing
- ☐ Three DC servo motors
- ☐ Spooling mode for fast winding and neat tape pack
- ☐ Integral dbx noise reduction
- ☐ Adjustable transport mounting angle
- ☐ Superior record/play audio performance from DC-coupled FET amplifiers
- ☐ 28 dB system headroom



THE 85-16 IS $11,500.00 COMPLETE
CALIBRATED TO SCOTCH 226 STUDIO MASTERING TAPE

SUNTRONICS
CREATIVE AUDIO TECHNOLOGY

(714) 985-0701 985-5307 P. O. Box 734 1620 W. Foothill Blvd. UPLAND, CA 91786

for additional information circle no. 96

busses (1 to 9) or 0 which is unassigned; and also a Group Master switch that enables any fader to become master of the group indicated by its display.

In addition, the FAX fader utilizes an Allison EGC 205M VCA, Penny & Giles fader, and can retrofit into almost any console which employs a modular fader. The system easily adapts to existing console wiring and is electrically installed by FAX. Other features of this exceptionally versatile fader include plug-in motherboard construction, a battery back-up to retain all fader settings during power failures, a Grand Master bus, a Master Calibrate for easy console alignment, an eight function Master Control Module, and a comprehensive owner's manual troubleshooting guide.

FAX AUDIO, INC.
29 ANITA PLACE
AMITYVILLE, NY 11701
(516) 598-1258

for additional information circle no. 98

MXR INTRODUCES DUAL LIMITER

MXR has recently developed the Dual Limiter, a versatile signal processor designed to perform in any compression/limiting application, offering many unique performance characteristics that facilitate its implementation and operation without sacrificing versatility. Its sophisticated internal circuitry enables the Dual Limiter to sound musically natural, even at extreme compression settings. It functions like two completely independent limiters that can be strapped together via front panel switches for stereo-limiting applications. Each channel has In/Out switch, Slope switch, Input, Output, Attack, and Release controls, as well as LED meter, displaying the amount of gain reduction. The detector of each channel is also accessible via rear panel ¼" phone jacks, allowing for external tailoring of the detectors' frequency response. This feature enables reduction of vocal sibilants and fulfills a wide variety of frequency-dependent limiting needs.

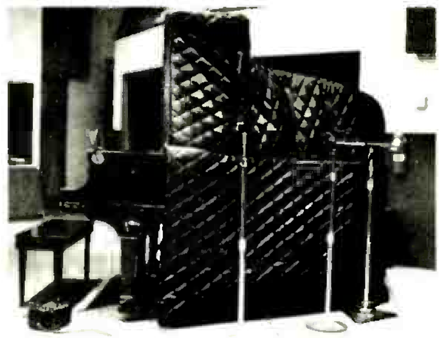
Its balanced inputs, ability to drive 600-ohm loads, +19 dBm input and output capability and standard rack dimensions of 19"(L) x 1¾" (H) x 6" (D) allow it to be easily employed in any professional system.

MXR INNOVATIONS, INC.
740 DRIVING PARK AVENUE
ROCHESTER, NY 14613
(716) 254-2910

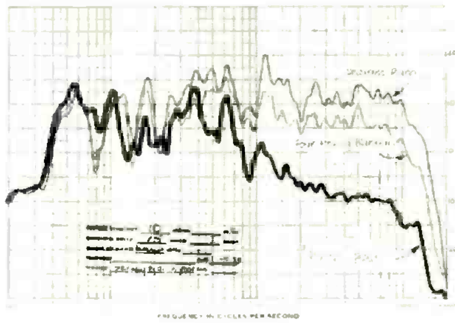
for additional information circle no. 99

ALPHA ACOUSTIC CONTROL PIANO BAG

The piano bag is said to represent an acoustical breakthrough, according to Alpha Marketing director Porche Lottermoser. "For the engineer who is tired of the moving blanket sickness and who wants to achieve a higher degree of separation when recording the piano in the studio." "Not only does the piano bag enclose the essential chamber area of the piano and separate it acoustically, but it performs a visual change in the attitude of how to record a piano," say co-designers Gary and Pamela Brandt.



Test results demonstrate a 15 dB inside-to-out reduction in high frequency information at 15 kHz, and a 3 dB downpoint at 500 Hz as pictured in the graph below.



The test procedure uses an omnidirectional calibrated microphone, centrally placed inside of the piano's sounding chamber, near the hammers, facing up. The top line is the uncovered piano measured on a calibrated flat loudspeaker/monitor system. The second line is the test of the same piano/speaker system using four moveable blankets as typically used by studios to isolate the piano. The results in this case at 15 kHz were no more than 6 dB at best. The third line on the graph represents the test using the Alpha Piano Bag, showing marked improvement in isolation (separation).

Details of construction include two-way zippers which entirely enclose the mike stands as well as full draping to the floor encompassing the entire piano. Every piano bag is custom built after the brand and type of piano is measured exactly. A two-ply

pattern is then cut consisting of two layers of vinyl and fiber-fill. These are quilted together, back-to-back, insuring the greatest amount of isolation.

Alpha Acoustic Control has plans to expand the line to include bass drum covers as well as other products to enhance the quality of sound in the recording studio. Custom design and consultation is also offered.

ALPHA ACOUSTIC CONTROL
BURBANK, CALIFORNIA
(213) 845-0197 & 760-1139

for additional information circle no. 102

NEPTUNE 1420 AND 821 STEREO MIXING CONSOLES INTRODUCED

Available for immediate delivery, with 14 input channels, the 1420 has many of the same features found on NEI's new 821 stereo mixer with many more that make it a truly unique multipurpose mixing system. Some of the professional features found on the 1420 include: (Input Channels) transformer balanced mike and line level inputs (D3F and ¼" phone), Monitor, Effects, and Auxiliary sends, 3-band input channel equalizer, gain control with peak LED indicator, locking solo switch, pan control and slide channel fader. Rerar panel input channel features a channel patching jack for use with external signal processing devices. Master control features and functions include: extensive headphone monitoring system for solo and function monitoring with loud 5-watt headphone amp with gain control, balanced (D3M) and unbalanced (¼" phone) line level outputs for left/right mains and monitor, slide-type master gain (left/right) and monitor master controls with LED VU level indicators, effects and abxiliary master sends and returns with hi or low level return jacks (effects return) and panning for both effects and auxiliary returns, plus much more.

The 1420 comes in an attractive steel chassis with wood sides, built to withstand the abuse and rough handling associated with use on the road or day-to-day in a club or studio. The 1420 is a natural for any type of commercial installation as well. Road/flight cases are available from professional road/flight manufacturers. Optional 220 V, 50 Hz line voltage available from the factory.

Engineered to be a multipurpose mixing console, the Neptune 821 mixing console is claimed to be ideal for a diverse range of applications that transcends both recording and sound reinforcement: stage or studio to club, church, or even keyboards. The 821 is designed for almost any role at a reasonable price.

Professionally appointed with features that include built-in reverb with master level and pan control, separate input preamp input/output jacks for signal processing of input channels, slide-type master output controls (left/right with LED VU indicators) and master monitor control. Other input channel features include mike/line switching, transformer balanced mike and line level inputs (D3F type and ¼" phone connectors), Monitor, Reverb, and Auxiliary sends (Auxiliary send with pre- or post-EQ/fader switching), 3-band input channel equalizer, pan control and oil-dampened slide channel fader. Master control functions and features include: balanced and unbalanced line level outputs for left/right mains and monitor (D3M and ¼" phone connectors), Auxiliary send master with hi or low level return jacks, return control with panning, left/right and monitor rear panle inputs for stacking another 821 for more input channels, plus many more features.

