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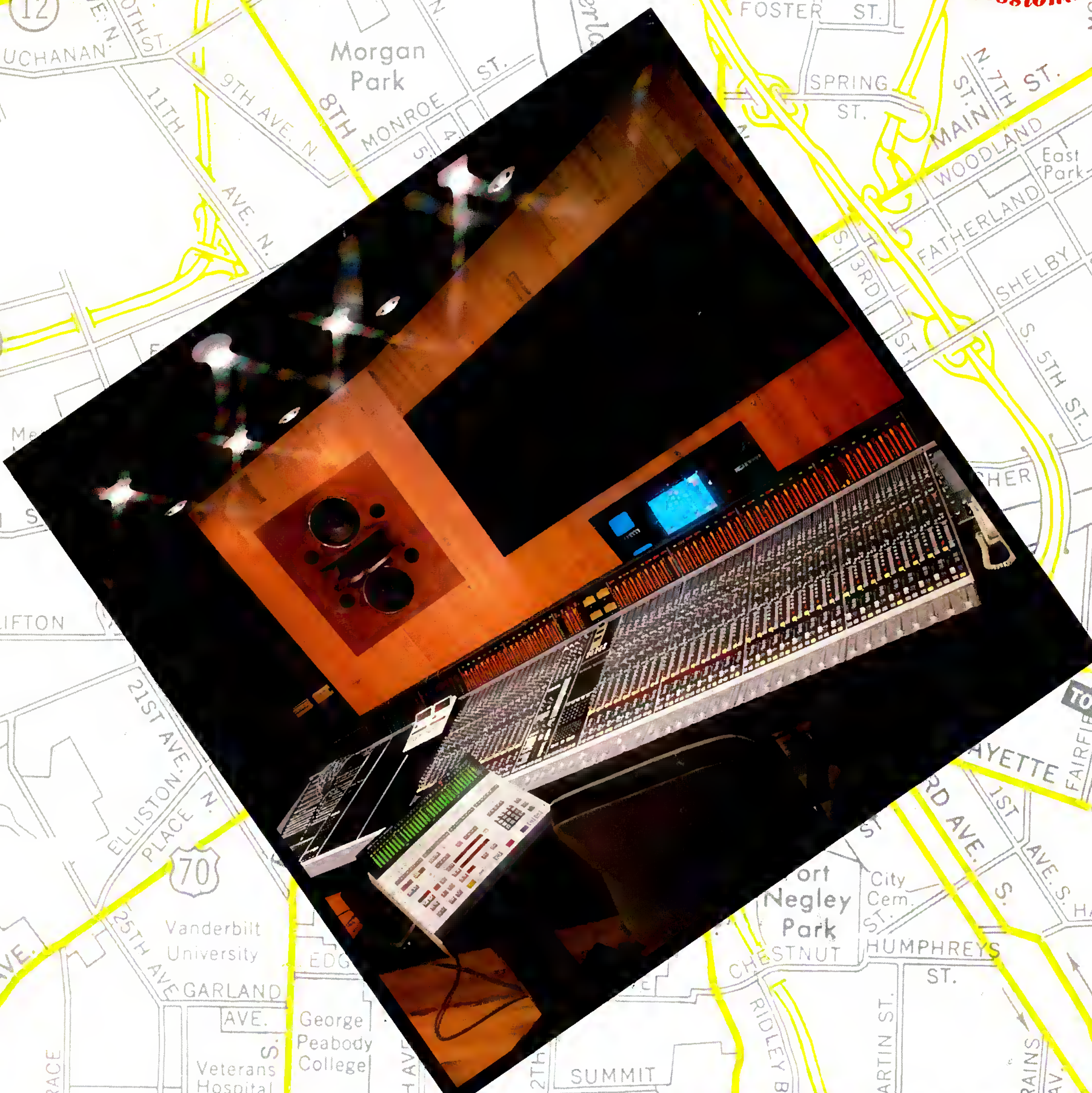
Guides: Consoles and Mixers



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July/August 1988 Volume 22 NO.4

The recording engineer

The sound contracting engineer

THE ELECTRONIC COTTAGE

About the Cover

● Masterfonics Studios with a map of Nashville, TN serves to introduce our feature story on this city. Beginning on page 4, Technical Editor Corey Davidson takes us on a tour of Nashville's recording excitement.

Editorial

When we went to Nashville, TN in 1980 to see what was going on, we found a still sleepy country-music town. Nashville was known then as the home of this special kind of music, but don't look for anything modern there! But, there was a stirring even then. Country music also needed quality recording. And Nashville provided it.

What has happened in those eight years? Seemingly the commercial world has found that there is something other than the New York City studio hectic pace, the Southern California laid-back studios all-over-the-place syndrome, and even the downtown Chicago studios scene.

Meanwhile, Nashville was reaching out—modernizing the famous studios on Music Row, while creating new ones as well. A shrinking world soon discovered that the two coasts and one Midwest location were being upstaged by a no longer sleepy southern town.

We'd been back to Nashville several times since that first visit, but still were not prepared for what this year's Nashville has to offer. When Technical Editor Corey Davidson went to Nashville earlier this year to find out what's going on, he found instead a vibrant mid-sized city with a eye toward the future, and a studio complex that ranges with the finest this audio world we are in has to offer.

Within a short drive from one end to the other, he found disk mastering, digital recording music studios and digital recording commercial studios.

Yes, it is still the capitol for country-music recording, but music, any music, cannot alone support a modern major studio system. Rather, it is the commercial post-production world that is footing the bill for the high-technology studios found today in Nashville and elsewhere. What is different today is that the commercial world itself has also found in Nashville a fine ambiance for efficient work.

Turn the pages to Corey's article and discover as we have that Nashville today is keeping up with the world, indeed, may well be leading it now.

I hear from major studio owners around the country how the music recording industry alone cannot support the investment needed for new consoles and new digital tape recorders. So, the word has to get out that there is a strong need for studios equipped to do more than music, indeed, able to do aspects of audio post-production work for television.

This is a wind of change that is blowing through the recording industry. We have articles in preparation that will detail the changes others have made to assure their economic viability into the future. Make sure your subscription stays current. L.Z.

db

THE SOUND ENGINEERING MAGAZINE

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HOT LINE

• • • • • • • • to the experts.

Could you get a hold of Sansui Electronics' current address and possibly a phone number so I can get a shop manual or schematic and parts list for a Sansui BA 5000 power amplifier? Also, I would like the current address and phone number for Soundcraftsmen so I can request a shop manual or schematic and parts list for the RP-2215-R equalizer. Thank you.

Disco Danny
East Greenville, PA

We contacted Sansui's parts department in California (213) 604-7300, but, while they were very helpful, they no longer have information on the BA 5000 since it is an old model. Soundcraftsmen, located at 2200 So. Ritchey, Santa Ana, CA 92705, (714) 556-6191, does have the information you requested. Darryl Paulsen, their customer relations person, will mail it to you. Our thanks to both companies for their assistance. By the time you read this, you should have the manual that you requested.

I was introduced to your magazine through a recording school, and have been following 2 to 8-track, and now the Electronic Cottage, with great interest. I feel ready to set up a small studio but would like the added confidence of additional information. Are there any manufacturers or suppliers that offer material about various aspects of the recording field?

J. Havestraw
Peoria, Illinois

While other manufacturers may have information available, we have these two sources on file. TEAC Corporation offers two booklets: *Understanding Synchronization*, and *How to Choose a Mixer*. They can be received by writing to Tascam, 7733 Telegraph Rd., Montebello, CA 90640. Yamaha offers a catalog that details digital music systems, and is titled *Introducing Yamaha Digital Musical Systems*. The address is: Yamaha Music Corporation, USA, Digital Musical Instrument Division, PO Box 6600, Buena Park, CA 90622.

Calendar

• On October 10-14, 1988, a short course on Underwater Acoustics and Signal Processing will be offered at Penn State University. Among the topics to be presented are: An Introduction to Acoustic and Sonar Concepts, Transducers and Arrays, Signal Processing, Active Echo Location, and Turbulent and Cavitation Noise.

For more information contact:

Dr. Alan D. Stuart
Penn State Graduate Program in Acoustics
PO Box 30
State College, PA 16804

• **The 130th SMPTE Technical Conference and Equipment Exhibit** will be held **October 15-19, 1988**, at the **Jacob K. Javits Convention Center in New York City**. The event annually provides a forum for discussions and demonstrations on advanced motion-picture and television technology. The theme of this conference is **"Innovations in Imaging and Sound."**

• Upcoming seminars for SYNERGETIC AUDIO CONCEPTS are:

Chicago—September 22-23
Minneapolis—September 27-28
St. Louis—October 6-7
Anaheim—November 1-2

Upcoming workshops are:

Sound Reproduction (Syn-Aud-Con Farm, IN) July 15-17 and August 18-20
Grounding and Shielding (Los Angeles Area) November 17-19
Concert Sound Reinforcement (Los Angeles Area) January 17-19, 1989

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COREY DAVIDSON

db Revisits Nashville

● In the April, 1980 issue of **db Magazine**, our readers were introduced to a new and blossoming Music City, USA. That issue was a veritable tour guide to the Nashville scene which any music-industry related person would have appreciated. In the 1980 story the editor made a promise: "In the future we'll keep a closer eye on Music City...there's a lot happening there, and we'd hate to miss out on any of it." In keeping with that promise, **db** has fulfilled its commitment to inform, educate, and stimulate.

I went to Nashville to find out why there is such a heavy concentration of digital recording. I was very curious as to why a city, that up until about ten years ago had typically shunted the rat-race of techno-hype, was becoming a forerunner of new mastering and production techniques, not to mention sparkling and innovative studio design concepts that seem to be epitomized by the new Tom Hidley 20 Hz control room to be found at Masterfonics.

There are some facts that should be known to fully understand why Nashville is so important. Nashville has a recording history that predates the things that are typically associated with early Beatles' recordings. The earliest rock and roll recordings (that is to say what was bold and daring in its day) were realized in Nashville. It is quite possible that as early as 1928, a Victor employee conducted live recordings of early Grand Ole Opry artists. The 1950s is the period in which Nashville states loudly and clearly that the spirit of pioneer recordings embodied in the early documentation of popular music was happening simultaneously in Nashville as well as other locations throughout the globe.

Nashville today is not just country music but a synthesis of what was and what is. Nashville today is attracting some of the world's most prolific artists, producers and engineers. These are facts. So please enjoy what **db Magazine** has put together in: *Nashville Revisited.*

DISC MASTERING—THE INTRICACIES OF MASTERING

Mastering records is an art that very few people in the world have mastered (excuse the expression). Some of our readers might be a little hazy about the intricacies of mastering. Randy Kling, of Disc Mastering, 30 Music Square West, took some of his precious time (he works a 14 hour + day) to tell us about the process. First a basic definition: Mastering is the process by which a master tape from the artist is documented onto the final medium(s) which will find its way to the marketplace. These mediums presently include records, tapes and CDs.

The most care must be taken when cutting a master disc or record. The spacing and depth of the grooves can literally make or break the final sound quality. Randy says, "The lathe's cutter (stylus), particularly at high frequencies, gets hot. Helium is constantly introduced to the cutter in order to maintain temperature, moisture and proper flow of motion. The amplifier that we use to drive the cutting head is a good 600 watts per channel."

There are a few things that one will see on a console designed for mastering that are not present on a conventional recording console. At first glance a mastering console seems simple. Of course there is no need for numerous input channels because most master tapes are two-channel affairs. "On this Neumann console, which was designed specifically for mastering work, there are gain controls for left and right channels in half dB increments. Nobody can hear a half-dB change, so some might wonder why this capability is there. The reasoning is that you are constantly balancing and watching meters. You know the music and you know that maybe at a certain point in the music there are dynamic changes that need special technical consideration. For example; if you need to ease down on some levels, for a particular passage, that are overbearing and you run into distortion problems yet want to keep what was before and after the event at a good level, the half dB changes allow you to avoid problems without revealing those changes to the listener."

AVOIDING EQ OVERKILL

"The type of equalizers and their bandwidth along with the numbers that designate frequencies, are frequencies that we have become accustomed to working with when making specific EQ changes. Over the years, mastering engineers have given their feedback to the manufacturers in order to arrive at a standard that engineers are used to working with. These frequencies are the result of trial and error. Nevertheless, they are now 'frequency standards' and mastering engineers find these frequencies quite useful. Most often the mastering engineer finds himself attenuating frequencies that can cause damage to the final product. This damage might be of the nature of sub-bass such that the cutter sees too much low frequency excursion. Skipping, overcuts and so forth are problems that are often alleviated by careful attenuation of specific frequencies. On the other hand, you don't need an EQ that is too complex. EQ overkill would be more of a hindrance than a help. Instead of lowering the level of a record because of the presence of unwanted frequencies, you can be creative and satisfy the client by selectively minimizing those problems. Here at Disc Mastering the client is always invited to observe the mastering process and sometimes the process can be a microscope that reveals problems that are on his master tape."

"With this kind of equipment you can attenuate some of the muddiness. A system like this one can hear those problems. You can take away the mud and build up other frequencies that will retain the impact of the sound without necessarily losing anything. This type of EQ serves as a guard rail. There are

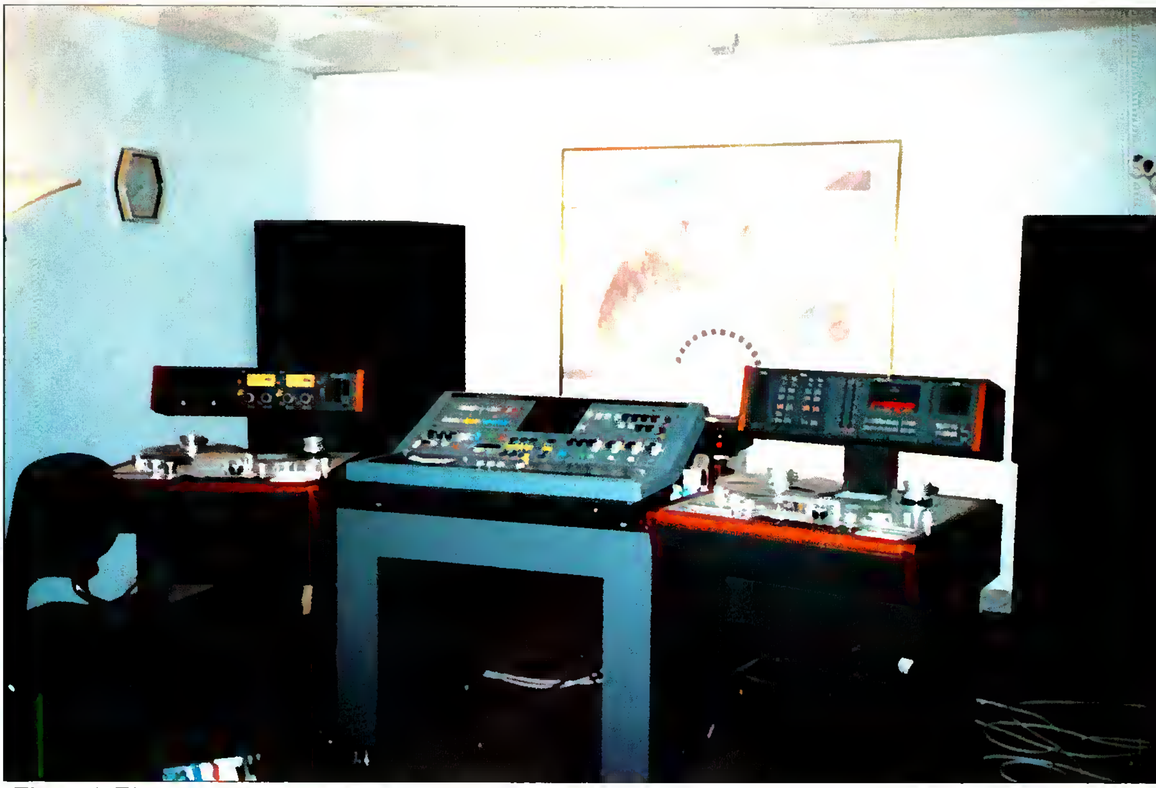


Figure 1. The digital mastering suite. At left, a Studer A-820, the Studer DASH at right, and the Neve DTC-1 console at center.

more frequencies on a mastering console of a given kind and nature than a mixing console. Neve EQ is also utilized in conjunction with the Neumann EQ. On the Neve you've got just about every frequency known to man. The Neumann has peaking and shelving, but on the Neve the peaking and shelving has alterable Q so even at the same frequency you can cut or boost in 1 dB increments. The Neumann cuts and boosts in 2 dB increments. This system of EQ has the ability to overlap in such a way that honky or nasally sounds can be controlled or eliminated. There are many master tapes that we get in here that are picture perfect right out of the studio. I love that. Yet if there are problems, the probability is very good that we can help to achieve a more effective product. Some other features on a mastering console include expanding gates and compression. I call this console my Nevemann."

"In a sense, this type of console can breathe new life into old recordings. Mastering can be a restoration process for older recordings that might be scratchy, thin, resonant or whatever."

"One thing that the mastering engineer must keep in mind is that every one of the client's products (tape, CD, record etc.) needs individual attention. One must understand the market that is out there for a given medium. What works for records doesn't necessarily work for cassettes. The same is true for CDs. For instance, consideration must be given to the way that the consumer uses a particular medium. Cassettes are often played in cars, therefore one needs an understanding of how a product is represented by a car's playback system. Another consideration is the duplication process for a specific medium. The engineer must be able to anticipate what the duplication process might do to the product."

"Digital (CD) is something else. The medium is so revealing that one can sometimes hear things that aren't as apparent in other mediums. On the other hand some things, such as noise,

Figure 2. The analog mastering suite. Tannoy monitors are seen, along with Randy's "Nevemann" console.

aren't there. Once we have a product in the digital domain we cut directly from the Neve digital console directly into lacquer. It's almost like a direct-to-disc operation."

At this point Randy offers to show me how he works. He grabs a half-track master of piano music, puts it on the open face deck, loads the lathe with a blank disc (LP), starts it spinning and says, "This is a half-inch, 30-in./sec tape and I'm threading it in this round about fashion so that the tape passes by the preview head. The other head (playback) feeds the lathe and the computer. The computer tells the lathe what it's going to do to the disc. If you have a dead groove (silence) and suddenly a big loud event happens, the lathe has to know that the groove space is going to be increased at the moment the event happens. If the lathe is sitting at 300 lines per inch for the silent section and the groove spacing remains small for just a little too long, you might hear a foreshadowing of the music. This is called pre-echo. If the song ends cold (abrupt halt into silence) and the lathe doesn't readjust, you risk an after echo that is similar to print through on tape."

Randy starts the master tape rolling and says, "This tape is Dolby SR and there's the tone, see the meter? If Dolby SR wasn't written on the box, then you'd know by the tone that this tape is SR. Now there's the 1000 Hz tone and I'll adjust the preview channel to zero. Now the sync head which feeds the lathe...that gets adjusted to zero. Now look at the scope. Tape azimuth is tweaked visually. We're looking for a straight lateral line...and there it is."

Randy now does the low frequencies, other side of the tape machine and checks the SR to make sure it's being decoded properly. For absolute azimuth Randy puts the channels out of phase and into mono and goes for a null. He then starts the master at the beginning of the music and tells me, "We're at 300 lines per inch so the grooves are practically butted up against one another. Now listen and watch. Think about a second before an event and watch what the lathe does."

The music is very soft sounding like the theme song from the television soap, *The Young And The Restless*. A split second before a louder note is struck, the spacing of the grooves increases and sho-nuf, the music gets louder and more active. "Here's the preview offset (he turns a knob) that actually controls the speed at which the lathe changes from one number of lines-per-inch to another number of lines-per-inch."

CONQUERING REFERENCING PROBLEMS

In the event that you receive a master tape that has problems that the studio was not aware of until it hit the mastering facility, what do you do? "This has happened more than anything else. The issue at hand in that case is the monitoring system. We have had many master tapes go back out to the studio for a remix and yet have never failed a client with our referencing."

Have those referencing problems been diminished by the increased use by producers and engineers of near-field monitors? "It has helped a lot. Many of the near-field monitors are a scaled-down version or mixture of what a bigger speaker does. All near-field really allows you to do is listen at a lower volume. Many people have discovered that you can hear everything just as well at a low volume as at a loud volume. The sensation is not as exciting, but the content is often better interpreted at a lower listening level. Nearly all of our work here is done at low levels."

Figure 3. The Tannoy listening room.

Randy escorts me into the digital mastering room that is home to the Neve digital console, the Studer A820 and the Studer DASH. He states, "Any source that may come in here can be put into the digital domain. In this room we can make composites of analog albums. We can take the 'A' side and the 'B' side and make one long continuous play, which enables us to make the CD masters. The Studer A820 is a dream machine. This deck is able to store and execute all the alignment procedures that we need to perform. The real mind blower is that it executes all those alignments automatically, all the while enabling the engineer to monitor those executions if he wishes...an incredible time saver. Besides accommodating various widths of master tapes and accepting 14-inch reels, the A820 is basically a fantastic improvement upon what was already the world's finest analog. In order to take analog materials into the digital domain, you must have the best analog gear that you can because you know it's going to digital and everything that is on that analog tape will most certainly be there on the digital. On the other hand any problem, as slight as it might be, will be very noticeable once it gets to digital. So the bottom line is that you better have the best analog...that's the A820. Sounds like an ad, I know, but that's what happens when you honestly fall in love with a machine. It takes you higher." We can relate to that, Randy!

THE DIGITAL ADVANTAGE

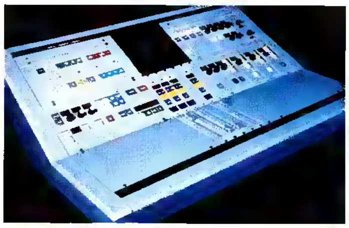
"The Neve DTC-1 console is the key to our digital work. It provides automation of fader moves and a control over EQ, the likes of which has never been known to mastering. The DTC-1 is a multi-facet console. It is the conversion device that enables us to do all of this fancy digital work. We think of it like an interpreter that can speak many languages. We can do all of our transfers from analog to digital via the console. After we convert to digital, the advantages become apparent. Our 26 thousand dollar editor becomes this 10 cent razor blade (Randy holds up a single edge blade). Intercuts and re-sequencing, just like we've done in analog for all these years, can be done with a physical sweep of the blade. This is actually faster than doing all that data entry, numbers etc. The blade is the quick, comfortable approach. The time codes

MASTERPIECES.

Randy Kling waited until he was sure. Waited until the exact equipment was designed and manufactured—the board to his parameters. He has finally assembled all the pieces for the ideal digital mastering situation. Now he is absolutely sure. After mastering 200 gold and platinum records on the analog counterparts of his new digital pieces, he deserves to feel sure. And nowhere in the world can the same combination be found. Now he invites the music industry to finish their masterpieces on his master pieces.

"As a member of the design team that created the Neve DTC, it is fitting that Randy Kling be among the first digital mastering engineers to experience digital audio in a way only possible with the NEVE DIGITAL TRANSFER CONSOLE. The union of our creativities is what makes masterpieces possible."

Tony Langley,
VP of Sales,
Neve



"Randy at Disc Mastering Inc. received the first Studer digital recorder delivered in the U.S. The D820X DASH format (Digital Audio Stationary Head) joins our long line of analog multitrack recorders, and shares with them a 40-year history of Studer performance and innovation."

Studer Revox America, Inc.

"Since the analog days of the 60's, Randy has depended on Tannoy reference studio monitors for the truth, the whole truth, and nothing but the truth. For the 80's and the demanding age of digital, now more than ever, Tannoy."

Bill Calma
Marketing Manager
Tannoy North America, Inc.

"Sony Professional Audio is proud to be associated with Randy—by supplying digital audio equipment to create even better masterpieces, and to proceed with him into the future of this new and exciting technology."

Graeme Goodall
Sony Professional Audio Division, Music City, U.S.A.

"It is befitting that Ampex, the first magnetic tape manufacturer to develop and market a digital audio tape, should be the tape of choice for the first all digital mastering studio. We at Ampex congratulate Randy Kling and are proud to be a part of his state-of-the-art digital mastering system."

Warren K. Simmons
Senior Product Manager
Professional Audio Tape
Magnetic Tape Division
Ampex Corp.

"We salute Randy Kling for being amongst the first major mastering studios to make such a complete commitment to Monster-Cable wire technology with the re-wiring of Disc Mastering. Randy's commitment to re-wiring is of such a magnitude that he has taken the time to wire through walls, fabricate his own custom Monster-Cable interconnects, and completely interface digital consoles, digital recorders, cutting lathes, electronics, and monitor speakers exclusively with Monster-Cable M Series. Financial and time commitments of this magnitude are not easy to make, and demonstrate to his customers his desire to master the best sounding recordings possible for the greater enjoyment of all music lovers."

Noel Lee, Head Monster
Monster-Cable Products, Inc.

DISC MASTERING

27 MUSIC SQUARE EAST/NASHVILLE, TN 37203 615-254-8825

will, of course, follow from machine to machine so that we can replicate at all times. Mainly, aside from the sonic improvements, the advantage that is gained using digital is that all the levels, fader moves and EQ, can be stored on a floppy disc."

Randy is now about to show me an exclusive Disc Mastering process. Their own version of direct to disc. "The signal from the DASH comes into the DTC-1."

Randy then takes cables from the rack that houses the 1630 and walks the ends into the other room (where the lathe resides) and plugs the cables right into the cutting amplifiers. "Now we've got the digital in all its glory going directly to the cutter head. Essentially the DTC-1 goes right to disc."

MORE LISTENING OPTIONS

One of the highlights at Disc Mastering that Randy is very proud of is their Tannoy listening room. "What we are trying to do with this listening room is give the clients more listening options. We take pride in having such information available. This Tannoy listening room helps us to establish a line of communication with the client. If a product comes in here and the client says, 'Gee, it sounds different in here than it did in the studio,' we can investigate various references in this listening room and narrow down the sound that the client is searching for. Once that is done, we now have a set of criteria which will enable us to communicate with the client about his product. Basically, if the client knows more about referencing, our relationship with that client is strengthened. Sure, there are some clients who have no interest whatsoever in these things. In many cases the client simply puts a trust in what we do here and their confidence in us is maintained. That is usually the case. As time and technology forge ahead, the number of technologically hip clients increases and we welcome them."

It is a curiosity that Nashville is a haven for digital recording. Most of Nashville's music traffic has previously been comprised of song-oriented music...that is to say, music that stresses the lyric rather than esoteric sonic quality.

If this is so, why has Nashville established itself as a powerful proponent of digital? Randy responds, "First of all, Nashville is not entirely made up of an engineering community that is strictly from Nashville. There are many people that have filtered down to these parts from big cities all over the U.S. Many of these people have long histories in recording and industry related fields. Secondly, there is a concern on the part of many studio owners not to get left behind. In most other cities or centers of recording activity, there has been a tendency to buy the very next and latest thing. Here, in Nashville, there is a fear of getting caught up in that game. The studio owners thoroughly investigate a product and shop by carefully comparing items. For instance, in many places the Sony 3324 is considered to be the one to buy. However, the fact is that many have accepted this for face value and assume that the machine that gains popularity is the winner. This is not so. In other places, outside of Nashville the 3324 is popular. In Nashville there are two. The rest are mostly X850s. This is because much research has been done with the X850 in conjunction with various types of analog filters. In Nashville, preferences are based on what works the best for that studio, not what everyone else is using. I am using a Studer DASH which by the way, not many other people are using right now. For me, it is what I want, what I need, and most importantly what is best serving and satisfying my clients."

MASTERFONICS—A HIDLEY-DESIGNED STUDIO

Figure 1. Glenn Meadows at the helm. The Neumann lathe and mastering console are also seen.

Audio professionals everywhere are turning to the Fostex E-Series recorders for their production and post-production needs. So much so, you hear the results of their work nearly every day — in movie soundtracks, commercial and cable television shows, industrial and educational films and videos and, of course, hit records.

The E-Series features gapless "punching" so there's no blank space after the punch-out point. Only recorders which are much more expensive offer this sophisticated function. But since you can't run a fully automated system without it, Fostex includes gapless punch-in/out as standard equipment on the E-Series.

Also standard is a synchronizer port which will interface with all SMPTE time code based systems. When used with the Fostex synchronizer, Model 4030, you can then use our software program to perform sophisticated audio assembly editing.

Models E-8 and E-16 are multitrack recorders with built-in noise reduction.

Models E-2 and E-22 (not shown) are 2-track master recorders with a third, center channel for SMPTE time code control. This is a standard feature, not an option. You will have complete compatibility with existing 2-track tapes, plus the ability to run computer derived edit decision lists and full automation.

Servo control of the reels in the edit mode will help you pin-point cues and spot erase. When the pitch control is engaged, the exact percentage of speed deviation is displayed so that when you need to re-set the control, you can do so precisely, and the real-time counter features search-to-zero even from the negative domain.

The E-2 uses 1/4" tape at 7-1/2 & 15 ips (15 & 30 ips speeds are optional); the E-22 uses 1/2" tape at 15 & 30 ips.

When an E-Series recorder is used with Fostex Model 4050 — autolocator and SMPTE to MIDI controller — you have programmable punch-in/out, 100-point autolocate capability. 10 programmable edits, a SMPTE time code generator / reader (all four formats), plus the ability to locate to the bar and beat.

So if you're looking for a professional recording instrument, there's a Fostex E-Series recorder that can help you wth two important "E" words: Efficiency and Effectiveness. The E-Series can also help you achieve the most important "E" word of all: Excellence.

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Figure 2. The Hidley Reference Design, one of two at the studio. Notice the vertical cone configuration with centered horn lens.

I arranged for an appointment at Masterfonics with Glenn Meadows at a time that he would be 'the least busy.' In reality, there is no such time for him. With two mastering rooms (complete with lathes), cassette duplication and a monumental mix room, the bookings are carefully scheduled by a very sharp staff. Nevertheless we completed an insightful interview which was supported by his chief technical engineer, Milan Bogdan.

Before we began any discussions, Glenn invited me into the mix room. At first glance, besides the unusual looking monitors, this room doesn't appear to be much different than any other world-class control room. "Have a seat," Glenn says. "Before we get started with the interview I would like to play you some music."

He put on a CD of recent Chicago. Glenn was sitting in the shadows at the back wall of the control room. I was comfortably seated at the center of the console in a cushy seat, rolling a couple of feet from side to side in the mix position about 15 feet in front of Glenn. I started listening. The control room volume approximated a home listening level...not very loud. In fact, Glenn told me (from 15 feet away) in a normal voice, "Put the monitor level to where you like it." I was now listening to a cut at a moderate level. Glenn then cued up a Lee Ritenour CD. Now I could really sink my ears into something. I began increasing the gain to levels that would have a pair of 813s lit and left the volume there. I was now frantically moving my seat from side to side to determine whether or not there were speakers behind a center curtain on the monitor wall directly in front of me. Glenn was grinning like a Cheshire cat. Upon noticing Glenn's devious smile, I began lowering the level over a period of maybe 40 seconds until the sound of my own whispering could be distinctly heard. The sensation of certain sounds coming from a central location was still there. What was so astounding was that the powerful bass was still as potent as at the higher levels. Glenn, they sound like near-fields at this low level! "I know," he said. Glenn, the bass is incredible! "I know," he said. Glenn, the separation and sensation is like that of headphones. "Yes," he said.

Figure 3. The A mix room (20 Hz control room).

I was curious as to how Masterfonics had achieved such effective bass response at low levels from such gigantic monitors. Milan explains, "The reason why they sound so good at low levels is that the impedance levels of the circuits are the same and there is a constant impedance control. As you change the level, the Z remains constant. This condition might show that the Fletcher-Munson curve is wrong. That curve might actually be an impedance curve and not a listening curve. There is no tuning in the control room. No electronic crossovers. That's because of the current demands of the room. The amount of current that we need to achieve the higher listening levels exceeds any normal current levels."

Glenn says, "If you're getting a peak of 120 dB sitting at the console, you're probably hitting 126-127 dB at the baffle board of the speaker. The room is actually inefficient due to the large volume of trapping space that is present in order to control the low frequency response of the room. That's going to suck up power.

Milan adds, "Part of the reason why we have so much trapping (18 feet in one place alone, above the control room ceiling) is so the room won't go into acoustic compression. This room won't start compressing until levels are at 160 dB SPL. So when a transient of a snare drum hits, the room hasn't compressed it and you can hear its real character. That gives you the ability to put it in the mix where it's supposed to be. The average control room goes into compression at about 110 dB."

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Glenn states, "Acoustic compression is like having a limiter across your mix bus. You push it up further in the mix at a high SPL in the room and the meters may go higher but the sound doesn't get any louder."

Milan adds, "All of the modifications to the console are going to be totally meaningless unless the room reflects the acoustic truth."

Glenn tells us, "Ninety percent of the sessions that are done in that mix room have no little speakers on the console. Not a pair of Auratones, no N.F.Ms. Right now the client has a single Auratone set up mono in order to check for compatibility on radio. They're not using Yamahas, MDMs, Tannoys, or any small speakers in that room. Tom Hidley's design philosophy in this new generation of rooms is: 'The wall speakers should be right under any condition at all times.' If the room is built right and the monitors are correct, you don't need room voicing or tweaking or near-fields. Behind the back wall of the control room there is a 20 cycle bass trap that goes up 18 feet and is 6 feet deep. All the way around the room there are numerous low frequency traps at various heights that are tuned to different frequencies."