The 821 employs a rugged, steel chassis that can be ordered with attractive wood sides or rack mount ears (the 821 is fully rack mountable) for installation in an equipment rack or top-mount road/flight case. Optional 220 V, 50 Hz line voltage available.

NEPTUNE ELECTRONICS, INC.
934 N. E. 25TH AVENUE
PORTLAND, OR 97232
(503) 232-4445

for additional information circle no. 103

PROGRAMMABLE RHYTHM MACHINE WITH REAL DRUMS FROM LINN ELECTRONICS

The LM-1 Drum Computer is a highly sophisticated, yet easy to operate rhythm machine with real drum sounds — actual digital recordings stored in internal computer memory. Drums include snare, bass, hi-hat, hand claps, cabasa, tambourine, two toms, two congas, clave, and cowbell.

Audio features include a 13 input stereo mixer with volume and pan switches for each drum, and individual outputs. The pitch of each drum may be individually adjusted. This amazing rhythm machine can overdub to tape and sync to practically anything.

The LM-1 Drum Computer holds up to 100 different rhythm patterns — all of which are programmable in real time. Programming features include automatic error correction, variable length of repeating loop (odd time signatures, long patterns, etc.), programmable dynamics, and versatile editing. A "human" rhythm feel is made possible by special timing circuitry. The LM-1 can be programmed to play flams, rolls, build-ups, open and closed hi-hat, etc. All programmed patterns remain in memory even with power off. In addition, tape storage feature allows programmed data to be stored on casette tape for later reloading.

"Chain" function enables the "linking" together of rhythm patterns into a song format. With this important feature, the LM-1 can be told exactly what to play throughout an entire song (intro, fills, verses, choruses, breaks, endings, etc.). Chains, it is claimed, can be quickly assembled and edited.

The LM-1 Drum Computer lists for $5,500.00.

LINN ELECTRONICS, INC.
3249 TARECO DRIVE
HOLLYWOOD, CA 90068
(213) 850-0741

for additional information circle no. 105

PROFESSIONAL TEST CASSETTES AND TAPES

MIS professional tape company has developed a complete line of 10 minute pre-recorded audio alignment cassettes and ¼" reel-to-reel alignment tapes which will allow tape users to keep their recording equipment at peak performance for the ultimate in fidelity reproduction. The line features laboratory produced test tapes, calibrated in accordance with the Dolby reference level, for level setting, azimuth adjustment, equalization, bias setting, and flutter testing. All recordings meet both the JIS standard (Japanese Industry Standard) and the ITA standard (International Tape Association).

The cassette tapes are housed in an exclusive MIS produced 100% aluminum cassette which has won an award for its precision. Although these test tapes have been reasonably priced with the home recordist in mind, MIS has maintained the highest standard of quality and reliability.

MAGNETIC INFORMATION SYSTEMS
415 HOWE AVENUE
SHELTON, CT 06484
(203) 735-6477

for additional information circle no. 108

PEAVEY MINI MONITOR

This recently announced system is a lightweight, extremely portable monitoring system designed for situations where a minimal amount of equipment is required and where ease of handling and operation are major requirements. This system may be used in a vertical/conventional manner, or may be placed on the floor and used as either a 30° or a 45° monitor. A very efficient and cost effective moulding process enables the unique double-wall, high-density polyethylene/plywood construction to form a rugged, roadworthy and economical package.

PEAVEY ELECTRONICS CORP.
711 A STREET
MERIDIAN, MS 39301
(601) 483-5365

for additional information circle no. 109

EMT TRANSIENT LIMITER

The EMT 266 is an audio frequency limiter using feed-forward and variable pre-emphasis techniques to control signal transients in FM broadcast transmitters and disc cutting systems. By employing a 300 microsecond analog delay line in the signal

How Do You Get Real Aural Excitement For Under $30 A Minute?

Real Aphex.

The Aphex Aural Exciter™ gives you a sonic realism obtainable by no other means. And now, for the first time, and in response to your repeated requests, the Aphex Aural Exciter is available for purchase as well as rental. If what you're looking for is just a bright high end, get it with EQ—but if you want the kind of true spatiality, detail and presence that producers of over 4000 albums have insisted on, do what they did: get *real* aural excitement. Get Aphex.

APHEX
Aphex West
7801 Melrose Avenue
Los Angeles, CA 90046
(213) 655-1411 TWX: 910-321-5762

Aphex Licensees
Aphex Audio Systems UK Ltd.
35 Britannia Row
London N18QH England
Telephone: 01-359 5275/0955
Telex: (851) 268279 (BRITRO G)
Aphex Chicago Ltd.
(312) 975-8117
Aphex Audio Systems Australia, Pty. Ltd.
(Sydney) Tel: 212-4920
TLX: (790) AA24035

Aphex Benelux
(Brussels) Tel: (02) 345.44.44
TLX: (846) 26409 (TEMBEL B)
Aphex Brazil
(Rio de Janeiro) Tel: 266-5117
TLX: (391) 1121008 (XPSPCBR)
Aphex Audio Systems Canada, Ltd.
(Toronto) Tel: (416) 363-8138
TLX: 06225500 (OCTOTOR)
Aphex Denmark
(Caphenagen)
Tel: (01) 59-1200

Aphex France S.A.R.L.
(Paris) Tel: 251-4995
Aphex Germany, GmbH
(Frankfurt) Tel: (0611) 55.65.66
TLX: (841) 414073 (ROCK D)
Aphex Hawaii, Ltd.
(Honolulu)
Tel: (808) 521-6793
TLX: 7430148 (SOUND)
Aphex Israel
(Tel Aviv) Tel: 232-143

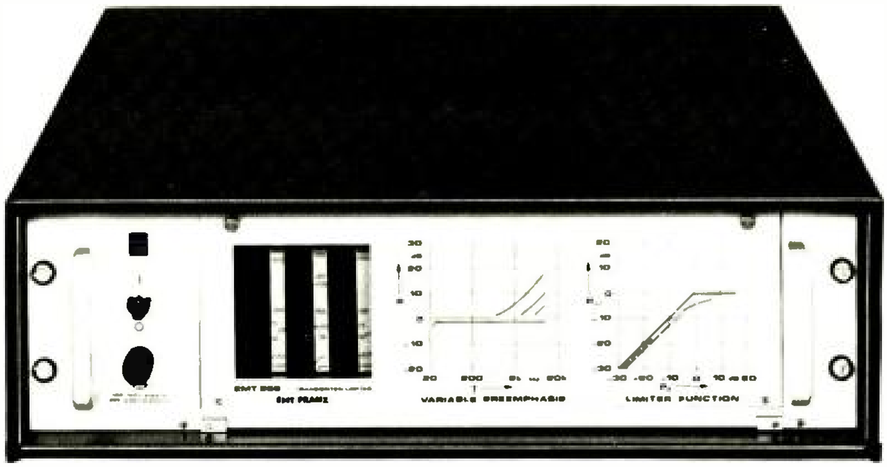
Aphex Italy
(Bologna) Tel: 051-76 66 48
TLX: (843) 511361 (BAUER 1)
Aphex Japan, Ltd.
(Tokyo) Tel: (03) 253-9022
TLX: (781) 222-7097 (APXIEH)
Aphex Midlantic
(Washington D.C.)
Tel: (202) 363-1228
Aphex New York, Ltd.
(West Orange, New Jersey)
(201) 736-3422 (212) 964-7444
TWX: 710.994.5806
(APHEX LTD WOGE)