ULTIMATE MIX ROOM

Glenn reflects, "The whole idea behind this room is predicated as follows: We had a certain amount of space and I wanted to build the ultimate mix room. It was presented to Mr. Hidley in just those terms. He came back to me and said, 'Look, I have this idea...a concept for this room that is flat down to 20 cycles and you've got enough space to build it.' He showed me the basic layout and I said, let's do it! I gave Tom Hidley free reign to do what he felt was correct. Tom's own construction crew built that room. The construction time is probably frightening to most people as to how long it took. We started our structural upgrade to the room on July 1st, 1986. We had to take off the roof on that section of the building to raise it 7 feet in order to get 20 feet clear inside. We had to build isolation walls, cut existing slabs and pour new slabs. That took a total of 6 weeks to be completed. We gave the room to Tom on August 14th, 1986 to work with his interior crew. The first session was on October 15th, 1986. His crews left Nashville with the room completed on October 1st. Seven weeks total. The other two weeks consisted of console interfacing and minor door trims. The glass fiber mesh in the floor was done in the 6 week period when we poured slabs. That was our doing. It was part of our site prep work." (The glass fiber mesh is explained later in this story.)

A subject that is certainly hard to avoid is the analog/digital controversy. Masterfonics has hosted many tests that are designed to compare and evaluate analog and digital mediums. However, Glenn and Milan have taken the issue a bit more seriously than most studios. "...after all, we're responsible for turning out a product that goes right to the consumer."

Milan is a graduate of Lawrence Tech, Detroit, Michigan. He studied electronics for eight years in Detroit. From 1964 to the present, Milan has held a chief engineer position at every studio where he has worked. He says, "I think this is the first time in the history of pro-audio that a consumer product is controlling the pro-end of the business. The fact that the CD is work-

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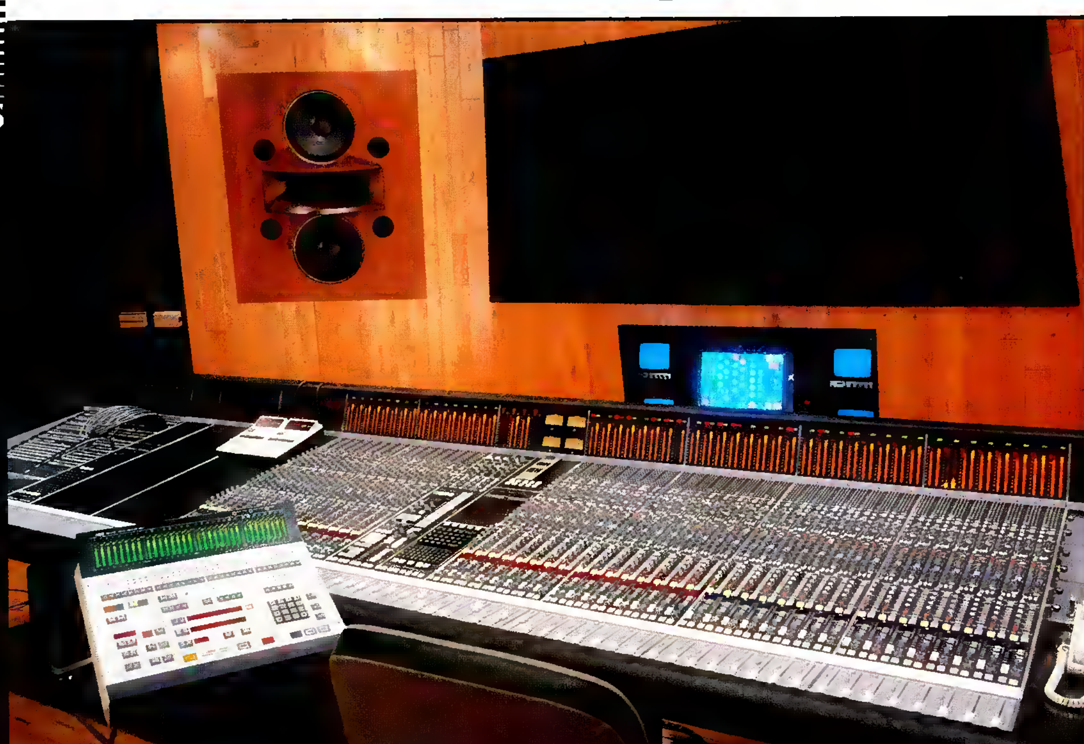

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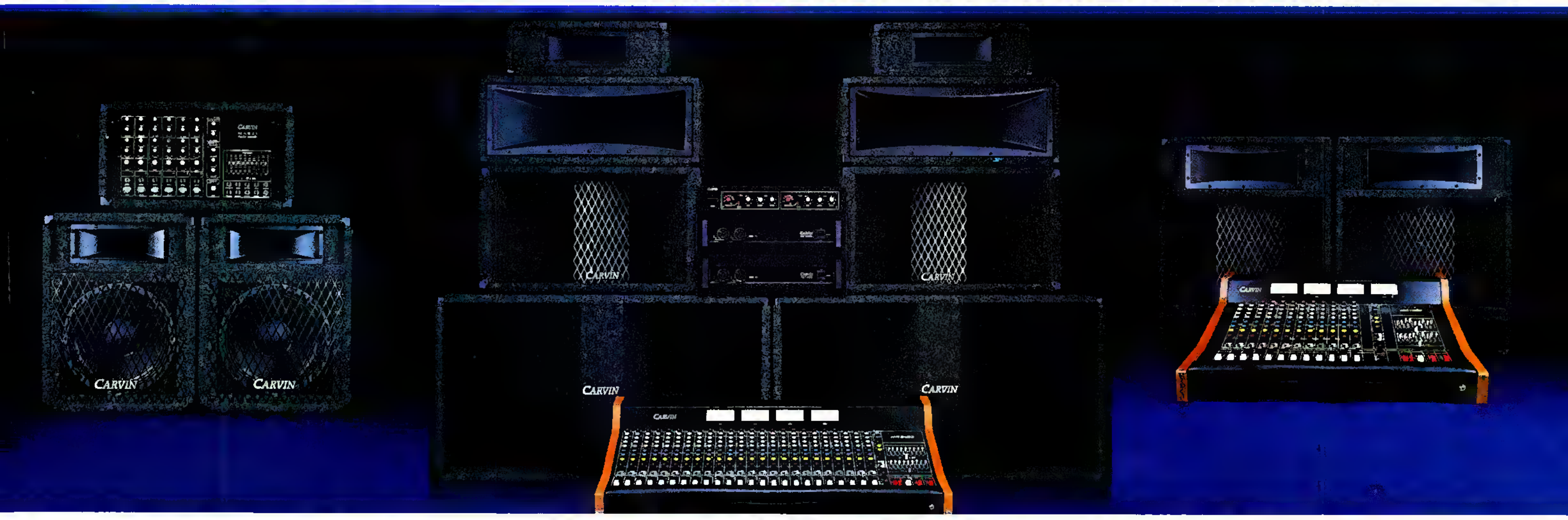

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DM93

ing at 44.1 kHz doesn't mean that everyone has to kneel down to that standard. What happened in the 1960s and 70s was that everyone was debating what the bandwidth should be for the mix bus on a console. Most of us agreed that the wider bandwidth was more desirable. Now, today, we have a consumer product that's got a hard brick filter at 20 kHz. In my opinion, only god can design a filter that's going to work that well. They all ring...they all have problems. The point that I am making is that whether or not it's digital is not the problem. I am a proponent and advocate of digital. However, if the medium is to get a fair shake, we cannot allow one company or one standard or one engineer's invention to clog up the works."

Milan continues, "Many people in the studios are encountering problems that instill fear in the hearts of owners. The fact is that right now, in order to be an integral part of the digital community, one must accept certain technical responsibilities because the manufacturers are not maintaining a high enough level of responsibility to the industry. The manufacturers are mostly concerned with making a studio owner swallow that next wonderful invention. The problem is not the fact that it's digital. The fact that we're scanning at a lower frequency (44.1 k) and having to build a filter that's very hard to design is the problem. What I want to see is an audiophile CD that scans at 70 kHz. These problems are not new."

Glenn adds, "When people record on junky consumer video decks, bring that tape somewhere else and play it, they find out that the tape won't interchange. If you have an F-1 that has an interface box that is displaying error correction rate, the error correction lights are on all the time. That's not a function of the digital...that's a function of the storage medium. Here, on our JVC systems utilizing half-inch tape, we can go for twenty minutes without a single blink of an error correction indicator light. Good results are proportional to the quality of the tape that you're using and the quality of the deck that you're using. Look at some of the PD format machines; the error correction lights hardly ever blink. Look at a Sony 3324 and observe the error correction lights; they're blinking like a Christmas tree. Look at a 3202 tape that is recorded on that machine and played back on the same machine that recorded it; the error correction lights are flashing like crazy. There are other versions of the same machine (3202) where the correction lights don't blink at all. Why does one do it and another doesn't. There are countless problems in the open reel format systems."

VOIDING WARRANTIES

One of the most intriguing facets of Masterfonics is what their philosophy is when it comes to out-of-the-box equipment. Milan says, "We have voided the warrantee on almost every piece of gear that has come into this studio. The SSL console needed design modifications right away. That console was smearing frequencies. We designed a corrective circuit modification and were even willing to offer our findings and design mods to SSL. They wouldn't talk to us about it. Isn't it funny though, how a number of studio owners have stated that they don't necessarily like their SSLs but have them in order to please their clients who insist upon the ability to recall. We wanted our clients to have that ability too, but were not about to settle for the sonic state of an out-of-the-box console so we took it upon ourselves to redesign. This has, of course, voided our manufacturers warrantee. Virtually every piece of equipment that we buy is altered in some way so that it can fit in with our standards of studio quality. We typically improve noise characteristics beyond what the manufacturer has established. Power supplies are modified, cases altered for mounting purposes, wires upgraded, you name it, we've done it...and it's all for our clients."

Glenn and Milan told me that they had actually eliminated that age old problem of electric guitar ground hum directivity (That's when the guitar player turns in different directions causing a change in a ground hum. Sometimes a quiet spot can be found if the player stays perfectly in position). Milan informs us, "A large part of that is coming through the reinforcing rods in the floor. Very often steel mesh is used. Much of that can be eliminated with proper design techniques. Here, we eliminated the problem by removing that steel mesh and got approval from the building code to use glass fiber reinforcing materials. (Earlier in this interview, Glenn mentioned the 'glass fiber mesh' that was installed during the preliminary stages of construction.) If construction didn't take this phenomenon into account, then one has to refer to prints and determine where your location is in respect to the RF fields. One also has to look into vibrational analysis. Digital machines go down to DC. Rumble control is more critical than ever. We can put a 1.5 v d.c battery on the input of our JVC and it will read 1.4 on playback. Now that super low-frequency digital-ability is going to effect the consoles where there are many capacitors, coils and transformers. That low rumble can cause offset and all kinds of problems."

Power conditioning has been a major concern at Masterfonics...so much so that commercially available tech-power wasn't good enough. Milan explains, "We have a million to one (1,000,000:1) spike suppression. We created our own tech power. The console supply, tape machine supplies and power amp supplies are run on 220 V. We use 220 V because it's a balanced circuit. We take advantage of the common mode rejection of the a.c. system at 220 V. This also offers more available current. The hum in turn is drastically reduced along with some of the RF. When a system like this is installed, the length of the cables becomes most critical. The supply wires have to be exactly the same length. As a studio owner, you have to be there all the time to oversee these steps."

Glenn adds, "We oversized all the wiring to outlets in the walls...everything is number eight wire. Even on a 15 amp circuit number eight wires are used. With a heavier gauge, if it's a long run to the panel, you're not going to get a voltage drop on that line. There are no daisy-chained neutrals from one outlet to the next anywhere in here. Every outlet has its own neutral. None of the conduit is grounded to any other piece of conduit. They wrapped everything that went through a wall in rubber."

Milan continues, "When you use the kind of transformers that we have to generate your own in-house tech power, you can ground the primary side (that passes UL code) which gives you a floating neutral on the secondary side. You can, for instance, run all those neutrals into a separate box. There's no ground plane in the control room...no hum, no RF because we have our own neutral on the secondary of those regulators."

There are no standard three-prong outlets to be found in the control room. Glenn tells us, "A client cannot come in here and just plug up his gear. We have special twist locks on all the outlets. You have to come to us for a special extension cord that will adapt you to our power. When a client plugs his rack of whatever he's got to our power, upon entering the console there is an absence of noise. No hums. Even gear that gives the client occasional problems outside of this studio will behave much better in this studio. It's not magic, just good old fashioned high-tech problem solving."

Milan informs us, "Once we increased the current to the power amplifiers, we noticed that the type of cable that we used became far less critical. The slew rate of our monitor amps was vastly improved. The most interesting thing about the improved transient response due to an increase in available current is that we could no longer hear the difference in the cable. This type of improvement shoots gaping holes in the claims of some of those high-resolution cable manufacturers."

What's down the pike for Masterfonics? Glenn answers, "Our long range plans include having a second 20 Hz control room with a studio that's accessible video-wise from either control room. The cloth in the center would be changed to a white cloth to enable video projection of the studio. It will look like you're looking into the studio. Now, with the penetration that we're getting with this mix room, we're not getting any more of those phone calls where they ask, '...well how much is your room?...you're kidding me.' or 'that's ridiculous' or whatever they used to say. Now we're getting calls like, '...when can I get in?' This is a mastering facility so what comes out of this studio is destined to become product. The fact that we have a mix room of this kind has cemented our place in this industry and has given us a new edge."

AN INDEPENDENT'S POINT OF VIEW

As we were wrapping up our interview, a client was getting started in the 20 Hz mix room. Pete Tillisch is a producer/engineer. Pete initially graduated from New York's Institute of Audio Research in New York City. His first job out of school was at Capricorn Records as a second engineer. A few years later Pete moved to Nashville. Ever since he has been busy engineering many of the country records that we know. I thought it would be interesting to hear some of his comments. You have chosen this room for whatever reasons. Care to elaborate? "The main reason is that this is one of the only rooms that I have found where you don't have to anticipate the sound outside of this room. In all the years that I have been engineering, I've always been second guessing and compensating. When I mix here I get exactly what I want. Upon leaving the room and listening elsewhere I find that it's the same. I have been to some of this country's finest control rooms and have never experienced this kind of truth."

DIGITAL RECORDERS – HYBRID DESIGNS

Digital Recorders is one of the largest studios in Nashville. With three rooms, three of the Sony 3324s, and an impressive availability of live recording space, this studio complex has received world acclaim and more than its share of superstar recording artists. Norbert Putnum, owner/chief engineer, informs me, "We started out two and a half years ago over on 16th Avenue South with one studio. We expanded last year into a three room complex located here at 49 Music Square West and we are anxiously awaiting the arrival of the newest 48-track Sony digital machine."

Norbert, in conjunction with some of this country's finest engineers and designers, has employed unique and innovative hybrid designs such as their Trident A-Range console with Necam (certainly not an out-of-the-crate item), and this country's only fully automated/computer controlled concert grand piano! Norbert continues to tell us about more of his studio's highlights...

"The Trident A-Range has been modified as the years have gone by and it has Necam 2 automation, which never existed on a Trident before. So our A-room console is a combination of Neve and Trident coupled with a 3324 recorder. Studio A has a 30-foot

Figure 1. Norbert Putnam and engineer Eric Paul at the console.

high ceiling on top of a 40 by 50-foot room. In addition is an 18 by 20-foot isolation booth. The monitor system consists of the UREI 815s and Tannoy near-fields on the console. In studio B, a 20 by 30-foot space with a 20-foot ceiling, we have a Sony MXP 3000 console with hard disk drive automation running with a 3324 machine. The 3000 is the quietist console that I have ever experienced. At times its quietness can reveal the limitations of other instruments and equipment. Flaws such as noise can become very apparent. The 3000 has the potential to emulate a microscope. You can see tremendous detail...which by the way is the perfectly logical choice for a 3324. Again, in this room you'll find the UREI and Tannoy monitors. Studio C has a Trident 24 console which is also attached to another Sony 3324 tape machine."

How would you describe the acoustic characteristics of your A and B rooms? "The acoustics in studio A are a tradeoff between a very live room and a reflective but controlled room. On the walls we installed 2-inch strips of oak which give the live but controlled character. On the other hand, the high ceiling gives this room the spatial character that might typically be associated with a tremendous room. The flooring is parquet wood and is what we feel most clients prefer...a live surface with a warm sound. The delay time from the center of the room is about a second and half. This delay time is actually short by comparison with what many of the English studios offer. By having a short reverb time we are able to do high-tech basics. By that, I mean that we can simulate gated type effects with just the room sound. In a three second room, this effect would be nearly impossible to achieve."

"There is a balcony that overlooks studio A. This makes for an excellent vantage point for video cameras. There is also a lighting grid. Emmy Lou Harris, Arlo Guthrie, and Bonnie Raitt were in 'A' last month shooting in 35mm film for a CBS documentary on Woodie Guthrie. Studio A has nearly all the characteristics of a laboratory theater with the exception of superior sound and recording."

SONIC RELATIONSHIP

Has the relationship between your consoles such as the Tridents and the Sony 3324 digitals given you any kind of an edge? "Oh yes. The reason why we have the Tridents is because there is a special sonic relationship with the 3324s. Without increasing the slew rates of the Tridents, we discovered that we were able to recreate the warmth and fatness that analog tape recorders typically deliver. The Trident delivers about 13 volts per microsecond, not the fastest by any means. However, we discovered that the bottom-end frequency response, due to the slewing, has a very potent psychoacoustic effect. This is probably the result of rollback of transient response at the very high frequencies. Psychoacoustically this is interpreted as a fat warm sound. Even in the event that this explanation doesn't convince the techy-types, the fact is that our producer, engineer, and artist clientele hear this phenomenon and are very impressed. By having the Trident A-Range we have achieved this special level of sound. By the way, many of our clients do not like a high tech sound. Let's face it, if you can hear the technology you probably won't like the sounds."

Norbert, why did you go Sony in a Mitsubishi town? "Because when we went into business two and a half years ago, Nashville was the only Mitsubishi town. I believe that Nashville probably got the Mitsubishi band-wagon rolling. I chose not to jump on that wagon because the Sony format was far more compatible with what we believed to be more familiar and simpler."

Please explain. "In terms of *familiar*, the compatibility of our studios with other studios is almost like establishing a standard in the industry. Incidentally, there are far more 3324s throughout the world than any other digital machine. The are seven Sony machines in London...there is one Mitsubishi."

How does *simple* apply to this issue? "Well, the word is out that Sony is close to a 48-track version of the 3324 format. Studer is also close to an available 48-track machine." Ah...now I'm getting the picture. "Yes sir, we're talking about a wonderful compatibility that will facilitate the continued use of our Sony 24-track machines after the 48-track version becomes a reality. Sony has already proven that their sync capabilities are phenomenal."

We know that the powerful synchronous operation of the 3324s is a key to recording via satellite (**db** May/June '87 Mastersound Astoria's live, bi-coastal, satellite session).

Is this a feasible endeavor for the Nashville scene? "At this very moment in time I do not believe that there are

Figure 2. Engineer Eric Paul seen from the other side of the console.

enough superstar artists to support that kind of process, but, Nashville certainly has no problem with the up-link and down-link. Video/satellite wise, Nashville is very well endowed. From this studio, we could certainly do it with ease."

Could you tell us about some of the recent projects at DR? "Etta James is now in recording. Although this record is a fairly traditional one, there is an interesting twist...the vocal documentation is pretty straight forward yet, the basics, namely the drums, are electronic. Roger Hawkins (recent recordings with Aretha Franklin and Paul Simon) is using a very high-tech kit. He's using the 'd-drums' for most of the sources. The tom-toms and kick are electronic, but the cymbals and snare drum are *real*. This is proving to be a super fast session. The toms and kick are gated and the space around the percussion sounds is wonderful. Those sounds were nailed in less than an hour. That's the beauty of the electronic drums. On this same session there is a lot of Hammond B-3. Barry Beckett is playing keyboards on this record and much to our surprise is getting a great deal of mileage out of the old Wurlitzer electric piano. So this session is the best of the older school of traditional vocal and instrumental recording in conjunction with an ultra-modern approach to the documentation of the basic tracks."

ONE OF A KIND

Would you describe the aforementioned special piano of yours? "I think that we have the only MIDI compatible Bösendorfer concert grand piano in America. Bösendorfer has been working on this for a number of years. At the present time it is a computer programmable grand piano. There are sensors underneath every key. That sensing information is interpreted by an IBM computer. The IBM computer scans the keyboard many times per second enabling the computer to store all the note and dynamic information. All of this information can, of course, be redefined in the MIDI format. What this system can do without the MIDI is basically this: A musician can sit down and play a piece all the way through. He can then punch a few keys on the IBM and that information is sent to solenoid controlled motors underneath the piano's action. The advantage is if there is a great performance by the musician but there are a couple of mistakes, we can go back and fix virtually any aspect of the performance including the addition of information that was never there to begin with. All of this information can be reviewed on the screen."

"Lets say that the performer botched 6 notes. He can go back to the screen and review every bar of the piece. The piano can play along while in this review mode. He can back it up a bar, slow down the speed of it, determine which notes need the correction, and then go back a play the correct passage. If the performer plays a real fast arpeggio but accidently leaves out two notes, he can now go back and re-insert the two notes, go through the whole piece and correct all the imperfections including the dynamics. It has full dynamic range. You can run the computer and while the piece is playing back in an overdub mode, go over to the piano push down on a key so slowly that the hammer doesn't strike the string. When you play it all back you will hear and see the piece zipping along and observe that one key slowly but silently sinks down. On the other hand you can pound on the keyboard and the system recreates that as well." This must be a dream.

"Bösendorfer essentially designed this system as an educational device. The first one went to the Royal Academy Of Music in London. It allows a student to sit at the keyboard and review Mozart being played by Vladimir Horowitz. The student can then adjust his performance to Horowitz's performance. As a teaching tool this system is unsurpassed. As a tool in a recording studio I think one can certainly see how powerful this is."

"We are presently in communication with Bösendorfer about designing a lock up with the IBM so that this system can chase SMPTE code. They are presently designing a MIDI interface. At the moment, the system's present state is very effective and exciting. We've got well over a hundred performances by some of the world's greatest performers. Sometimes we go into the studio at five o'clock with a cocktail and unwind with a little Oscar Peterson at the world's greatest piano bar!" Norbert, I think I will be coming real soon for that experience. (With that we both roared with laughter. There's got to be a lighter side to studio life. We just wonder if Vladimir Horowitz and Mr. Peterson occasionally unwind at a piano bar like this one.)

IN THE WORKS

In a studio of high caliber, such as yours, there is often the presence of tech people who have new and exciting ideas. What are some of those ideas that might already be in the works? "Indeed there are some special things in the works. Howard Steele, a premier engineer, is presently working on a new modification for the Trident A-Range to increase the slew rate. What we are hoping to offer is 10 modules of super slew rate. The rest of the console will be maintained as is, so that there will be a choice of the *super slew* or the other character discussed earlier."

"Another project in the works which Howard is spearheading is a new approach to the near-field concept. Basically, without getting too involved, this idea incorporates a giant sub-woofer system in the room with a closer proximity of the higher frequency content. This will hopefully relieve some of the strange things that occur in the translation of low end information when using near-field monitors."

Getting back to tape machines for a minute...in light of the fact that you would like to see more than 24-tracks on a single machine, why didn't you go 32-track? "I never felt that 32-tracks was enough. I was locked-up analog 48-track prior to my move to digital recorders (good name for a studio, huh?). You see, many of us were accustomed to a dual 24-track analog configuration. 2-24s locked up has been my 48-track experience. It's the way that many of us are used to working. It is a mystery to me as to why certain manufacturers bothered to build a 32-track machine. Thank heaven that two of the greatest machine manufacturers have addressed 48-tracks. On Studer's part it is a very logical choice because their machines have been locking up for a long time. On Sony's behalf I can safely say that they have truly maintained a trust and commitment to fulfilling the needs of the people who use their machines rather than trying to impose some *out-of-touch* format that won't be around down the road."

"I'm more confident than ever that I made the right choice with the 3324s."

REEDER AND NICHOLS—AN ELECTRONIC COTTAGE IN NASHVILLE

As I was preparing to leave Masterfonics I sat in the lounge and talked with a woman named Conrad Reeder. She informed me that her husband, Roger Nichols (Grammy award winning engineer best known for his work on all of Steely Dan's records) was meeting with Glenn Meadows. This conversation led to another interview that Saturday. Conrad, Roger and I spent a casual afternoon/luncheon together and finished up a taped interview in my hotel room later that day.

We began discussing our mutual interests. I soon discovered that Conrad (Connie) was quite involved in a private home recording environment where she has been able to circumvent the pressure and expense of commercial facilities in order to realize her own compositional endeavors. She is an extremely intelligent person with an intuitive grasp of electronics and the world of MIDI recording techniques. I wasn't at all surprised to discover that Connie has an extensive background in performance as a singer, actress, dancer, model, writer and producer.

Her credits include John Denver, Phoebe Snow, Jack Mack and the Heart Attack, and the rock band Dakota. Connie attended the Del-Mar Media Arts Commercial Training Course, HB Studios where she studied acting with Stephen Strimple, and Austin Peay State University where she received a B.A. in Music Education. She is an instrumental expert on the oboe and sports an instrumental command of piano and synthesizers.

"I started doing studio work here in Nashville about ten years ago for Tree Publishing as a staff artist for two years. I loved being in the studio. It was like being in a space ship...the lights and all the dials...like being in your own ship. From there (Nashville) I moved to New York. Tree Publishing was trying to bring in pop and rock writers into Nashville as a means of opening up enterprise. I don't think that Tree met with very much success at that time because...well...it just wasn't working back then. It was this which took me to New York, where there was a lot more action in pop and rock music. In New York I worked on a record with Phoebe Snow. With that record I became a bit more visible and began doing back-up vocals and woodwinds with some major shows more frequently. In 1983 through 1986 I got pretty busy...even did my own show in the New York nightclubs billed as *The Conrad Reeder Show*."

"My first really positive experience outside of the back-up work that I was doing was the recordings we did for Columbia records with Dakota. The band Dakota enabled me to compose and sing which was a nice transition from having been just a background singer doing other people's stuff. Shortly after my jaunt with Dakota I began an interest in trying to record things on my own. To write my own songs and to be able to record them seemed like a dream come true. I started with a little 4-track Fostex cassette and a couple of little Roland boxes that could play a click-boom kind of drum sound and a bass line. You know those little silver Roland boxes? One did drum beats and the other did a bass line...a simple system but one that was good enough to sketch out my ideas."

How far did you go with those funny little silver Roland jobs? "I did demos on those things which actually generated interest on the part of Morocco Records. They were interested in me as a solo artist and they signed me to the label. In the meantime I was on the road performing. It took about six months between the period that I was signed and before I finished the demos. Morocco Records dried up and I was left with my tracks and more ideas, but I was still on the road. At this point I had to concentrate on money and making a living. I was on the road with John Denver for almost five years doing work with various other artists on their albums as well."

What resulted from your experiences with a record company? "During the period in which I was signed to Morocco, I made some observations. There was money being wasted on inefficient use of studio time as well as unnecessary pressure that was caused by some people having the notion that songs are recorded in one shot on one given day. This kind of pressure had a negative effect on the music. The

Figure 1. Conrad Reeder and Roger Nichols.

budgets only allowed for one shot demos. There were many instances where I wanted things changed and nothing could be done because there was no time left or the money had run out."

BEYOND 4-TRACK

What began to happen in your home recording environment? "I felt that I was ready to step up my equipment and tracking capabilities beyond the 4-track level. The bouncing was what was beginning to stifle me. It just doesn't sound good when too much information winds up on one track. I looked, with my husband, for recorders that would satisfy my needs. I got the AKAI MG1212. Now that I had twelve tracks available to me, there were more things...*treatments* that I wanted to give to my music. One thing led to another and I started adding outboards and effects to my little Electronic Cottage. I soon discovered that a sequencer would help me to record certain ideas and purchased an MSQ700 sequencer. The 700 served as my bass player. The sequencer also helped me to save space on tape when it came time to record keyboard parts. Now I use a Roland MC500 which uses discs for storage. Occasionally I will still pull out the MSQ for extra parts or space. My forte is really singing so I want to be sure to keep my available tracks for vocals and vocal arrangements."

"The drums are generated by a DMX that I still like very much but the real neat sounds that I get are from, thanks to Roger, the little Wendel Jr. units. The drum sounds on the Wendel are real sampled drum sounds so if you have any talent for programming a drum machine, you will have no worries when it comes to the sound of the drums with a device like a Wendel. I also don't think that it's desirable to have a drum beat that is too clock perfect. Once again the Wendel serves me well by reintroducing a human feel to programmed drums."

Roger reveals some things about Wendel, "Wendel Jr. triggers off of audio signals that it sees. We do not tolerate MIDI delay which is what you get when using MIDI to fire drum events. Wendel drum sounds are 50 kHz samples of drums with decays that are as long as you want. The human feel is a built in variable for Wendel. A normal drum machine will make a rapidly played snare sound like a machine gun. The Wendel knows when it sees a signal like that and will play a pre-programmed alternating set of samples so that the nuances of separately attacked drums are brought out. The cymbals in Wendel work in the same fashion. No glitches."

Roger continues, "Connie has 4 Wendels but when I'm home from the road or a gig, I bring my Wendels home and Connie now has an army of Wendels. I guess if we get divorced, the stickiest part of the settlement will be how we split up the Wendels."

"The Electronic Cottage has enabled me to move into other areas of production such as advertising." Connie says. "I discovered that I could write and record tunes of good enough quality that the pieces could actually be used on the air. The clients that I have acquired are pleased to get this kind of sonic quality at a cost that is far below the larger facility's. The MG1212 has a sync track that is used for SMPTE. The SMPTE is run into a Roland sync box, providing the synthesizers with all the MIDI clocking, song pointers etc."

BETTER SONIC CLARITY

"Roger is very nice to have around especially when I encounter a problem. I once discovered a noise that was infiltrating many of my recordings. I told Roger about it and he later discovered that the mic inputs on the MG1212 were very noisy. He hooked up a a really good mixer so that the mics would go into that mixer and the line outputs from that same mixer would in turn go to the line inputs as opposed to the mic inputs of the MG1212. The noise was gone. With the Sony F1, I can now do things in the digital domain and retain a better sonic clarity. This digital capability enables me to make copies of my work, especially the ad work that I'm doing, that aren't a generation down. Basically I can maintain a much cleaner and noise-free master. The only thing that is still a hindrance to the sound quality of my studio is the fact that I am still using analog recording. When I go all digital in my cottage studio, I will be ready to interface to any outside facility and truly take advantage of what a bigger studio has to offer especially in terms of mixing."

Roger explains and predicts, "When synchronizable R-dats are on the market, your going to see a big increase in the amount of the cottage studios. Some will be in direct competition with

Figure 2. Connie, caught hard at work.

the big guns. Fostex is actually the first company to introduce a chase-locking R-DAT machine that works. Sony's isn't available yet. Anyway, lets say that you have two R-DAT machines that lock up and a stack of blank cassettes. You could do some overdubs on one of them and then do some overdubs on the second one and then mix them down to something to put on another reference tape. Then you can do some overdubs on the second one, mix them down to an F-1, put them back onto another reference tape etc. When your done you still have your individual little tapes...a tape with the horns and time code, a tape with the vocals and time code, and a tape with the drums and time code. Then you take all these little tapes and take them to a studio that has a stack of R-DAT machines. Lets say this studio has 16 R-DAT machines. You put your 16 little tapes in there, they all lock up and now you can transfer them to the 32-track. Then you mix. The point is that there has been zero signal degradation because you've been digital all the way. This is going to enable the cottage studio to become directly involved with the generation of consumer product (music). The cost of recorded materials should be expected to come way down."

"The major studios have been against this sort of thing. They think that the competition will hurt them when in reality they will benefit. What's going to happen is that everybody will make out. The high quality time or rather the expensive time in the studio, which is doing tracks and mixing, is the kind of traffic that the big studios will maintain. Nobody will be doing overdubs in the bigger studios anymore. They might go in to do original tracks and mixing. So now a higher percentage of the studio's time can be concentrating their efforts on doing tracks and mixing. The studio winds up selling their high ticket time while at the same time increasing cash flow per project. The studio will also have a greater amount of sellable time. Rather than hear this old story; '...well, we're sorry we don't have the time right now because so and so is doing overdubs for the next 6 weeks,' you might hear instead; '...sure, you can have time Wednesday when so and so is done mixing a song.' Those vocal overdubs that might take 6 weeks can be done in the little studio that has this R-DAT capability. Your album that you're doing partly at home and partly in the studio is cheaper and is the same quality as if you did it all at the studio. I believe that this will come about within the next year or so."

ACOUSTICS AND WORK SPACE

What kind of acoustical considerations have you had to make for your home studio. Connie replies, "You try to find the best room that will facilitate acoustics and the right kind of work space for the equipment. Roger and I are purchasing a house here in Nashville and I will be moving my studio to this new home. Nashville has some beautiful homes with high ceilings in quiet surroundings that cost much less than in other parts of the country. Nashville has already exhibited some of the trends that Roger has just mentioned. Not that there are R-DAT studios but there is a lot of digital recording here. Roger loves digital recording. As he says in his resume: 'I am a major proponent of digital audio for all aspects of audio recording, and employ that medium whenever possible.' As my studio upgrades, I hope to do the kind of recording that uses the studio for mixing instead of overdubs."

What else has attracted you back to Nashville? "When we first started this interview, I mentioned that about 10 years ago the pop music thing wasn't really happening here. Well, today it is. My music and the contacts that I had are stronger now than before. The industry, here in Nashville, is more receptive to more kinds of music than ever before. For anyone who is looking for a center of activity and environment for their creative lives whether as a song writer, musician, engineer or whatever, Nashville has much to offer."

GRAND PRIX PRODUCTIONS—THE POWER OF EXPERIENCE

Clyde Brooks and Howard Steele are weathered veterans of this industry. Both men have extensive backgrounds in the musically technical areas of endeavor. It was after a while that I discovered just how busy they have been since their teen years.

Clyde's credits include a host of gold and platinum artists for which he has done everything from writing their songs to producing their records. His music credits look like a *Who's Who* of country music...Lynn Anderson, Loretta Lynn, Dolly Parton, Barbara Mandrell, Ronnie Milsap, B.J Thomas, Kenny Rogers, George Strait, Oak Ridge Boys...yet he has reached beyond pure country into the realms of rockers such as Don Henley, Ted Nugent, Jason & The Scorchers and Johnny Winter. He has worked on and produced movie sound tracks such as for "The China Syndrome," "Tender Mercies," "Sweet Dreams," "The Soldier," and "Stacking."