Aphex Norway
(Oslo) Tel: 14 93 71
Aphex Phillipines
Tel: 704-714
TLX: (722) 23071 (JMGPH)
Aphex South, Inc.
(Nashville)
Tel: (615) 327-3133
Aphex Spain
(Madrid) Tel: 267-5222

Aphex Systems (Suisse) SA
(Le Mont-Sur Lausanne)
Tel: 021/33.33.55
TLX: (845) 24107
(VOGUE CH)
Aphex Texas, Ltd.
(Dallas) Tel: (214) 351-6772
Aphex South Africa
(Johannesburg)
TLX: (960) B-2440 S.A.

for additional information circle no. 110

path, the limiter's control circuitry is able to calculate the gain reduction required for each transient before it arrives. The EMT 266 thereby completely eliminates transient overshoots that plague even the fastest conventional limiters. The "Adaptive Pre-emphasis" circuit of the EMT 266 further processes this limited signal via a variable time constant filter. This controls high frequency signals which would otherwise cause overmodulation due to pre-emphasis in the ensuing signal chain. The combination of these principles makes the EMT 266 ideal for applications where overmodulation due to signal transients is a troublesome problem. The EMT 266 is rack mountable unit intended for 'hands off' operation.



GOTHAM AUDIO CORPORATION
741 WASHINGTON STREET
NEW YORK, NY 10014
(212) 741-7411
— or —
1710 N. LA BREA AVENUE
HOLLYWOOD, CA 90046
(213) 874-4444

manufacturer suggests direct contact

NEW SHURE PROFESSIONAL QUALITY MICROPHONES AVAILABLE WITH EBONY OR TAN 'SUEDE-COAT' FINISHES

A new, decorative dimension has been introduced to the audio scene by Shure Brothers, Incorporated, Evanston, Illinois, in the form of two new Starmaker™ microphones with attractive ebony or tan "Suede-Coat" finishes.

Designated the SM-77 and SM-78, both are professional quality dynamic microphones, designed for hand-held or stand-mounted vocal or instrumental use. Their sophisticated appearance and comfortable handling qualitites are features entertainers will find especially well suited to a wide variety of live and televised performance situations.

Both units are lightweight (a special bonus to vocalists) and extremely rugged. Their "Suede-Coat" finish is durable and easy to clean. Even consistent in-out use in a microphone stand will not mar the finish.

The cardioid pickup patterns of both microphones effectively reject feedback and make them suitable for sound reinforcement applications where room acoutics or ambient noise is a problem. When used for closeup vocals, wind and pop protection are provided in the SM-78 by a build-in filter. An accessory windscreen is available to serve this purpose on the SM-77.

The SM-77 and SM-78 have a frequency response of 50 to 15,000 Hz. Each model is available with either an ebony or tan finish and can be purchased with or without cable.

User net price of SM-77EB (ebony) and SM-77TN (tan) without cable is $117.00; with cable (SM-77EB-CN and SM-77TN-CN)

$138.00.

User net price of the SM-78EB (ebony) and SM-78TN (tan) without cable is $150.00; with cable (SM-78EB-CN and SM-78TN-CN) is $171.00.

SHURE BROTHERS, INC.
222 HARTREY AVENUE
EVANSTON, IL 60204
(312) 679-4020

for additional information circle no. 112

"HEADROOM HORSEMAN" INTRODUCED BY SLEEPY HOLLOW

According to Ted Hammond, cheif design engineer for Sleepy Hollow, "the *Headroom Horseman* fills a void that every professional studio engineer and all semi-pro studio personnel have been aware of for some time. In typical studio applications, the semi-pro/hi-fi levels of -10 are rarely brought up to the +4 level standard. This results in an inability to make adequate A/Bs of cassette and 1/4 track copies. In production applications, the -10 equipment will not properly drive a 600 Ohm load." Hammond continues, "we have taken a basic operational studio problem and found a very simple solution with *Headroom Horseman*. We even went so far as to pad the inputs of the hi-fi gear so they are driven with the levels they were designed to be interfaced with."

Deliveries will begin in mid-October. Price with the 50 volt power supply is $175.

SLEEPY HOLLOW
141 CROTON AVENUE
OSSINING, NY 10562
(914) 762-3197

for additional information circle no. 113

HUTCO TO INTRODUCE MIDORI FADERS AT NEW YORK AES CONVENTION

A new professional audio fader manufactured by Midori Precisions Company, Ltd., of Tokyo, Japan, claimed to be the highest performance fader available anywhere in the world, will be exhibited at the Audio Engineering Society Convention in New York by Hutco, Inc., of Huntsville, Alabama.

According to Hutco executive Tip Turpin, the Midori fader is a new product in the manufacturer's green pot series of conductive plastic potentiometers. Available in five difference standard models with an impedance choice of 600, 5,000, or 10,000

ohms, the MFP-1000 series faders are designed to be incorporated into audio and video switchers, panels, and other professional studio equipment. MFP-1000 faders are also available in custom versions to suit the customer's requirement for impedance range.

In general outline and form, Midori MFP-1000 series faders resemble products manufactured by Penny & Giles of the United Kingdom. Turpin says, however, that internal structural differences make it possible for the Midori product to specify lower insertion losses, lower noise, better linearity (or in the case of logarithmic faders, better tracking), and measurably better matching accuracy than any other fader on the market.

Information regarding end user and OEM

purchases of Midori faders, as well as sales representation and distributor agreements may be obtained from Hutco, exclusive U.S. agent for Midori Precisions.

HUTCO, INC.
2913 GOVERNORS DRIVE
HUNTSVILLE, AL 35805
(205) 533-9232

for additional information circle no. 115

NEPTUNE REAL TIME ANALYZER

NEI's (Neptune Electronics, Inc.) 2709A Real Time Analyzer is an improved tool for the professional sound engineer, recordist or musician. Designed to be more than an effective tool used in conjunction with 1/3-octave or parametric equalizers for flattening room response and eliminating feedback, the 2709A can be used to check response in every part of a sound reinforcement, recording or stage instrument system. The 2709A is designed to employ the band's or system operator's microphones, loudspeakers, and equipment instead of just the standard "calibrated" microphone or line source. The true response of any system being checked is displayed on a LED matrix consisting of 27 1/3-octave bands with 9 steps of amplitude. Marked by high quality electronics and superior design, the 2709A comes with internal pink noise generator with self-checking response, balanced input and line in/out jacks with gain control (D3F and ¼" phone connectors), sensitivity control with switchable range (3 or 1 dB) and more.

For installation in a road/flight case or equipment rack, the 2709A comes in a fully rack-mountable EIA package. Optional 220 V, 50 Hz line voltage available.

The 2709A is available for immediate delivery. See an authorized NEI dealer or factory representative.

NEPTUNE ELECTRONICS, INC.
934 N.E. 25TH AVENUE
PORTLAND, OR 97232
(503) 232-4445

for additional information circle no. 118

ALTEC LANSING INTRODUCES NEW AUTOMATIC MICROPHONE MIXERS

Altec Lansing has announced the addition of the new Model 1674 and 1678 Automatic Microphone Mixers to its Industrial/Professional line, according to Gary Rilling, national sales manager.

The 1674 (4-input) and 1678 (8-input) incorporate Altec's patented gain sharing principle which allows the system to deliver maximum acoustic gain while helping to prevent feedback in multi-microphone operations.