Howard's engineering/production credits are equally impressive having worked with the likes of Diana Ross, Bee Gees, Chicago, Lee Ritenour, Steely Dan, Carly Simon, Alice Cooper, Leo Sayer, Larry Carlton, Joe Walsh, Manhattan Transfer...and the list goes on and on.

Howard tells me, "I started with a total fascination with electronics and music. As a teen, I built gadgets such as little mixers and played in bands. This all eventually led to a career as an engineer and technical person. Not too long ago I had my own company (Quantum Audio Labs) that designed, built and marketed consoles. I became engrossed in recording both my bands and other people's bands. This experience brought me to Los Angeles where there was certainly a lot of traffic (industry activity) and I found a niche there as an engineer. My work as a producer occurred simultaneously with the engineering gigs. I discovered that the people that I was engineering for were beginning to rely on my ears as well as my *techy* engineer abilities."

Did you have a background in electronics? "I started trying to teach myself from when I was about 8 years old. This continued until I was old enough to attend college where I finally got the formal education (a two and a half year program) in electronics. From there I began to work in the electronics field doing things such as working on the NI missile system, and with Bell Telephone."

It was this varied approach that enabled Howard to slide into the design field. Audio was his early love and it was that love that drew him back to the music industry. Howard continues, "When I do projects, I can go into a room and I'm not intimidated by the equipment. I just need to spend a few moments with an item and I can pretty much figure out how it works. There were many instances earlier on in my career where I went to a studio that was not 100 percent operable. It was in those more poorly-equipped studios that I really utilized my abilities to fabricate a usable studio out of one that wasn't really working when I arrived. Those studios were usually in rural areas where there was no alternative other than to make do with what was available and try to improvise...or rather, make good of a bad situation."

THE PLAYER NEVER DIES

Clyde, would you give us a glimpse of your background? "I started out as a musician and am still a musician. From the time I was about 9 years old I was fascinated by watching performers on TV and listening to records. At about age 14, I fell in love with rock and roll. The more involved I became with the music the more I wanted to know about the technical end of recording. Bands were great to watch but the more I studied on my instrument, the more I became aware of the immensity of the world of music and recording. From the time I got out of high school, I wanted to be a studio musician. I went to Berklee School of Music in Boston and at the same time was studying privately in New York with a man named Henry Adler. Many of his students were doing the studio percussion and drum work in New York. He used to send me to the studios with them so that I could watch and absorb. I sat in on many of Steve Ghadd's sessions...a truly unforgettable experience."

What and where were your early studio experiences? "I discovered, through the grapevine, that there was work for a rookie like me in Chicago. Although I was the new kid on the block, I found that I was not intimidated by the terminology and techniques. The sessions that I had observed really paid off in the sense that I had a head start resulting from having watched some of the best artists lay down tracks. I was fortunate while in Chicago because I was fairly young yet was in the main stream of recording work and found myself doing records sooner than I thought. After 4 years of session work in Chicago I moved to Nashville in 1974 where I continued to do the same kind of work, however there was much more work for me in Nashville. In 1978 I was given a chance to do some producing. In Nashville, unlike other cities, the producers and publishers count solely on the musicians and engineers to come up with the right ideas. There are many people here that have a good ear for a song but find themselves in a bit of a quandary when it comes to assembling their ideas in a recording studio. I discovered that there were others who were taking credit for doing things that I and many other musicians had been doing all along. It only seemed logical to me that I should begin to take credit for what I had done. It was just a matter of time before people started asking me to help them accomplish their goals. The first project that I did as a producer was with my ex-wife, pop-rocker Nancy Brooks. We went into the studio, recorded three sides worth of material and took the finals to Clive Davis at Arista Records and got her a deal."

THE NASHVILLE STIGMA?

What was the label's (Arista's) point of view about the Nashville scene and what were your choices of studios to work in with Nancy? Clyde remembers, "It was in the late 70s and there was really no power structure (industry support) for rock and roll in Nashville. Rock still had its stronghold in New York and Los Angeles. Upon meetings with Clive, he said, 'I don't care what you do, don't cut the record in Nashville. Go to New York or L.A. because I don't want the Nashville stigma attached to the record.' So we didn't do it in Nashville."What is the Nashville stigma? "Well...at that time, Nashville was strictly country music. Nashville had built its base upon country artists. What made it even worse were various appearances by country artists on TV such as the Tonight Show etc., and they would say things like, 'The musicians don't read music' and 'they don't do things that they do in the other cities' and so on. There were many hip things going on here, but you couldn't take a

Figure 1. Howard Steele and Clyde Brooks, Howard at left.

tape of a hard rock band, run it to a label here and get it placed. They were strictly for signing country acts. At the time I got Nancy signed this was certainly the case. Today however, the scene has changed somewhat. Nashville is stepping up to a more progressive mode. Just look at some of the studios here. They're quite impressive." That's true...mainly the reason why I came for interviews.

So now, at this point in time, you and Howard have established a production company which is the link between your creativity and the industry. Clyde answers, "When Clive Davis gave the green light to do the aforementioned record with Nancy Brooks, I had to decide upon an engineer. I had done a lot of listening to records and Howard's name appeared on the ones that I thought were particularly in line with what I wanted to do with Nancy. I called Howard out of the clear blue. Unlike so many friendships in this business that are built on a short term business relationship, my relationship with Howard has lasted over a 10 year period. Howard was in L.A. working and I was here in Nashville doing my projects. We always kept in touch and every year or two we found ourselves working together on a given project. About a year ago Howard decided to move here...a move that made our work and collaboration that much easier. At the same time Nashville, now, is progressing so much that people from New York and L.A. are flooding in here because work in either one of those cities is not the greatest. Ironically, Nashville is one of the last places where live musicians are working...five musicians looking into each other's faces saying, this feels good."

Howard says, "I must agree. As far as engineering goes, I don't necessarily want to sit in a control room with one guy with a synthesizer for six months making an album. I come from the same place as Clyde which is live players doing live things. At a time when the gear has become so sophisticated, one can easily become trapped by the very equipment that was designed to free us. The trick is to thoroughly understand the equipment and yet not become entirely dependent upon machines. If you can't improvise you can't really create."

BRINGING OUT THE BEST

Clyde says, "What we are trying to do with Grand Prix Productions is bring the best and the worst of our experiences to the industry. This *hard knocks* sort of experience is invaluable in pre-production as well as the latter stages of production. We are ultimately concerned with: the things that comprise a successful recording session, what makes a studio work, why you would want to work in a particular room, what kind of equipment you want to work with, and even—how to understand and work with musicians in a group. Bringing out the best in an artist and making the music bigger than life is the name of the game. All of this working knowledge is very valuable to record companies and artists. The important thing is that it's not a hit and miss project but rather a well calculated approach to getting the job done from start to finish."

Howard adds, "We can look at an act and we can see if they really have potential. We then plan out what we feel can be done with that act. We're not afraid to take the time to develop that act. Record companies, on the other hand, no longer spend time developing artists because they are looking for acts that are ready to cut. When we feel strongly about an artist or act, we carefully plan out everything that we feel is necessary in order to make it label ready...ready to record the album, nothing less. We can take an act and woodshed it, do the pre-production, even conceptualize or clarify an image."

Clyde says, "Most up and coming new bands don't know how to listen to one another. A typical scenario might be as follows: A given band might have been playing their songs together for many months. The drummer is doing his thing, the bass player is doing his thing, the keyboard player is doing his thing, the singer is singing, guitar, etc. When I ask the bass player if he hears what the drummer's kick drum is doing he might say, 'Well, gee, I never really noticed because I'm concentrating on the parts that I have to play.' When I ask the keyboard player if he hears what the guitar player is playing he might say, 'Well, the guitar is so loud that I have to really concentrate on what I'm doing or else I can't really hear myself.' When I ask the guitar player if he is listening to the vocal melody line he might say, 'I really can't hear the vocals at all.' The question that needs to be raised is: How can this situation be improved, assuming that this act has something special that is worth cultivating? The fact is that the lack of punch/impact might be the interaction between the kick and bass. The keyboards might not be nearly as effective as they could be because he is playing in exactly the same registers as the guitar. The guitar line might be conflicting with the vocal melody line because he can't really hear the vocal. Now I know that one might ask, 'If this band has so many problems, then what the heck are the redeeming qualities?' The fact is that there is a world of talent out there that only needs a little tweaking. Part of our job as producers is to catalyze the communication that already exists. The trick is not to drastically change anything but rather to enhance the character that is already there."

How do you deal with the defensive artist who isn't necessarily receptive to your suggestions? Howard responds, "If I hear something that needs to be changed, I'm ready to say why I think the change is needed. A suggestion to change the direction of a creative process has to be justified and qualified if you want people to be receptive to your input. On the other hand, one needs to know when not to make a suggestion. Sometimes it's not the right time to introduce a new idea. You have to be able to anticipate the right times to make changes. But without the backup (an explanation) you can't expect an artist to swallow your ideas."

Clyde adds, "With single artists, if they're using session players, producing is much easier. Howard and I have been in the business for so long that we both know who is the right player for a given genre of music. The session players show up on time (usually), play their part, take direction, their instrument sounds great and they're out of there. Bands are another story. With bands there's attitude, emotion, a whole rock and roll thing that's been there since the mid 50s. In order to help them realize their dreams you have to be able to tune drums or to adjust a guitar amp for a great sound that works in conjunction with the artist's style."

Could you share your views on the various recording mediums and how they correlate with particular projects? Howard explains, "I have to say that I'm willing to use whatever is available. However in the event that I really have a choice, I have to weigh out the strengths and weaknesses of a given medium. I like digital for certain things but have not yet recorded on a digital system that faithfully reproduces a *crunch* guitar. I have worked on systems that facilitate tamer varieties of music. Right now I'm equally at home working with analog and digital. If I was working a project that was mostly ballads and quiet sections, I would want to do it in digital just to keep the overall noise down. I also like to mix everything to digital...rock or whatever. But when it comes to the initial recording process, I think that the music and the attitude transcends whatever medium with which you are recording."

MUSIC AND PSYCHOLOGY

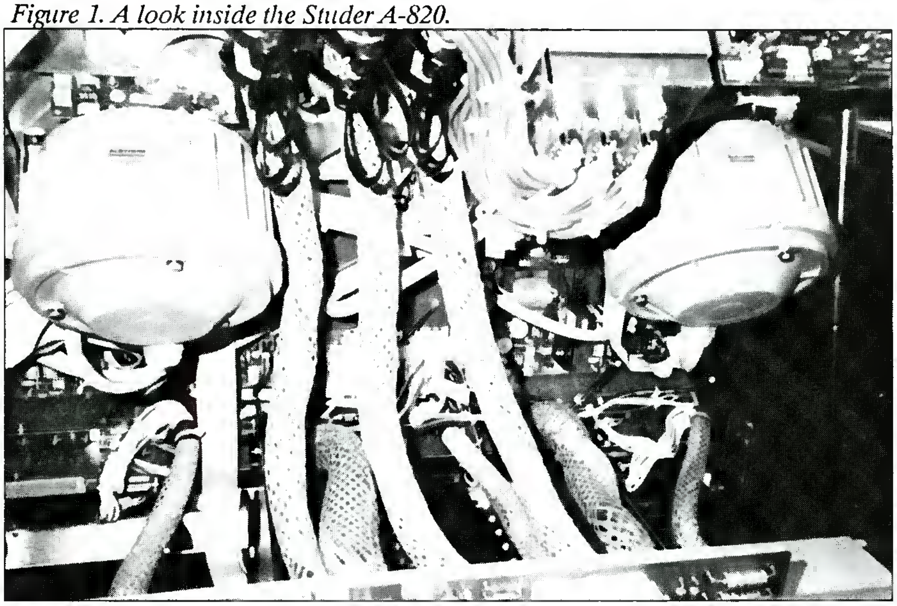
The producers' world can be very lonely and frustrating. Today especially, the producers are under more pressure than ever before. Clyde continues, "Most new artists are very naive when it comes to the music business. Many rock and rollers have become involved with rock and roll because they like the lifestyle...'I can dress like I want', 'I can be rebellious', 'I attract a lot of women', etc. Many of those rockers eventually become concerned with the music. They might get really serious and successful, but they haven't necessarily been exposed to the business world. They begin to find out that if you write songs, someone's going to want a piece of those songs. Someone's going to want to record you. Someone's not going to want to record everything that you do. A good stereotypical example is Warner Hodges with Jason And The Scorchers. He sat on my couch when he was 19 years old in pink spandex with one white dance shoe and one black dance shoe with a punk band saying; 'This is all nonsense man! I just wanna play...I'm into the Sex Pistols and this business is all garbage, talking about all this business stuff and songs...I just want to play guitar.' So in a case such as that one, you become the manager, the booking agent, the psychiatrist, and the music producer. It's one thing for a label to say: 'We signed these guys so go and see them. If you like them, see if you guys get along and we'll do the album.' But it's another thing to go into a punk club and say: 'Howard and I like what your doing and we'd like to help you get a record deal. You guys obviously want to do more than you're doing, how can we help you do that?' There's a real void there. You've got to establish credibility, you've got to make them respect you, you've got to make them feel that you understand what they are about, and you've got to be honest with them. You've got to be able to somehow turn them around, be a friend to them and give them some real hope along with a taste of reality."

"Country music has made a very important point in the world of music today. In a sense, country music is about a guy that just got off from work, makes $200 a week and is at a bar with a beer in his hand, listening to the jukebox. If that song on the jukebox doesn't convey a relatable message to this guy, then the song is meaningless to him. It won't matter how great the sequenced synthesizers are,...it won't matter how big the drum sounds are. There must be that basic, certain something that strikes a note of emotion. This is the lesson that country music has taught me. Today, in my work, I still haven't forgotten that a certain something still must be there. Even though we have the greatest machines in the history of recording, nothing makes up for a lack of emotion. At times we all notice things that seem to be on the verge of being very noteworthy. The fact is that there is still a wealth of creativity in the world and the producers' responsibility is to make a connection between the creativity, machines, business and ultimately...the listeners, who by the way, are the reason why we are here."

STUDER REVOX—A WORLD CALLED STUDER

My visit to Nashville certainly would not be complete without a visit to the Studer headquarters. For those of you who are in some way intertwined with the audio industry, and can still appreciate the earliest Packard (Yes! The car) and the latest BMW, chances are that Studer, for you, is a household word. If your history (like mine) doesn't go back quite that far, the chances are still very good that Studer is a household word. 1988 marks the 40th year for Studer Revox as a leader and innovator in the pro-audio industry. Maybe the company's success

Figure 1. A look inside the Studer A-820.



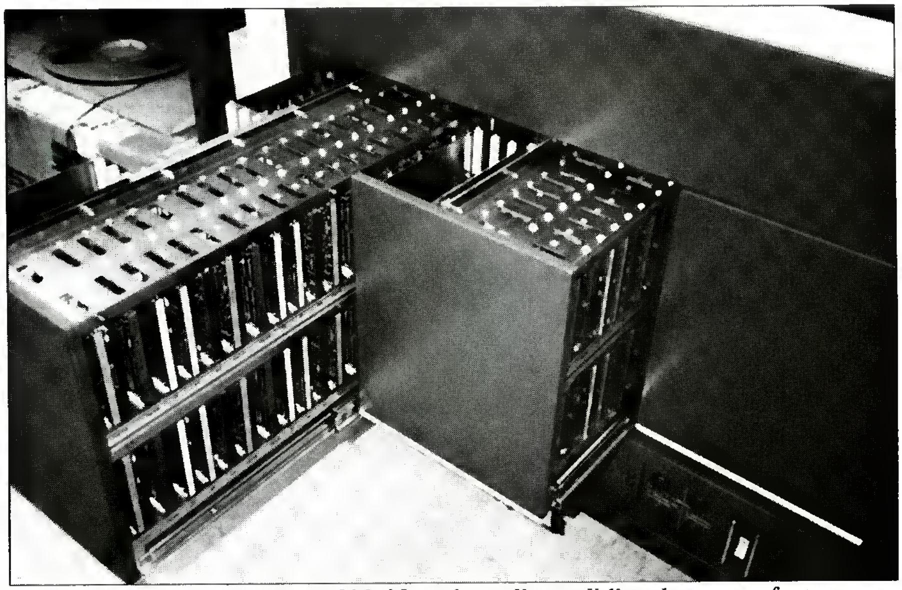


Figure 2. Again, inside the A-820. Note the pullout, sliding drawers, a feature found on most Studers.

lies in Willi Studer's quiet kind of genius, or in the company's commitment to its goals of maintaining the highest degrees of excellence in everything they have ever manufactured. Maybe it's the powerful line-up of personnel and technicians. Whatever it is, Studer is certainly a prime example of a company's products' ability to speak for themselves.