"Analog computer circuits look at the level of each input channel, compare that level to the total of all inputs and adjust the gain of each input in a manner which holds the overall mixer gain constant," said Larry Lutz, product manager, Commercial Products. "Plus, the 1674 and 1678 are designed to compensate accurately for the difference between coherent and non-coherent signals, thus avoiding potential mixing errors."

In addition to their automatic mixing capabilities, the Models 1674 and 1678 provide a number of other versatile application features including: balanced mike or line level inputs with phantom power for condenser microphones, TTL compatible logic outputs for custom applications such as automatic switching of speaker zones, channel line outputs for logging tape

recorders, remote muting and priority override control, switchable 200 Hz hi-pass filters and auto/direct bypassing in each channel and complete international selection of power standards.

The 1674 may be used as an extension of the Model 1678 or separately for systems requiring four channels or less. Up to forty channels of automatic mix may be achieved through appropriate linking of mixers.

The Altec Lansing Model 1674 and 1678 Automatic Microphone Mixers offer performance and flexibility to meet the demanding needs of the audio industry.

ALTEC LANSING
1515 S. MANCHESTER BLVD.
ANAHEIM, CA 92803
(714) 774-2900

for additional information circle no. 119

NEW RUSLANG CONSOLE OFFERS MORE RACK SPACE

Studios seeking additional rack space are said to find it in Ruslang's newest tape transport console . . . the RL 800. This console, which accommodates all standard 19" x 15¾" tape decks, featrures 7" of conveniently angled rack space positioned immediately below the deck.

There is an additional 14" of rack space in the lower front portion of the console. Add an optional overbridge rack and you can more than double that figure. No other console on the market offers so much rack space.

Other features include Ruslang's exclusive "hinge-up" of the tape transport for convenient servicing of the underside. An optional back panel is attached with Velcro, eliminating the need for screws or screwdriver. And, the RL 800 comes with casters for easy maneuvering.

The RL 800 joins a distinguished console line that embodies the industry's most modern design features. The entire line is assembled with the same craftsmanship, style and quality that the company builds into Scully consoles. (Ruslang has been a major supplier of Scully consoles for 15 years.)

For details and pricing on the entire line of tape transport consoles, contact:

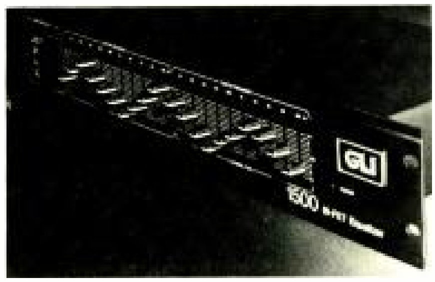
RUSLANG CORPORATION
247 ASH STREET
BRIDGEPORT, CT 06605
(203) 384-1266

for additional information circle no. 121

GLi INTRODUCES EQ-1500 GRAPHIC EQUALIZER

GLi has introduced a 10-band EQ-1500 Graphic Equalizer. In addition to providing state-of-the-art gyrator and low-noise Bi-FET circuitries, the unit offers a high slew rate and includes three sets of slectable inputs complete with tape monitor function. There is a circuit bypass mode with LED status as well as one octave-wide filters with center detent for easy flat response location. The EQ-1500 has a range of ±12 dB and provides a high output capacity of 10 V before clipping. A relay circuit mutes the audio output

New Products



preventing turn on/off transients from reaching amplifiers and speakers. Designed for the professional, the EQ-1500 mounts in a standard 19-inch rack.

GLi, designer and manufacturer of speaker systems and related audio electronic components for professional use, is a division of Integrated Sound Systems, Inc. For more information, contact:

INTEGRATED SOUND SYSTEMS, INC.
29-50 NORTHERN BOULEVARD
LONG ISLAND CITY, NY 11101
(212) 729-8400

for additional information circle no. 122

DISCO SPEAKER SYSTEMS NOW AVAILABLE FROM ELECTRO-VOICE

A three-piece disco speaker system consisting of a low frequency cabinet, a midrange/high frequency cabinet and an electronic crossover/equalizer is now available from Electro-Voice. "We use the term 'disco' because it's a well understood term," said Dave Rothfield, E-V's general sales manager, "but in reality, this is a professional system for the dance environment. We make that distinction because this system superbly reproduces live music in addition to recorded music.

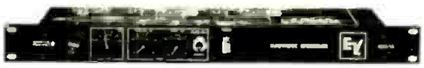
"Two basic objectives of a high performance discotheque-type sound system are to get the lows down to the floor where you can feel them, and the highs up where you can hear them. The LF118 sub-woofer and the HF12-3 high frequency module do just that. When combined with the associated XEQ-1A crossover/equalizer this system becomes the hottest, most accurate, reliable and convenient packaged system available today."

The HF12-3 features an EVM™ 12L woofer, E-V's exclusive VMR™ vented midrange speaker and the famous ST350A tweeter. The wide 120° dispersion, uniform over the entire frequency range, exsures that

everyone on the dance floor is treated to the same high quality sound. The unique trapezoidal shape and short vertical dimension of the HF12-3 make it simple to install, even where low ceilings would hinder the installation of other high frequency arrays.

The LF118 is intended to be floor mounted where it will produce "visceral," undistorted bass down to 40 HZ. The LF118 can be "stepped-down" with the XEQ-1A crossover to provide extended bass response down to 28 Hz. This is ideal for live music, where synthesizer notes down to 32 Hz are not uncommon and for recorded music when subharmonic synthesizers are employed for added impact. "Although we normally recommend one LF118 to each stereo pair of HF12-3s, in the "step-down" mode we recommend two LF118s," added Rothfeld.

Protecting the woofer from damaging subsonic information generated by record surface irregularities is an important bebefit offered by the new XEQ-1A crossover/equalizer. This is accomplished by an integral high-pass filter. Crossover frequencies are determined by plug-in modules, with the recommended crossover of this system being 125 Hz. The build-in switchable Thiele equalizer extends the response of the LF118 to 28 Hz. One XEQ-1A is required for each stereo channel.



"Until now," concluded Rothfeld, "high quality sound systems have been the exclusive province of those large clubs that could afford the services of an expert sound design and installation team. We've changed this. We've taken our many years' experience in working with these custom installers and put together a sound system package that will blow you right off the dance floor."

DANCE ENVIRONMENT SYSTEMS
ELECTRO-VOICE, INC.
600 CECIL STREET
BUCHANAN, MI 49107
(616) 695-6831

for additional information circle no. 124

SONY INTRODUCES ONLY DIGITAL REVERBERATOR WITH NON-VOLATILE MEMORY AND DIGITAL/ANALOG COMPATABILITY

Sony has introduced the remarkably advanced DRE-2000 at the 67th Audio Engineering Society Covnention. The digital reverberator features a ten-program memory, convenient hand-held controls, four reverberation modes, two for each, two for delay, as well as the unique non-volatile memory and the unprecedented flexibility of direct interface with both analog and digital systems.

Utilizing an advanced microcomputer as its brain, the DRE-200 offers the obvious advantages of freedom from external vibration problems under any operating conditions. The four reverberation modes provide not only greater program choices but also the widest range of basic reverb time, pre-delay time, level, and delay time of both early reflection and sub-reverb, and frequency response of outgoing signal. The variable parameters for the echo modes include feedback factor, delay interval, and frequency response of outgoing signal, and those for the delay modes are delay time and frequency response of outgoing signal.