On January 2, 1948, Willi Studer and two other men began the Willi Studer company. With a staff of three, the purpose of the company was to produce special oscillographs for high-voltage test bays. It was at this time that Willi began his investigations concerning a.c. motors and the development of a tape recorder for home use began. In 1949 Willi successfully produced this machine under the name "Dynavox." This research and development led to a professional tape recorder for broadcast applications. In August of 1951 at the International Music Festival in Lucerne, a professional prototype was introduced and utilized in cooperation with Radio Studio Basle. Willi says, "These were the milestones of the early years. The staff of three expanded quickly and many of the employees were as enthusiastic as I was. In retrospect this was the most exciting period of my time as an entrepreneur."

TAPE MUSIC

My first exposure to the Studer aura was 14 years ago at the State University of New York at Albany where I was a technical assistant to John Cage in his assemblage and performance of Bird Cage, an environmental piece utilizing the simultaneous play of 8 open-face, quarter-inch, half-track tape machines. John Cage was probably the first musician to use a tape recorder as a compositional device. (There is much to know about Cage's background, and a good place to start investigating is in the Who's Who Of Music.) He was known to cut tape that had already been loaded with previously recorded

Figure 3. A procession of Studer's recording technology tape decks. From left to right, Studer A-820 two-track, ReVox B77, DASH format digital machine, A-807, A-807 rack mountable, and a corner of a 900 series console. Founder and chief executive, Willi Studer's portrait hangs on the wall.

material into thousands of fragments only to splice it all back together (often arbitrarily) into one continuous piece. This process became an aesthetic and to many people around the world, Cage became the "father of tape music."

Of course, when doing this kind of tape work, which involved some very strange and daring splicing techniques, the type of tape recorder that is used is important. The heads and transport system must be capable of withstanding this unconventional usage. The Bird Cage piece was taken on the road and I accompanied John Cage as his technical assistant. Those 8 tape recorders were ReVox A77s. I wondered how long those machines would last. They were already kind of beat up from many students' hands having been all over them. God only knows how many times the machines had been used by non-technical people. We cleaned and demagnetized the machines before every performance. The key to the piece's impact was that it should sound like you were there. Most of the sounds were of a concrete (actual acoustic sources) nature, so the reproduction of those sources on tape had to be as clean and consistently responsive as they could possibly be.

Joel Chadabe, the Director of Electronic Music at SUNY Albany (my instructor for 4 years) insisted that the A77 was the machine for me to use when I performed *my* electronic music for the University community. I used the A77s for everything from live tape feedback delay, to long loops stretching 20 feet and more. While at SUNY I witnessed a colleague who dropped one of the machines, cracking a huge chunk of gray plastic off the casing, exposing one of the transport motors. That machine was operated for nearly a year, with the motor exposed as a reminder to the students that they should avoid being clumsy and careless.

Upon arrival at Studer, I was greeted by Charles Conte (public relations manager), Bill Muggler (executive vice president), Doug Beard (director of technical and marketing services), and David Bowman (director of professional dealer products). The five of us went to lunch where we exchanged anecdotes from our respective audio experiences. Following a very un-businesslike lunch, I took a tour of the Studer facility with Doug Beard as my guide. The highlight of that tour was an in-depth orientation with the new A820 24-track analog. We began the tour in the conference room where Doug familiarized me with the entire line of Studer and Revox products. Their consumer line reflects the same strengths found in the professional line.

SOME HIDDEN HIGHLIGHTS

We exited the conference room and proceeded on to 'shipping and receiving.' "Not very interesting," said Doug. Actually, though, there was something that I thought *was* interesting. Doug told me that every machine that is manufactured abroad and received in Nashville, following QC (quality control), is repacked. Moving parts are anchored down, still other parts are removed and packed separately. The machines are then re-crated. Why not just send the units out just the way you get them from overseas? "We have to know, first hand, that everything that goes out of this Nashville station is working perfectly." Doug replied. We then entered a room where many of the larger tape machines were sitting, looking like a huge traffic jam of multitracks, some with reels of tape threaded up. This is the QC room. Every machine is burnt-in and electronically measured. It was at this point that I noticed some very elaborate test equipment.

Much of Studer's service and QC areas are complimented by customized arrays of test gear that facilitates the work to be done on a specific machine. Some of these special arrays of test gear are on wheels. One such array consists of a computer (IBM) that interfaces with a Sound Technology analyzer. All modes of signal comparisons, distortion, frequency response, etc. can be performed with this mobile station. Another mobile array that caught my eye consists of a mil-spec signal generator, digital and analog VOM, frequency counter, and the old reliable Hewlett Packard HP 27A dual-trace, delayed-sweep oscilloscope. Yet another mobile unit consists of a very high frequency Phillips O-scope, signal generator, dB metering, amplification and a set of stereo speakers. Basically there are mobile stations for analysis and spec'ing, troubleshooting and basic repair.

The bench stations are somewhat different. At these stations one notices a uniformity from bench to bench. One can see that it was necessary to standardize the repair process. If one technician is tied up with a piece, another technician can work from the same point of reference with the same troubleshooting gear. This certifies consistency in results. It is at these bench stations that most of the smaller gear is repaired. One tech says to me as I ask him what he is working on, "It's pretty rare that we see one of these (an older Studer cassette deck) with a bonafide electronic problem...but then again this machine is 10 years old. We tried to get the customer to buy newer decks but he won't part with these. He has two of them. Here's the other one that he sent us to be tuned-up. Usually we get 10 and 20 year old tape machines that, electronically, still work but something of a mechanical nature has worn out."

Doug informed me, "Even in the case of a very old machine, we usually have every part right here on hand." In the shop, there are a couple of component chests (about the size of a mechanic's huge tool chest) with hundreds of compartments that, according to Doug, "...contain virtually every component that is found in the smaller gear and decks."

Our next stop was the real parts inventory...another room dedicated to nothing but parts and more components. I chuckled as I gaped down aisle after aisle. Doug told me, "Even in the event that we don't have an item here in Nashville, our overseas traffic is so good that we can have anything, even the most obscure item in a matter of hours." Doug allowed me to explore this room and I spent nearly 20 minutes opening drawer after drawer...just looking. A lot of this stuff appears to be mil-spec. "Many of the components are mil-spec." For those of you who are unfamiliar with electronics jargon, mil-spec refers to a level of quality control and tolerance that is utilized by the military. Typically, mil-spec items are rated at twice the normal tolerances. This is not to say that everything at Studer is built to military specifications, however, I think one can get a flavor for the level of quality that permeates the Studer Revox products.

THE LEADING EDGE

The last stop on the tour was a run-through on the new A820 24-track analog. This led to a discussion about that age old dilemma: DIGITAL vs ANALOG. With the Studer Digital Audio Stationary Head (DASH) format recorder a reality, why has Studer put so much research, development and energy into another analog machine? Doug told me, "Studer is in no way a newcomer to the world of digital electronics. As a matter of fact, some of the big digital manufacturers have based their findings and research upon discoveries that were made by Studer long ago. Most of these manufacturers are still doing their R&D based upon comparisons that are still being made with the Studer analogs as the control (point of reference)." Even as we speak, someone, somewhere is comparing a digital machine against an analog Studer. The fact is that Studer is very much involved in some of the world's most leading edge, digital audio research.

So...back to the A820. This machine is actually impressive just seeing it. Sounds corny, I know, but lets see what this monster has to offer. Upon opening up the machine (an easily accomplished task) these massive blue motors were staring me in the face. These powerful motors, in conjunction with sophisticated servo control systems, enable tape spooling speeds of up to 50 ft./sec. with extremely fast acceleration and deceleration. All of the electronics are wonderfully accessible. The audio, control, and noise reduction circuitry are to be found in pull-out drawers...a Studer tradition. The capstan motor has its own dedicated microprocessor control for very smooth starts and very gentle tape handling. Reverse play is a stock function along with 3 tape speeds. All transport operating functions are user programmable and offer a choice of more than 40 functions, which are all assignable from an internal software library. Digital control systems govern practically all audio processes. Audio alignment parameters can be set for all 24 channels simultaneously and automatically! Digital memories store alignment parameters for two tape formulations as well as for 8, 16, and 24-track head blocks. There are special D/A converters to optimize the erase current on each track. Ready for this?...All communication between the overbridge and the transport deck is by serial data exchange. This serial data communication allows for remote placement of the overbridge display and is absolutely functional up to 300 feet away from the deck with connection via a single 4-conductor cable. Of course there is a Dolby SR option.

Digital technology has many faces and Studer has already added a few new ones. With a great deal of experience in digital controls and high precision synchronization that has been gathered over many generations of tape recorders, Studer has clearly proven to the industry that their experience will prove invaluable in the development of interdisciplinary control, automation and editing systems. The decision to go digital or analog has never arisen for Studer. The refinement of analog technology proves that consoles and tape recorders with integrated noise reduction systems are well suited for PCM applications. At Studer, analog audio technology always has a future. db

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Sound Reinforcement in South and Central America—Part III

ED LEARNED

This is the further adventures of our peripitatic sound-reinforcement author. Nor is this the end of them!

Sunday, March 29, was the first of two performance days in Managua. The daylight gave us our first good look at Managua, which was kind of depressing. Most of the city had been destroyed by a massive earthquake in 1972, and the ruins were never rebuilt; most of the relief funds were skimmed by the Somoza regime.

It's as though a bomb hit the city, destroyed most of the buildings, but left all the people. We held our briefing at around 10:30am, and the first bit of news was good. Wade's guitar had been found in San Salvador; for some reason it had been off-loaded there.

It would be sent over on the evening flight, so it might arrive in time for the second half of tonight's concert. For at least part of the show he'd have to play the borrowed guitar, which turned out to be a Gibson Birdland. PA information was not as promising. Our first two concerts would be outdoors for 2000 capacity, and I was told that 4 Peavey SP-1 speaker systems and 2 Peavey CS-400 amps would be provided. This was not sufficient for our needs, and didn't even meet the minimum requested in my advance cable.

After a certain amount of complaining, I was promised 8 SP-1s and 4 CS-400 amps, which I felt would be adequate. Set-up and sound check was scheduled for 3:30pm, to spare the equipment and us the worst of the day's heat, which was oppressive. I elected to go with Pauline, Jim and José to the concert site to have an early look; the band preferred to spend their time at the Casa Grande, a huge colonial-era mansion that was formerly home to the U.S. ambassador.

It had since been converted into an American visitors guest house, complete with pool, recreation room and snack bar featuring U.S. cuisine.

The Grand Hotel Ruins were the remains of the best colonial hotel in Managua; all that remained were the foundation, ground floor, and some of the first floor. The "stage" was a concrete slab, centrally located in what was once a courtyard. The audience area was a series of terraced concrete slabs with folding chairs, protected by the remnants of the first floor (Figure 1). The a.c. power came from a panel built into the wall off-stage left; it supplied 220 V power, but had no equipment ground. I noticed a metal water spigot about 15 feet from the panel, so Jim went to procure some wire that I'd tie to it for grounding. After joining the band for R&R at the Casa Grande, we all returned to the ruins at 3:30 as planned. To our surprise, no PA equipment had arrived. We set up as far as we could with no sound, and I sent the band back to the hotel while Lee and I waited. When the band returned at 7pm, there was still no PA; it finally showed up at 7:15, and consisted of only 2 SP-1s and a single CS-400 amp. Lee and I were very upset, and made it clear that without the extra PA we'd been promised there would be no concert. No vehicles were available for this, so our band bus was dispatched for the extra stuff, returning at 8pm, the time our concert was scheduled to start. The crowd was let in at this time, so we weren't able to sound check. It was a real "wing it" situation, and, while the band was incensed at the treatment we'd received, they gave their best and we pulled it off.

Monday featured another performance at the Grand Hotel Ruins, this one scheduled to be broadcast live by Radio Sandino. We had most of the day free, with a sound check scheduled at 4:30pm. The group was the talk of the town, and tickets were like gold as the show quickly sold out. Our day was much more smooth: we enjoyed a productive sound check (*Figure 2*), and I had a real chance to tune the PA, resulting in much better articulation on the vocals. We had some invited guests for sound check, actors and actresses from the Alex Cox film "Walker" who shared our hotel, and had befriended the band. With a packed house ready to rock, Wayne and the guys played an inspired set; the rock chestnut "Let the Four Winds Blow" ignited a dance-a-thon that swept the entire crowd! I gave the radio a straight house mix split, and was complimented on my mix by several USIS employees who'd listened in on their car radios. After the show, the group and various movie cast members crammed into several hotel rooms for a jam session that lasted until the wee hours of the morning.

AUDIENCE INSPIRATION

Tuesday's schedule began at 10am, when our bus arrived to drive us the 1-1/2 hours to Granada, where we would play an outdoor concert. My major concern here was again PA size: I'd requested a system of 3000 watts minimum for this show, and Pauline had been calling around for the last two days trying to locate enough gear to do it. We stopped in town for lunch, a visit with our movie friends, and a quick look at the set of "Walker," which dominated the center of Granada. Around 2pm we took off for the Plaza Xalteva to set up. This was a large dirt square in front of the local militia barracks, across the street from a large church. A temporary stage had been set up in front of the barracks, and some of the PA equipment had arrived. I suggested that sound wings be built to raise the PA to a level over the audience's heads, and two were fashioned out of wood, with 50-gallon oil drums providing extra height. The PA was a real mixed bag of Peavey PA and instrument cabinets, along with some custom studio monitors and generic bass amp speakers (*Figure 3*). The whole conglomeration was powered by Peavey CS-400, CS-800 and Yamaha 2200 power amps. The a.c. power came from the barracks: I wired my transformer tails directly to the building's power panel, and got my equipment ground from a water pipe on a sink near the panel. The concert proved to be a special one; the word was out about the group, and over 10,000 people showed up, filling the square and clogging the streets around it with a solid mass of humanity. The Nicaraguans cheered wildly whenever the U.S. was mentioned in announcements, showing all of us how they really felt about America. The band, inspired by the great response, played a passionate set that had the crowd dancing all evening. The friendship displayed by the people of Granada gave everyone a good feeling.

Wednesday, our last day in Nicaragua, involved a concert outside on the grounds of the Casa Grande. The band would play a shorter set for an in-

Figure 1. The stage and seating area seen from the mix point, Grand Hotel Ruins, Managua, Nicaragua.

vited audience comprised mostly of the diplomatic community of Managua, Sandanista officials, and various members of the press and international development agencies. I'd made arrangements with the sound guys in Granada to use some of that gear at this concert, so for a change here I knew that everything was covered. Two Peavey 1210 HS cabinets per side powered by a Yamaha 2200 amplifier comprised my PA for the day. I ran power from inside the Casa, tapping into a wall outlet with functional equipment ground. The evening was essentially a large cocktail party cum entertainment, and proved conclusively that ideological differences are soon forgotten when everybody elbows up to the same bar! The press corp were the first people up and dancing, and the rest of the audience soon followed their lead; a good time was had by all.

INTO HONDURAS

Thursday, April 2, was our travel day to Honduras. Equipment and baggage were collected the night before and taken to the airport by embassy staff to expedite our departure. We left the hotel at 6am, allowing plenty of time to catch our SAHSA flight to Tegucigalpa, which only took a half-hour of real flight time. Once we'd collected our gear and cleared customs, a process that proved painless, we went immediately to our hotel and caught up on our sleep. At 4pm, Lee and I were picked up and taken to the "Manuel Bonilla" Theater, the National Theater of Honduras, to have a look and meet the PA guys. We'd play a concert here on April 5. The capacity was about 750, arranged in the European opera house tradition, with three balconies that wrapped completely around the floor seating area. Reverb time was around 1.5 seconds. Engineer Carlos Rubio owned a small studio near by where the PA equipment was stored, so we drove off to have a look. I picked out 2 Peavey SP-1 cabinets and 2 CS-800 power amps, and also asked for extra cabinets to augment these. He agreed to come up with the extra stuff, and gave me a tour of his studio. He eventually played me some masters that he was working on, and I helped him come up with a quick mix. CAO Lois Mervyn prepared a delicious dinner for us that evening, and we used the informal setting to relax and discuss our Honduran schedule, which involved three concerts over the next three days in three different cities.

Friday's concert was in San Pedro Sula, about a four hour drive from Tegucigalpa. The drive was very scenic, winging through mountains and valleys lush with vegetation. We broke up our journey with a lunch break at Lago de Yojoa, a large lake about halfway on our journey. We pulled into town around 2:30pm, and after checking into the hotel we all went over to the concert site. The Centro Cultural Sampedrano is the home of the USIA Bi-National center in San Pedro Sula; we would perform in the building's auditorium. It held around 500, taking into account extra seating added for us. The floor had a rug, but the hardwood walls and reflective ceiling made for a reverb time of almost 2 seconds, with some discrete early reflections as well. The PA was quite large, but featured a real mixed bag of gear. Per side there were 2 Acoustic full-range cabinets, each with a 12-inch woofer and single horn; 2 Acoustic guitar cabinets, each with 4 10-inch woofers; 1 Acoustic guitar cabinet with 2 12-inch speakers; and 2 diffraction horns with 1-inch compression drivers. This whole conglomeration was run full range except for the diffraction horns, which were passively crossed over. 2 Yamaha 2200, 1 Peavey CS-800, and 1 Crown DC-300A provided power for the PA. My a.c. power came from wall outlets stage

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left; the voltage was 110 V, but the grounds were non-functional. I tied to a toilet pipe off-stage-left for ground. I'd been told that these outlets supplied only 15 amps, so I insisted that my outlet be used only for our stuff, with the PA equipment tied in on a different outlet and breaker. The PA was quite out of balance, with lows and mids dominating; I managed to partially tame it by turning down the amps powering the full-range stuff. I still had to seriously EQ to smooth out the mid-range area, which remained overly dominant. Our 8pm concert was very well attended, and the crowd got right into it, clapping and dancing in what little space they had. I was concerned about the load on the a.c. system, and my concerns were well founded: about halfway through the show the PA system cut out. Since the band equipment was on a separate outlet, we were not affected, but Wayne had to stop the show while the PA guys located a third outlet to split their load. Once done, we finished the show with no further trouble. Before we packed up, the band adjourned to an upstairs room for a reception in their honor, where they got to meet the audience, sign autographs and pose for pictures.