The main unit is rack-mounted and controlled by a hand-held 10-key board. Up to ten programmed combinations of variable parameters can be stored in the non-volatile user memory for repeated use. The non-volatile feature protects against loss of program from accidental power failure or cable disconnection. After switching the unit on, immediate operation is possible by either the pre-set programs for each mode or by the programs stored in the user memory. Instantaneous change of any one of the parameters in the program then running is possible, as well as changes from program-to-program while operating.

To meet a variety of professional applications, the DRE-2000 incorporates

— *continued overleaf* . . .

for additional information circle no. 125

New Products

provisions for both analog and digital system connections. With the unique built-in A/D and D/A converters, the unit functions as a high performance reverberator with analog mixing consoles, as well as allowing direct connection to digital systems.

Reverberation time range of the DRE-2000 is 0 to 9.9 seconds, while delay time varies from 9 to 999 milliseconds. The flat 10 to 15,000 Hz frequency response can be varied by the use of variable low-cut and high-cut filters. The unit provides processing to 16-bit quantized digital signals without affecting the high quality of the original signals. Sub-reverb and non-linear reverberation insure bright, wide, and deep stereo imaging and ambience effect.

The DRE-2000 measures only 5-3/16" x 16¾" x 8-1/6", is easily rack mounted, and complete with hand-held control weighs only 28.8 pounds. Price: $15,000.

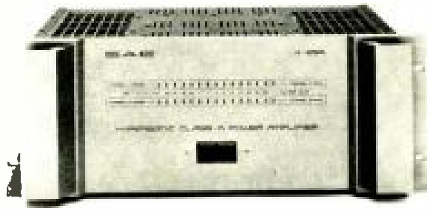
SONY CORPORATION OF AMERICA
9 WEST 57 STREET
NEW YORK, NY 10019
(212) 371-5800

for additional information circle no. 126

CLASS A AMPLIFIER FROM SAE

Identified as the X25A Hypersonic Class A power amplifier this 250 watt-per-channel (minimum RMS per channel into 8 ohms with less than 0.02% THD) amplifier is the first of a series of power amplifiers to incorporate the patented Class A techinque.

According to the company, unique circuitry has allowed the construction of an amplifier which realizes the low-distortion benefits of a conventional Class A amplifier (using the classic design approach) while generating only a fraction of the amount of heat usually resulting from such a design. Because an "X-series" amplifier loses such a low percentage of its power to heat, it does not need to be as large or as costly as other Class A amplifiers.



The X25A incorporates the complementary-symmetry circuit design associated with SAE. Each channel of the X25A employs two independent amplifiers, one which handles the positive slope of the waveform and one which handles the negative slope. Since both amplifiers are operating at all times, any non-linearity in the output of either device is minimized without the need for feedback correction.

Also included in the design of the X25A is linear open loop bandwidth extending beyond the range of audibility. Therefore, feedback is not necessary to control frequency or signal level. It is used only to control gain.

The X25A features a separate 15-LED power level indicator for each channel. This display senses both voltage and current, thus displaying true power output of the amplifier. Unlike other SAE components which are made of black anodized aluminum, the "X-series" are constructed of a specially developed gray anodized aluminum.

Retail price: $1,500.

SCIENTIFIC AUDIO ELECTRONICS
(SAE)
701 MACY STREETLOS ANGELES,
CA 90012
(213) 489-7600

for additional information circle no. 127

STUDER A80RC 1/2 INCH 2-TRACK MACHINE

Desgined for top quality mastering, Studer has announced a half-inch version of the A80RC. This machine is equipped with newly developed record and playback heads. For For maximum erasure, a dual gap Ferrite full track head is used.

Signal-to-Noise ratio is specified at 75 dB at a reference flux of 1040 nWb/m. All other specifications remain unchanged.

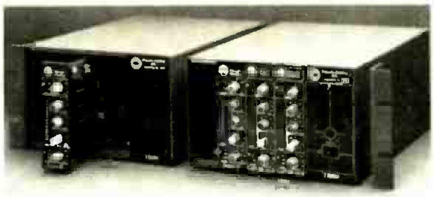
STUDER ReVOX AMERICA, INC.
1425 ELM HILL PIKE
NASHVILLE, TN 37210
(615) 254-5651

for additional information circle no. 128

VALLEY PEOPLE CLAIMS "INTELLIGENT" LIMITER/ COMPRESSOR/DUCKER

As its first product development since the August 1980 merger with Allison Research and Valley Audio, Valley People announces Gain Brain II, a Limiter/Compressor/Ducker. The new design culminates extensive research into the psycho-acoustic relationship of waveform complexity to desirable levels of perceived loudness.

Proprietary waveform recognition circuitry has resulted, which allows the device to comprehend the intent of the performer, in terms of optimum output loudness for each note, syllable or accent. The effect claimed, is a significant improvement in the preservation of dynamic integrity during limiting, and less "limiter sound" (pumping, etc.), than for the



conventional peak or RMS responsive devices.

Paradoxically, in spite of a perceived increase in program dynamics, Gain Brain II is stated to offer a flatter VU meter output than any competitive product.

A full complement of front panel controls allows optimum settings for a variety of effects other than normal limit/compress functions. As an Interactive Gain Control Device (Ducker), settings may be made such that the presence of one program source (such as an announcer) may be made to cause a reliable gain reduction of between 0 dB and 48 dB in a second program source (such as background music). Other effects such as controlled Impact Accentuation on percussive tracks can give the effect of expansion, even though the device is actually limiting . . . thus controlling the maximum excursion.

Provisions have been made for stereo intercoupling, side chain operation for frequency dependent gain control, and for remote VCA and/or remote GR metering. Superior noise, distortion, and slew rate are realized as a result of the employment of the Valley People EGC 101 VCA, as a feed forward gain control element.

Gain Brain II is designed to mount in the TR-804 processing package, which is adaptable for either portable applications, or for rack moun multiple units.

In single units, Gain Brain II is priced at $380.00.

VALLEY PEOPLE, INC.
2820 ERICA PLACE
P. O. BOX 40306
NASHVILLE, TN 37204
(615) 383-4737

for additional information circle no. 130

news

Continued from page 30 . . .

symphonic suite from the original motion picture score, is being released as a Digital dbx Disc. The Soundstream digital recording system was used to make this recording. With encoding in the dbx format, the sonics performance is said to be virtually indistinguishable from the digital master tape.

Tim Weisberg's critically acclaimed album, *The Tip of the Weisberg,"* is being released as a Digital dbx Disc according to Nautilus Recordings president Steve Krauss.

"The Weisberg album was made with the Soundstream digital recording process — the finest recording technology available," said Mr. Krauss. "Combined with dbx encoding, the sound is absolutely astounding. I hope that someday every record company will adopt it.

"My belief in the dbx format goes back to 1976 when we produced our first record, *First In Line.* We'd released a limited edition dbx encoded tape that was sold at $75 to hi fi stores for demonstrations of equipment. When we did direct-to-disc, we made a dbx version simultaneously. It was beautiful! I just couldn't believe that kind of sound quality could be obtainable from a disc," added Mr. Krauss.

3M AND NORTH AMERICAN PHILIPS ANNOUNCE VIDEODISC AGREEMENT

Frank L. Randall, Jr., vice chairman of North American Philips Corporation, and Daniel E. Denham, Jr., vice president,

"WELCOME BACK WALLY"

More than 800 members of the music industry turned out to welcome back recording studio pioneer, Wally Heider. A lavish welcome back Wally party was held at Heider's Hollywood facilities featuring the playing of Les Brown and his "Band of Renown." Los Angeles Mayor Tom Bradley declared September 16 as official Wally Heider Day.

Wally Heider (right) with long time friend, bandleader, Artie Shaw

Recording Material Group/3M, have jointly announced an agreement for 3M to manufacture reflective optical videodiscs.

John C. Messerschmitt, vice president of North American Philips, and John E. Povolny, vice president of the Magnetic Audio/Video division of 3M, said that production quantities of 3M replicated discs for laser-based reflective videodisc players are anticipated in 1981. Povolny said Lloyd A. Troeltzsch, manager of 3M's Optical Recording Project, will be responsible for 3M's videodisc business.

Under the agreement, North American Phillips is providing disc-mastering equipment to 3M and the two firms are exchanging patent license agreements on selective optical videodisc technologies.

Mr. Messerschmitt, who has overall program management and coordination responsibility for all aspects of North American Philips videodisc activities, said 3M's commitment to replicate the optical videodiscs will result in both greater quantities of discs available for the system and a greater variety of programs.

"This agreement," he said, "means there now will be two major producers of optical videodiscs — DiscoVision Associates, the present supplier, and 3M, a company with a history of expertise in the recording products field."

Mr. Troeltzsch said 3M's decision to replicate the reflective videodiscs is a recognition of the versatility of that system and its emergence as one of the standard videodisc systems. He noted that 3M earlier agreed with Thompson-CSF to replicate transmissive optical videodiscs.

He said 3M has no plan to acquire disc programming but instead will act as a contractor for mastering and replicating videodiscs for program suppliers — in quantities for both consumer releases as well as short-run industrial and educational programs.

3M's Optical Recording Project is a part of the firm's Recording Materials Group which manufacturers a wide range of products including magnetic tapes, videocassettes, digital audio recording systems and data recording products.

LITERATURE FOR THE MAGNETIC RECORDING INDUSTRY OFFERED

R. K. Morrison Co., Kensington, California, is offering a new, four page brochure covering magnetic tape reproducer calibrators, service aids, and illustrative materials tailored to the needs of the magnetic recording industry. This informative brochure includes prices and is offered without charge to interested individuals and companies by writing direct to R. K. Morrison Co., 819 Coventry Road, Kensington, CA 94707.

FAX AUDIO FORMED

Michael Consi, former director of engineering at Automated Processes, Inc., (API), together with several other former API engineers, has announced the establishment of a new professional audio company, Fax Audio, Inc. According to Mr. Consi, "Fax will design and produce a complete line of state-of-the-art products and systems geared for the recording and broadcast industries."

Fax has already introduced a highly innovative programmable fader featuring touch switches, a sophisticated array of visual displays, and a wider range of functions than previously available in modular automated faders. The company has also developed a discrete op-amp which features the finest specifications and performance criteria of any op-amp on the market and can be used to upgrade any API Model 2520 circuit. The op-amp is also employed in a complete series of plug-in amplifier cards.

Future Fax products will include a line of competitively priced state-of-the-art recording and broadcast consoles which are currently under development.

For more information contact:

FAX AUDIO, INC.
29 ANITA PLACE
AMITYVILLE, NY 11701
(516) 589-1258 or
(516) 261-6085

RELAPPING SERVICE OFFERED BY JRF

JRF Company recently began offering a highly specialized relapping service for audio magnetic heads. The new service is made possible by the firm's completely equipped and staffed relapping facility. It enables JRF to provide fast, complete and professional servicing for a wide variety of heads used throughout the industry.

According to John R. French, Jr., the founder of JRF, "Every magnetic head receives a comprehensive optical and electrical inspection prior to relapping which includes gap condition, depth of wear, inductance and DC resistance. This is followed by our precision relapping and polishing process. A final inspection and lab report conclude the relapping procedure. If a head should fail its final inspection, it will be returned to the customer with no charge for relapping. That's how strongly I believe in our program."

A newly published brochure provides a detailed description of the firm's procedure.

JRF COMPANY
17 BYRAM BAY ROAD
HOPATCONG, NJ 07843
(201) 398-7426

ALLEN & HEATH BRENELL APPOINTS INTERNATIONAL AGENTS

AHB is pleased to announce the appointment of the Otari Electric Company, Ltd., - Japan, Industrial Products Division - Import, as official agents for the AHB range of professional audio products throughout Japan. Otari Electric is located in the Otari Building, 4-29-18 Minami Ogikubo, Suginamiku, Tokyo 167, Japan.

For the west coast of the United States, AHB has announced the appointment of ACI Filmways - U.S.A., 7138 Santa Monica Boulevard, Hollywood, California 90046 as agents.

— *continued overleaf . . .*

news

In Canada, AHB has announced the appointment of White Electronics - Canada, located at 6300 Northam Drive, Malton, Ontario L4V 1H7, Canada.

SONY DIGITAL RECORDING, EDITING USED BY COLUMBIA AND METROPOLITAN OPERA

A new Columbia album with artist Kenny Loggins, utilizing Sony digital recording and editing equipment, has just been released. "Kenny Loggins Alive," a double LP, includes eighteen songs (three of which are new) and was recorded live in Sacramento, Chico, Los Angeles, Lake Tahoe, and Atlanta. Bruce Botnick, executive producer for Columbia Records, handled production of the album.

Mr. Botnick, a pioneer in recording, with the first Dolby in rock 'n' roll (with the group "The Doors") to his credit, said that his choice to use Sony digital stemmed from his use of the same equipment on the "Star Trek" soundtrack album.

The mixdown of the album on the Sony PCM-1600 was completed at Ocean Way Studios with engineer Rick Pekkonen, and then edited at Digital Magnetic using the Sony DEC-1000 prototype editor.

Mr. Botnick commented, "The editor allowed us creativity we didn't have before. I know we made edits that we couldn't have done with a razor blade. In order to make the crossfades, we used two Sony PCM systems in sync, so we were literally running a 4-track didital system."

The Loggins LP was mastered at Precision Lacquer with engineer Arnie Acosta.

"By making a digital cutting master and patching it straight into the cutting head amplifier, Arnie was able to bypass all the electronics in the console and make a perfectly matched disc every time," Mr. Botnick recounted.

The impact of digital audio recording is being increasingly felt throughout the music world. One of the most positive signs of its acceptance has been the recording using digital equipment of the Metropolitan Opera's "Live From The Met" telecasts during this past season.

Over the past 40 years, the Met has recorded its Saturday afternoon radio broadcasts and maintained audio tapes of all its opera productions. The Met initiated digital recording this past year, inspired by the ability to accurately record some of its most important productions with the finest stars in the opera world. According to the Met's telecast producer, Clemente D'Alessio, digital recording offered the advantage of capturing the best available sound quality with the technology that is slated to become the standard in the near future. Mr. D'Alessio added that he is looking forward to the potential inherent in multitrack digital recording.

Recording the Met performances with Sony's PCM-1600 digital recorder represents a unique opportunity, according to Roger Pryor, manager for Sony's Professional Audio Division. "The simultaneous videotapings and digital recordings ensure

PETER JENSEN of DIGITAL RECORDING SYSTEMS operates the Sony PCM-1600 digital processor, recording a performance of the Metropolitan Opera. Among the performances already recorded are *Othello*, *Elektra*, *Don Carlo*, *Un Ballo in Maschera* and *Manon Lescaut*.

that the Met can build a digital/video library which will be totally compatible with future playback formats."