IN SEARCH OF GEAR

Saturday's concert in La Ceiba necessitated another van drive; this one only about 2-1/2 hours long. The land was more flat and devoted to growing fruit: this was banana country! La Ceiba was right on the Caribbean in the heart of the banana region. Standard Fruit Company, which operates a large facility in La Ceiba, would sponsor our concert here in a small auditorium at the Camara Junior. We arrived at the Standard Fruit guest house around 11am; while the band got settled, Lee and I went off to check out the hall with Margie Dip, Standard Fruit's representative. It was a small auditorium/gym, all linoleum, with a reverb time of 4 seconds, just horrible for music. Seating was rows of folding chairs, already in place, in front of the stage, which was at one end of the building playing to the long dimension of the room. The PA proved to be another big problem: all we had was a single Perkins-type cabinet per side, each with 2 JBL 15-inch woofers. Amplifiers were 2 Peavey guitar brains. The PA had no horns, and as a result no high end, a situation which was totally unacceptable. Margie took us to the local "rent anything" place in search of more gear; we procured 2 small Peavey cabinets that contained a 12-inch woofer and horn, as well as a Kustom 200 watt PA head. I planned to use the little Peavey speakers as monitors, so that Wayne's Peavey International cabinets could be used to augment the existing PA. We joined the band for lunch, then all returned to the hall to set up. With careful EQ, I was able to get enough top end out of my makeshift PA to maintain intelligibility in this very reverberant hall. The a.c. came from U.S.-type ungrounded receptacles on the back wall, which sup-

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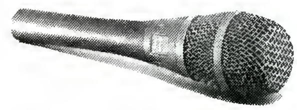

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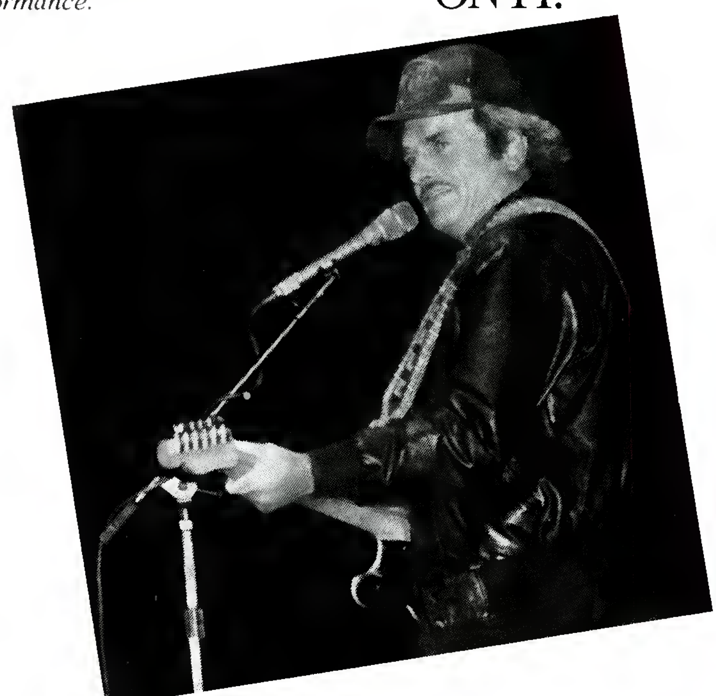

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Figure 2. Wayne Toups and Zydecajun Band at the soundcheck for the second show at the Grand Hotel Ruins in Managua.

plied 120 V. I tied my ground wire to a water pipe outside the building about 25 feet from the "stage door." The concert began at 7:30, and we had about half a house, around 350 people. I'd refrained from mic'ing the drums, and amplified instruments only on solos, having the guys watch their stage volume. By concentrating on accordions and vocals, I was able to maintain a marginally coherent mix in this very reverberant field.

Sunday required a return to Tegucigalpa, capital of Honduras, for our concert at the National Theater. After our 5-hour drive, punctuated by another lunch at Lake Yojoa, we got to the hall at 4 to set up. Carlos had come through: my PA was comprised of 1 Peavey SP-1 and 2 Peavey SP-3 cabinets per side. Power came from a board off stage left, which supplied 220 V power. Again, there was no ground available, so I located a bathroom at the rear of the hall, and ran a wire to the water pipe there. The Peavey power amps, however, developed some problems, with either the left or right sides cutting out during sound check. I had Carlos go and bring in another amp, which was a Yamaha 2200, to cover for the bad Peavey amp channels. I needed full power, as our 8pm concert was sold out, damping the room even more. It was by far the best sounding concert in Honduras, and the band was finally able to "play out" without the volume constraints posed by the other venues. The crowd was very demonstrative in their approval, with a tour group of kids from Christian universities in the southern U.S. leading the cheering section.

RECEPTION IN EL SALVADOR

Monday, April 6, meant it was time to move on, travelling to El Salvador. We left our Tegucigalpa hotel at 6:30am to head for the airport and our 8am TACA flight to San Salvador. The flight was only 45 minutes; we were met by PAO (Public Affairs Officer) Jake Gillespie and CAS Beatriz de Cortez. They assured a quick transit through customs, and acted as guides during the half-hour drive into town. The weather was gorgeous, with clear skies and sunshine, so many of us opted to spend the rest of our day at poolside. I had an afternoon meeting with the local sound people, where I determined that we'd be well covered here. That evening, Jake and his wife held a reception in our honor where we got to meet José Napoléon Duarte, President of El Salvador, and Virna Passelly, Miss El Salvador. At 10pm, the reception crowd thinned out; they hadn't left, they'd just crowded into Jake's upstairs study to watch the Leonard-Hagler fight, led by Wayne, who is a serious boxing enthusiast.

Figure 3. A view of the stage and p. a. system during soundcheck at the Plaza Xalteua, Grenada, Nicaragua.

Tuesday was our first performance day in San Salvador, with a 6pm concert scheduled at the National Theater. We all went over to the theater at 1pm for set-up and sound check. The hall was beautiful: it seated 1000 in a large floor area with padded seats and thick carpeting. The floor area was surrounded by private box seats, and there were two balconies, also with some private boxes. Reverb time was just over 1 second, due to the absorption of the plush appointments. The PA was already in place: 4 Peavey SP-3 powered by 2 Peavey CS-800 amplifiers per side. We also used 2 additional SP-3 cabinets as side fill monitors, run off the monitor output of Wayne's 600-B. That output fed a Yamaha 1/3-octave graphic EQ and another Peavey CS-800 amplifier. The a.c. came from a U.S.-type receptacle, with ground pin, on the wall off-stage right, supplying 125 V. However, the ground pin was non-functional. I discovered that the conduit run to this outlet was grounded, so I simply tied my ground wire to it. The dead acoustics coupled with the powerful PA made this room a dream to work; it was the best sound on the tour to date. The concert was scheduled to be taped by Salvadoran television, so I provided a straight PA feed for them: in this room it would sound great. The major difference in this concert was security; 10 men armed with automatic weapons stood guard outside, with another 6 stationed inside. We couldn't help but notice the extra security precautions, such as never taking the same route to the theater, and having an armed guard with us in our vehicle at all times. Such is the way of life in El Salvador; with guerrillas on the left and terrorists on the right, both Jake and the Ministry of Culture wanted to take no chances with our concerts. Our first was a smash from the start (*Figure 4*): the capacity crowd went wild, dancing up a storm as the TV guys struggled for un-obstructed camera angles.

SURPRISE FOR ED

About three-quarters of the way through the show, Wayne called me up on stage; the monitor amp had cut out. While I was bent over, investigating the connections, wild applause made me turn around. Bea was bringing a cake out on stage, and I realized that she and the band had conspired to get me up there to celebrate my birthday! Wayne and the band led 1000 Salvadorans in singing "Happy Birthday" to me, and after making the obligatory first cut I returned to the console and finished the show. It was a complete surprise, and many in the audience stopped by to shake my hand and wish me well on their way out. Definitely a night to remember!

Wednesday's show in San Salvador was outdoors at the National Museum. The group set up on a concrete slab located on a low hill directly in front of the museum. My PA system was 8 Peavey SP-3 and 3 CS-800 power amps per side. We also used the same side fill set-up from the National Theater. The weather was just perfect for an outdoor show, although those without suntan lotion paid dearly. My biggest problem here was a.c. A power board was dropped by the slab, stage left, for us; voltage was 125, and I got a ground from a sprinkler pipe near the slab stage left. During sound check, the voltage dropped from 125 V to 100 V on peaks, with the system distorting badly every time it did. At my request, extra lines were run into the museum to power the PA system amps and the lighting system from a separate outlet. It still wasn't enough: during the show my power dropped to 107, which was within acceptable limits, but the lights would go dim with attendant PA distortion on peaks, so I was forced to mix very conservatively. A separate service for lights might have helped, but it couldn't be procured on such short notice. It was the first time ANY musical event had been held here, so I chalked it up to venue growing pains. The crowd didn't seem to mind—young Salvadorans turned the grassy area behind the seats into a dance floor. Security was again heavy: the street in front of the museum was blocked off at both ends by armored personnel carriers, and there were plenty of armed soldiers around.

IN WAYNE'S ABSENCE

Thursday, April 9, found us on a rare afternoon flight, bound for Guatemala. Upon arrival in Guatemala City, CAO Bob Gibbons and his staff whisked us through customs and to our hotel, where after a few media interviews we enjoyed a free evening. Friday involved a 4-hour drive through very scenic mountains to Quezaltenango; we stopped at Lake Atitlan, a gorgeous lake framed with volcanos, for a bite to eat on the way. We arrived around 1:30pm; after a short rest we went over to the hall to set-up and sound check. The Municipal Theater seated approximately 800; it had lots of hard surfaces that contributed to a reverb time of 2 seconds. The PA was of the component variety, with 2 horn-loaded 15-inch woofers, 2 12-inch woofers in Perkins-type enclosures, and 2 60-degree radials per side. Peavey CS-800 power amps and some home-made amps provided PA power. The system was electronically crossed over, and by rebalancing the sections I was able to

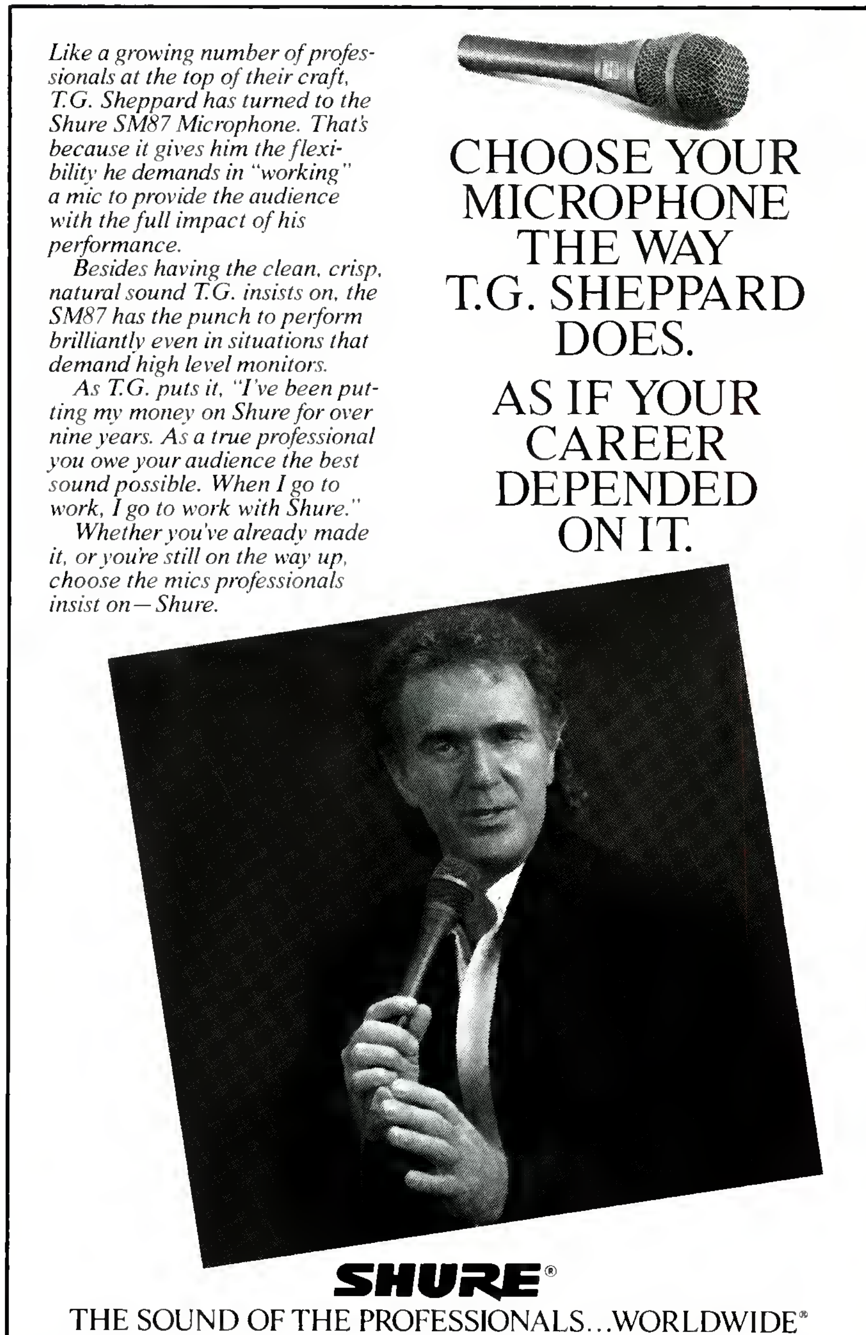

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Figure 4. This view of the band on stage was shot from the balcony at the National Theater in El Salvador.

smooth out the system, which had sounded incredibly harsh at first listen. After a brief sound check, we returned to the hotel to rest before the 8:30 concert. Wayne wasn't feeling too well, so everyone else went over early, with a vehicle going back to get him just before show time. The band slammed through the first four songs, to the delight of the audience. Wayne left the stage during Waylon's featured vocal on "Louisiana Man," and didn't return. He'd become very ill, so much so that Bob had taken him to the local hospital. The band played the rest of the show as a quartet, which resulted in some on-the-spot arrangements of songs; the guys pulled them off flawlessly. Waylon and Wade handled the vocal chores in Wayne's absence, and the crowd was treated to several scorching blues numbers featuring Wade's guitar gymnastics. The audience loved it, even calling for an encore, so the show was a success. We were all worried about Wayne.

Saturday, our last work day, began with the long ride back to Guatemala City. Wayne was still in a weakened state: he'd been diagnosed as having been victimized by amoebas in his digestive tract. The only thing that he was concerned about was making the evening's performance; with medication, rest and lots of willpower he might possibly do it. When we arrived in Guatemala City at 1pm, Wayne went directly to the room to rest up; we would sound check without him to spare his energy. The rest of us went over to the Instituto Guatemalteco Americano to set up. The auditorium there seated only about 450, in a single seating area with very steep raking. The a.c. power came from off stage left; I tied directly to a power panel that could provide either 110 V or 220 V power, with a functional ground on the panel. My PA was 2 Peavey SP-1 cabinets per side, powered by a single CS-800 amp. The hall was ideal acoustically: the carpeted floor, and acoustical treatment on the ceiling kept the reverb time to around 1 second. Sound check was smooth and quick; the guys decided to keep Wade's blues numbers in the set to give Wayne a little breather every few songs. We returned to the hotel, and I returned to the room to see how my roommate was doing. Wayne looked much better; the medication was starting to work, and he'd actually eaten something. When the show started at 8pm, Wayne really showed what he was made of, spinning and dancing all over the stage as though he'd never been ill. The capacity crowd ate it up, and brought our tour to a happy conclusion by calling for several encores. Wayne commended everyone for their wisdom in planning the set list: by spacing out Waylon's and Wade's featured tunes through the set, he didn't have to play more than three songs in a row, which gave him the chance he needed to catch his breath.

Figure 5. Wayne at the National Theater, El Salvador. Note the accordion mounted mic pickup.