Digital recordings for the Met's archives were made by Digital Recording Systems Co., Inc., based in Elkins Park, Pennsylvania. According to DRS's Peter Jensen, the engineer who executed the Met's digital recordings, his company began on-location digital recording a year ago with the purchase of the PCM-1600. In its first year, it has recorded for several major labels, including RCA, CBS, and Vanguard.

Referring specifically to the Met assignment, which has involved the recording of *Othello*, *Elektra*, *Don Carlo*, *Un Ballo in Maschera*, and *Manon Lescaut*, Mr. Jensen remarked, "Sony digital audio recording has enabled the combination of uncompromising recording clarity with the exceptional ability to record long performances easily using economical cassettes.

"The Sony equipment has proved ideal on the Met assignment," he concluded, "because it is completely portable and compatible with the Met's video equipment, which is used for simulcasts and the preparation of video discs."

CLEAR-COM APPOINTS:

Peter Giddings as international sales manager. Giddings is co-founder of Beyer Dynamic/England, former VP of marketing for ReVox Corporation, New York, Hollywood, Canada, and former director of marketing for Hammond Industries/England.

Patrick Hayes as operations manager. He will apply his expertise to manufacturing and company expansion while maintaining Clear-Com's high level of customer service and product availability.

12TH TONMEISTERTAGUNG SCHEDULED FOR NOVEMBER 25 - 28, 1981

Announced to take place in the congress building of the Deutsches Museum, the convention will be organized by the Verband Deutscher Tonmeister (VDT), in cooperation with the "Bayerischer Rundfunk," the "Bavarian Atelier Gesellschaft" and the "Westdeutscher Rundfunk." It will be promoted by the "Hochschule der Kuenste Berlin" and the "Hochschule fuer Musik Detmold."

The Tonmeistertagung, which is held

every three years, addresses itself to the experts of professional sound work in the fields of the recording industry, broadcasting, television, film, and theater and thereby offers the possibility to exchange experience between practical users, constructors, manufacturers, and representatives of the relevant sciences.

The lecture program of the convention will be accompanied by an extensive exhibition of studio equipment and completed by numerous excursions to production studios in the Munich area.

Based on the experiences by the Tonmeistertagung 1978 held in Berlin, more than 2,000 participants and over 100 exhibitors from all over the world are being expected to attend.

QUANTUM AUDIO LABS ACQUIRES AUDIO LOGIC

Quantum Audio Labs, Inc., has acquired Audio Logic — a spin-off of Uni-Sync, Inc., formerly located in Westlake, California. Audio Logic's operation will be consolidated into the Quantum Audio Labs' factory facility in Glendale, California.

According to John Pritchett, president of Quantum Audio Labs, sales and service for both product lines will be handled out of the Glendale facility — with Audio Logic operating as a wholly owned subsidiary of Quantum Audio Labs, Inc. Audio Logic's product line includes the Discorama professional disco mixing console and specialized signal processing and monitoring equipment and will include expansion into advanced live music mixing systems.

Andrew C. Thompson, former president of Audio Logic, has been named national sales manager for Quantum Audio Labs, Inc., and Audio Logic.

QUANTUM AUDIO LABS, INC.
1909 RIVERSIDE DRIVE
GLENDALE, CA 91201
(213) 841-0970

MARTIN AUDIO ANNOUNCES SPENCER, V.P., OTHER APPOINTMENTS

Bruce Martin and Norman Kassel, principals of the Martin Audio Video Corp., New York, are proud to announce the appointment of Courtney Spencer to the position of vice president. Mr. Spencer is currently the general nanager at Martin. He will retain this position as he assumes his new executive capacity.

Mark Friedman has joined the staff of Martin Audio and Video Corp., where he will be in charge of field sales. Before joining Martin Audio, Friedman was studio manager and chief engineer at Producers Recording, New York, and staff engineer at RPM Sound Studios, also New York. In addition, he has several years of experience as a professional musician.

Bob Quinones has been appointed manager of the Parts Department at Martin Audio Video Corp.

GOLDSTEIN REASSUMES ORBAN MARKETING RESPONSIBILITY

Orban Associates, Inc., has appointed Sid Goldstein as marketing and sales manager for Professional Audio Products.

Goldstein was formerly with Parasound, Inc., a San Francisco-based independent marketing consulting firm. This firm had acted as Orban's worldwide marketing and sales agent until early 1978 when an internal marketing organization was installed.

John Delantoni, Orban's general manager, has been performing this function most recently.

. . . News continued on page 176 —

BOOKS

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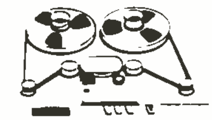

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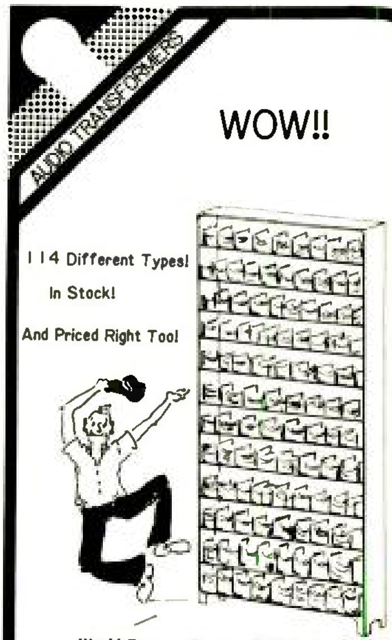

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— continued from page 173

news

ROBERT CARR JOINS SESCOM

Robert W. Carr has joined SESCOM, Inc., in Las Vegas, Nevada, as vice president of marketing. Carr was formerly with Shure Brothers, Inc.

JBL CONTINUES ITS SUPPORT OF ASPEN AUDIO RECORDING INSTITUTE

James B. Lansing Sound, Inc., this year continues to support the Aspen Music Festival's Audio Institute, a "hands-on" approach to teaching live recording techniques which completed its third operating season August 24. JBL vice president for market planning, John Eargle, past president of the Audio Engineer Society and author of *Sound Recording*, visited the school to lecture each of its three consecutively-run sessions held throughout July and August. In addition, the company donated several hundred Aspen Music Festival/JBL T-shirts to the school for resale among students and concertgoers.

In 1978 and 1979, JBL made substantial donations of studio monitor, power amplification and sound reinforcement equipment to the Audio Institute. Other companies which have supported the program include TEAC and Ampex.

John Denver monitoring a take with Aspen associate director Alan Kefauver . . .

On August 19, Audio Institute faculty and students participated in a special John Denver benefit concert for the Festival which netted $50,000. Performed along with the Aspen Festival Orchestra in the school's official concert tent, the Denver benefit was filmed by ABC-TV and recorded by the Institute for broadcast as a televison special this spring.

THE AMERICAN UNIVERSITY DEPARTMENT OF PYHSICS ANNOUNCES A PROGRAM IN AUDIO TECHNIQUES

The newly announced program in Audio Techniques will concentrate on the technology of electronic recording and reproduction of sound.

Audio Techniques is a program setup by four divisions — the Department of Physics, the School of Communications, the Department of Performing Arts, and the University Audio-Visual Services. Faculty, staff, and the technical facilities of the four divisions enrich the program and broaden its perspective.

This four year program, which leads to a Bachelor of Science degree in Audio Techniques, includes courses in communications, music, the performing arts, physics with application to sound and two semesters of laboratory; two semesters of basic mathematical methods, two semesters of music; three semesters of communications; and three semesters of theater techniques. There are also recommended courses in business and law, depending upon the particular career objectives of the individual student. The final two semesters are devoted in part to internships at active professional recording studios, radio and television stations and theaters.

Inquiries should be directed to Dr. Romeo Segnan, Department of Physics, McKinley Building, Room 106, The American University, Washington, DC 20016.

UNIVERSITY OF MIAMI STUDENT ENGINEERS WIN *down beat* CONTEST

University of Miami jazz musicians and recording engineering students swept three top prizes and five outstanding performance awards in the 1980 *down beat* magazine's Third Annual Student Recording Awards national competition.

A staff of ten judges — composers, arrangers, musicians, publishers, authors and critics — picked them from among hundred of entrants from the U.S. and Canada.

Students enrolled in the music engineering program directed by Bill Porter made a clean sweep of the 1980 "deebee" awards for their live and studio recording projects. Les Brockmann and Larry Revit co-engineered a session that took first place honors in the Best Engineered Live Recording for "Mixdurata," a performance by a student sextet. Said the judges, "Everything is at pro level, exceptionally good recording of relatively difficult material." Mike Sak and Christopher Jacks were awarded an Outstanding Performance Award in the same division for their recording of a performance by a student octet.

To take top honors in the Best Engineered Studio Recording category Christopher Jacks, Paul Hugo, Pierre Porter, and Mike Sak combined their engineering talents to record an original composition by Jacks using such unusual instruments as a Sears radial-arm saw, the base of a dentist's chair, and shoe strings in addition to more conventional musical instruments. For their efforts the judges commented, "Completely professional, a winner at any level."

The UM Fusion Ensemble won the coveted "deebee" award for Best Jazz Performance by a Group in the college division. Members of the ensemble include John Lovell, trumpet; Bill Ross, alto sax;

Randy Russell, tenor sax; Dave Grygier, trombone; Jeff Campbell, guitar; Reed Arvin, piano; John DiModica, bass; Rob Echelman, drums; and Ron Miller, conductor and faculty advisor. Comments of the judges included, "Out-of-sight group, soloist excellent but alto player is superb . . . top arrangements, I'd pay to hear them live."

The Mike Moore Quintet with Mike Moore, piano; Pete Moutis, drums; Larry Kornfeld, acoustic and electric bass; Bob Curtis, soprano sax, flute; Paul Bollenback, guitar; and Bob Gower, faculty advisor, captured an Outstanding Performance Award in the Best Jazz Performance by a Group category.

Pianist Reed Arvin won an Outstanding Performance Award for his jazz instrumental solo performance of Wood Dance (Ron Miller) with the Fusion Ensemble.

The nationally acclaimed UM Concert Jazz Band, under the direction of faculty advisor and conductor Whit Sidener, captured one of three Outstanding Performance Awards given for the Best Performance by a Big Band.

For individual effort, David Roitstein received an Outstanding Performance Award in the Best Original Composition category for his work, "Couquina."

The "deebee" categories and general judging criteria are patterned after the Grammy Awards which annually recognize exceptional achievements in the professional recording industry.

The UM's jazz and recording engineering programs are considered among the finest in American universities and colleges.

book report

THE PLATINUM RAINBOW

by

Bob Monaco & James Riordan

Most books about the music business are either biographical or encyclopedia-like and have little practical value when it comes to carving out a career. *The Platinum Rainbow*, by Grammy Award winning producer Bob Monaco and syndicated music columnist James Riordan, succeeds in offering a step-by-step guide to success in a variety of careers in the music industry.

While geared primarily to aspiring recording artists, the book also has valuable sections on becoming a successful producer, engineer, songwriter, agent, promoter, publisher, session musician, and manager. There is also a chapter devoted to the structure of a record label and what it takes to land any of the various jobs that exist there.

The book pulls no punches and the authors tell you up front that after reading it you may decide a career in music is not worth the hassle, but if you do decide to "go for it" they will tell you what you have to know and what you have to watch out for. The thrust of the book is teaching the reader how to "focus on reality in a business based on fantasy." One of the keys to the process is "really learning from your mistakes instead of just pretending that they didn't happen."

The Platinum Rainbow is more than readable. In fact it is a very funny book. Monaco and Riordan relate many humorous experiences and their witty descriptions keep even the most technical material from becoming dry. In discussing the difficulty of obtaining a big-time manager they caution the reader that he "must be able to impress someone who may be so jaded that if you could combine the seven wonders of the world into one stage show, they would say something like, 'Well, it's not bad, but of course, it needs a lot of work.'" While emphasizing that the odds against success are high, Monaco and Riordan claim that taking a realistic approach makes it attainable if you want to work hard enough for it.

The book quotes some of the biggest stars in the business including Frank Zappa, George Harrison, Lindsey Buckingham, Graham Nash, John Lennon, David Crosby, and others. Producers Ted Templeman, Arif Mardin, Ron Nevison, Johnny Sandlin, John Carter, and Paul Leka are also quoted as are Woodstock promoter John Morris and personal manager Irving Azoff.

The beginning of the book is devoted to exposing some myths about the music business. Example: "The Golden Reel-To-Reel and The Platinum Turntable Myth," which is the belief that people in positions of power in the music business are rarely wrong. The truth, according to the authors, is that many executives are "rarely right and they keep their jobs by effectively playing the percentages." The way to overcome this myth, according to Monaco and Riordan, is

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to "not allow yourself to be intimidated by fancy offices, gold records, power, prestige." You must be open to constructive criticism, but realize that you can't please everybody.

Chapter Two focuses on "getting serious" and includes sections on goal setting, realizing your limitations, kicking your brother out of the band, not advertising that you're an idiot, and other essentials to a professional approach. Chapter Three discusses the "Stage and Rehearsals" and how to make them work for you.

The song, being a songwriter, being a publisher, and other related sections to both the creative and business end of tune smything are covered in Chapter Four. Chapter Five reports on the demo and the all important "Getting A Deal."

After you've gotten your attitude, stage show, songs, and demo together it's time to go for the big time according to the authors and Chapter Six describes how to find the right producer, engineer, and studio. It includes special sections on becoming an engineer or producer which offer some rare tips on the practical aspects of both careers. The next chapter deals with preparing for the studio which includes a good section on becoming a session player. Chapter Eight does an effective job of breaking down the various components of a major recording session in a way that can be understood and applied.

Manager, agents, promoters, and attorneys are the subject of Chapter Nine which covers not only how to deal with them, but how to become one as well. Chapter Ten focuses on record labels and how to get a job at one. This chapter also reports on radio, critics, and the charts. "The Creative Businessman" is the thrust of Chapter Eleven which also includes a great section on "Getting Screwed."

The final chapter teaches the reader how to apply what he's learned, the importance of believing in himself, and overcoming his biggest enemies — "fear and greed."

The appendix includes a glossary and a comprehensive directory of record labels, managers, agents, promoters, engineering courses, producers, recording studios, publishers, and performance societies. This directory alone would normally sell for more than the book's ten dollar price tag.

The Platinum Rainbow teaches you how to give any career in the music business your best shot and it does so in a very enjoyable way. It contains a great deal of valuable practical information which very few other books include and it shows you how to use that information to your advantage. The idea is to win at the music business game and to enjoy yourself both while you're playing it and after you've succeeded at it. In the end, conclude Monaco and Riordan, "you've got to feel good about it all." *The Platinum Rainbow* even teaches you how to do that, too. It's the ultimate career book on the music business. ☐ ☐ ☐

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