TO MIAMI, THEN HOME

Sunday, April 12, marked the end of our jaunt through Latin America. Bob, Lee and I went to the airport early to handle the various shipping arrangements. My gear was separated and taken to Pan Am Air Cargo, where it would be sent to Detroit as air freight. The band's equipment would travel with us, as accompanied baggage, to Miami. We left Guatemala at 11:30am, arriving in Miami at 3pm. At that point, we split up: the band and their gear went to New Orleans, Lee Cross and his baggage to Washington D.C., and me and my stuff to Detroit. We met as strangers; we departed as friends, bound by the experiences of shared music and friendships with the wonderful audiences of South and Central America. db

Artist Accommodation

Sports! Almost everybody loves to either play them or watch them. Working in the live sound industry is a sport. It is a demanding one, requiring strength, stamina and the patience of a saint. If you want to win, you'd better learn how to play the game and the game is cooperation and artist accommodation.

Starting a business in any industry is rough. The Department of Labor produces statistics saying that less than 1 in 10 businesses starting out today will still be here in 5 years. So, with three strikes already against them, why do people go into a business where the rewards are long hours and no respect? Because they love music, and hope to have fun. I can't help you love music more, but maybe I can make it easier to have fun.

PEOPLE COUNT

The basic goal in the sport of sound reinforcement is to be able to think clearly and quickly while a world-famous recording artist and 10,000 fans are breathing down your neck. Many companies start out in garages with this dream, yet few make it onto the stages of arenas and festivals. One company that has is Modular Sound Reinforcement of Princeton Junction, N.J. Why?

In this business, personality has as much to do with success as anything else.

No company, no matter how intelligent its founder, survives the rigors of business without a core of attentive personnel. Any sound company that has survived more than 10 years is going to have had some personnel changes. The most important act of any business owner is to select people who share the same dream and have the commitment to make it a reality.

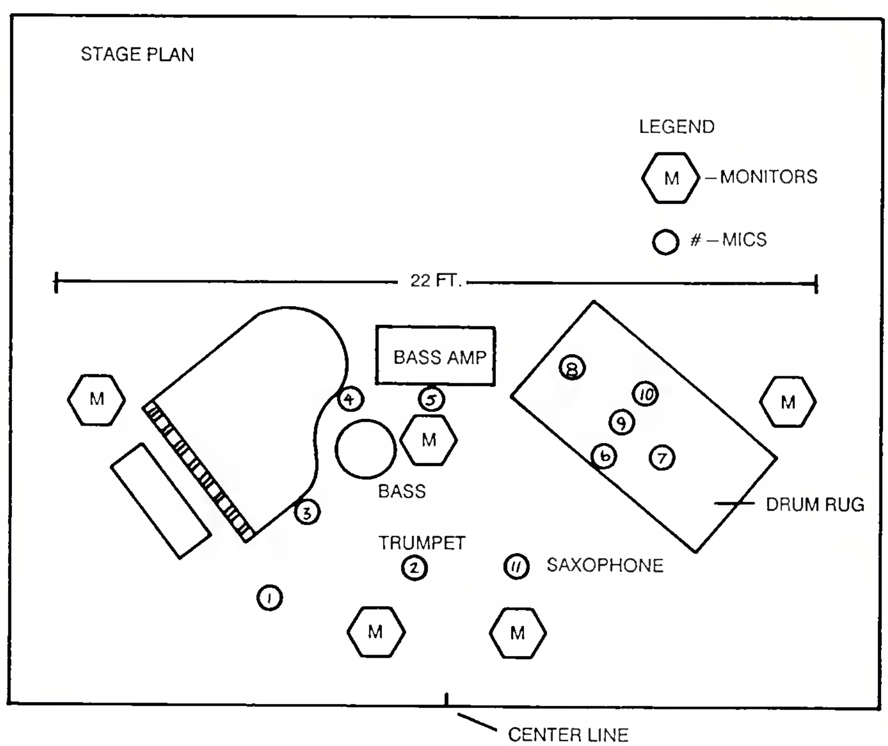
Robyn Gately is a touring soundman working out of the greater Philadelphia area, and also operates Joe's Sound and Salami. The illustrations in this article are recent examples of work the author has done.

During the initial conversation (usually on the phone) we proceed to tell people how horrible the sound business is.

Maintaining the consistency of quality personnel becomes your trademark. Therefore, it becomes mandatory that the selection of new personnel be treated like a Training Camp. You need to expose yourself to as many people who are interested in the sound business as possible, then be able to train the most highly qualified. Modular Sound utilizes two methods of ensuring quality personnel. The first is called the Apprentice program, while the second is an on-going series of classes for the personnel.

The Apprentice program is an open invitation to all interested people to talk to us about a career in sound reinforcement. During the initial conversation (usually on the phone) we proceed to tell people how horrible the sound business is. We tell them about the long hours, low pay, hard work; in fact, we tell people that becoming a soundman for a major act is a lot like becoming a

Figure 1. This stage plan was used for the Wynton Marsalis performances.



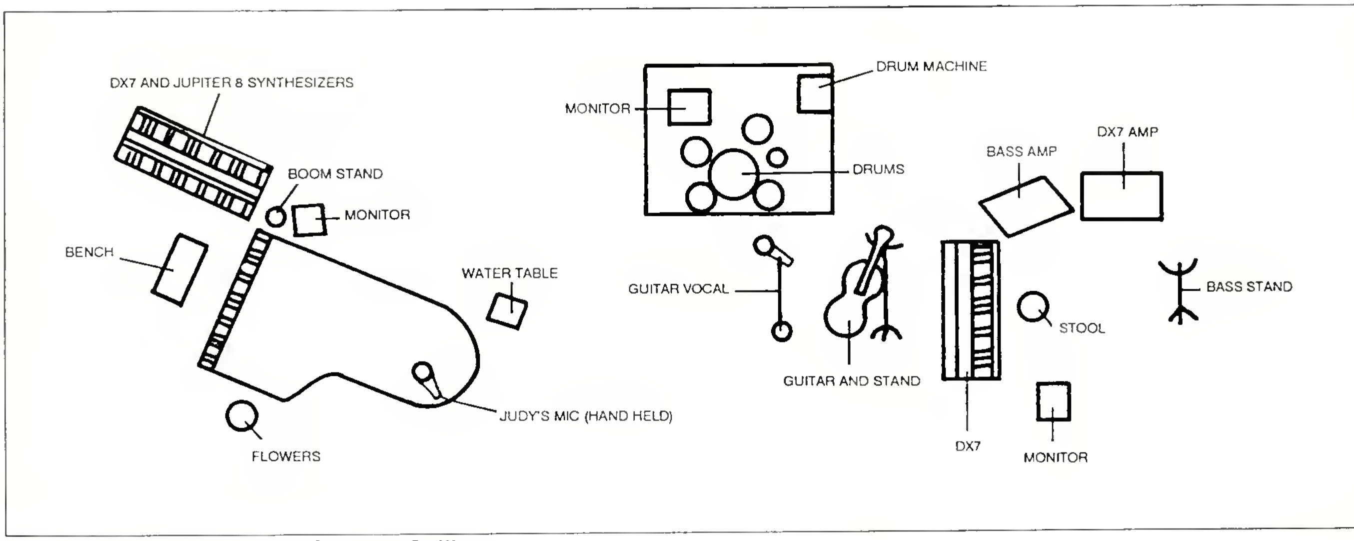


Figure 2. The concert setup for Judy Collins.

doctor or lawyer—it's going to be 8 to 10 years before you make any "real" money. It's going to be a long time before your mother brags about "my son—the soundman."

The point of all this is to eliminate all but the most eager, energetic and far-sighted individuals. Nine out of ten people never call again. But the one out of ten rookies is really worth training.

ATTENTION TO COMFORT

Training is a combination of exposing an apprentice to how a sound system operates at shows and through shop maintenance and classroom instruction. Class consists of taking a different section of a sound system each week and talking about it for an hour or two. (Modular Sound ensures attendance by offering free dinner.) The first class in the series is always "Artist Accommodation."

Artist Accommodation is really just making a musician feel comfortable on stage. However, it is important to realize that the "star" on stage is really like the quarterback on a football team. He may get all of the glory, but he's severely handicapped without a great

Figure 3. The house sound system for Judy Collins and Arlo Guthrie's performances at Carnegie Hall. Note that the left hand mid-high cabinet wears lenses to increase dispersion of the high and super-high horns, while the right-hand box has been tilted at a 30° angle to cover the balconies.

front line. Helping him is as much what you don't do or say as what you do.

I was impressed and will never forget his professionalism, but it never would have happened if he hadn't appreciated mine.

It is important to make sure that an artist feels that you know your equipment and are competent. It is even more important to go out of your way to make sure an artist feels comfortable on stage, so that they can relax and put on the best show possible. People like to be surrounded by bright, cheerful people, not grumps. Even if the speaker stacks crashed to the ground and burned like Rome, the musicians want to feel that everything's just fine and it's going to be a great show.

Many people, when confronted with their first opportunity to mix monitors on stage, turn up the vocals in the monitors and then sit and wait for the "star" to tell them what is needed. After watching this phenomenon, I discovered that the basic reason was that the monitor mixer was afraid to disturb the "star."

If you want to have fun and be successful in this business, realize that the sound technicians are there to *help* the "star" reach the audience. In almost every case, the performer appreciates being asked what else is needed, etc. Performers appreciate the fact that you are trying to help their career when you ask what you can do to improve their sound (and comfort on stage). It also allows you to meet some fantastic people.

THANKS, SAMMY

The success of a team is determined both by how well the team can recover from mistakes. This team spirit is crucial when people work together to present a professional show. Things do happen, so when a mistake is made, admit it. This can come back to save you sometime.

One of Modular's employees, Bill Magod, has said many times that I should write a book entitled, "And Then Sammy Davis Jr. Saved My Life." Over the course of an eight-show stand, I accidentally did something during one show that caused Sammy *to stop the show and start over*! I almost died. But, when it was time for heads to roll, I stated what I had done, and accepted full blame. Surprisingly, Mr. Davis appreciated my honesty, thanked me for it and let the matter drop.

Later in the week, there developed a problem with the promoter. Mr. Davis gave me the services of his lawyer and assured the protection of my equipment. Then, on the last day, he apologized to me for the actions of the promoter. Needless to say, I was

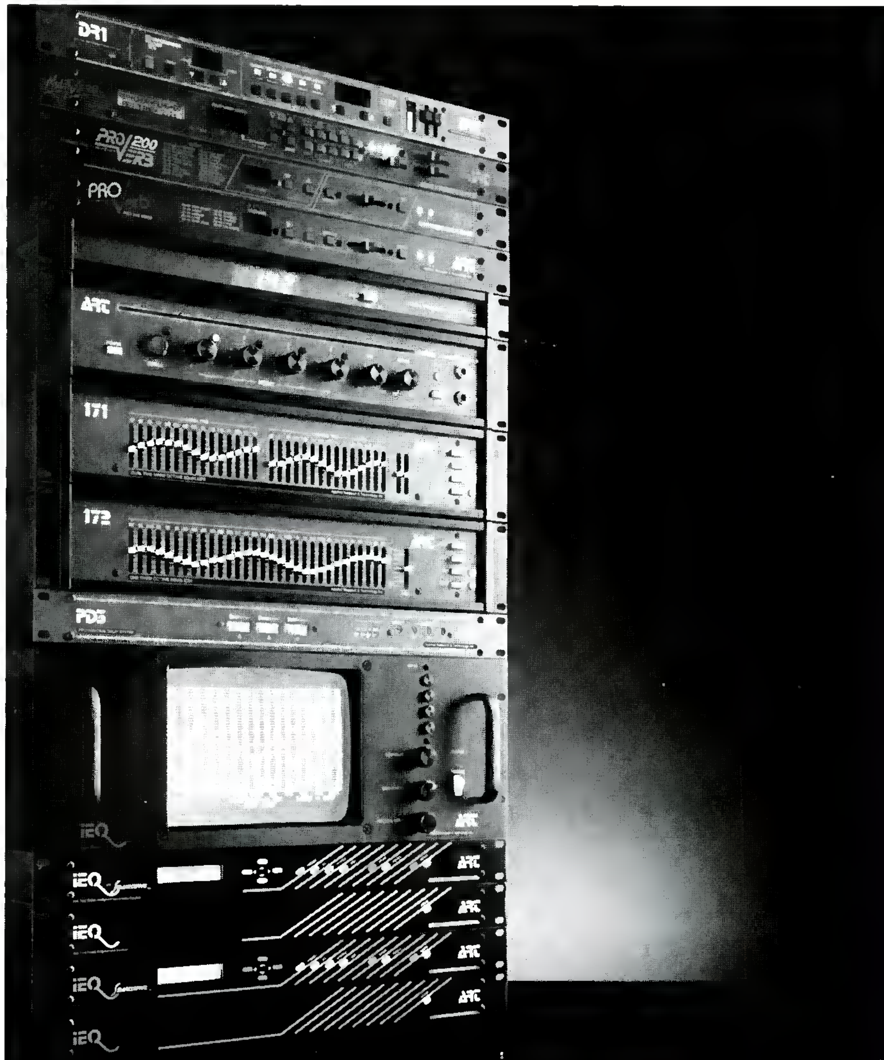

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Figure 4. Chuck Berry on stage with a Modular Monitor System at Rock-A-Rama, Philadelphia, PA, June 1987. A closer look reveals that the monitors are not the typical wedges, but are rectangular boxes with an adjustable kick stand, and are designed by Joe's Sound.

impressed and will never forget his professionalism, but it never would have happened if he hadn't appreciated mine.

ORGANIZATION AND SUCCESS

Similar to visiting football teams, many bands travel the country without carrying a sound system. Although they try to send out their requirements in advance (in something called a "rider"), the quality of what they get can vary greatly from show to show. Yet, in talking to these bands, one often gets the sense that they would rather have a company that has responsive, attentive employees with mediocre equipment, than top equipment with mediocre employees.

Organization, flexibility and preparedness are the three things that make the difference between a mediocre and good team or sound crew. Obviously, it is helpful to be fully prepared with all the right equipment and spares to make a show successful. Organization determines whether the show has a chance to be technically successful, while flexibility is the ultimate determining factor in success in live music. Modular has made a name for itself in festival production in the last few years through this flexibility/organization.

At Modular, there is an organizational meeting prior to each festival, where each crew member's job is spelled out. This can be compared to the preparations prior to a big game. At this meeting, it is also emphasized that any person must be ready to do whatever is necessary, at any time. The team spirit is the only thing that allows 30 unfamiliar people to come together for one weekend and present 20 or 30 acts without much sleep and without killing someone.

At one festival last year, our site coordinator for the show was asked to get some sodas for the stage. Why? Because he was the only person not doing anything at the time. The fact that he returned with an ice chest filled with soda was noticed by all on stage. Who can resist that spirit of cooperation? It's got to be good for your business. It makes people want to work with you, and makes the gig more fun.

STAGE SET-UP

Even the console inputs have to reflect the idea of flexibility within organization. Inputs need to have all of the known microphones permanently laid-out. You know that you're going to see drums, bass, several guitars and keyboards in at least one band. Set these inputs up permanently on the left-hand side of the board, leaving the right-hand side for the varying vocal inputs.

Modular generally sets up 5 mics across the front of the stage and a drum vocal. This way we can just remove from the stage the mics that each act does not use. We make sure that we have already covered the biggest band we'll deal with at the beginning of the show, this way the day is the most organized and the easiest.

The concept of being prepared, organized and flexible creates a work situation that is able to respond to the realities of live sound. Things change constantly—riders are incorrect, bands' membership changes, shows run late, and the responsible soundman accepts that as part of his job.

The only winning strategy is to approach each show with a positive attitude, and do everything possible to make the show a pleasant experience for everyone else. When cooperation is the basis of your game plan, then you will have a long, profitable and fun career in the sport of live sound. db

JOHN BARILLA

DESIGN CONSIDERATIONS: PART I

• In the May/June issue of **db**, we took time to define the electronic cottage concept, how it has affected our lives and businesses, and where it is likely to take us as the future unfolds. We did well to lay the groundwork, but now let's get down to nuts and bolts.

Designing an EC (electronic cottage) poses a unique set of problems. I, for one, would like to see many features of the "classic" recording studio incorporated—within the limitations of budget (small) and space (also small). Obviously, there are some trade-offs that have to be made. We simply cannot do all the things larger studios do quite so easily. Hence, we must seek creative solutions in everything we do.

So, let's consider four basic areas of concern when planning an audio facility: *room layout*, *electricity*, *interconnections*, and *monitoring*. I do not represent this to be a totally comprehensive list, but the most critical areas which every EC owner will have to deal with. Realizing, of course, that there are tremendous differences in implementation, I will try to be as general as possible, while drawing anecdotes from my own odyssey in putting together an electronic cottage.

ROOM LAYOUT

The area of most radical departure from classic recording studio protocol is that of room layout. While some high-end electronic cottages sport acoustically designed spaces, this is not the normative case. Most of us are simply grateful to have any space to set up our equipment, and so, we work within the parameters of what we are given. Usually, that means it's a box: a spare room, a basement, an office. Call it what you will, a box by any other name is still a box. Two pairs of parallel walls and a low ceiling is hardly an acoustical ideal, but that is what most of us have to work with.

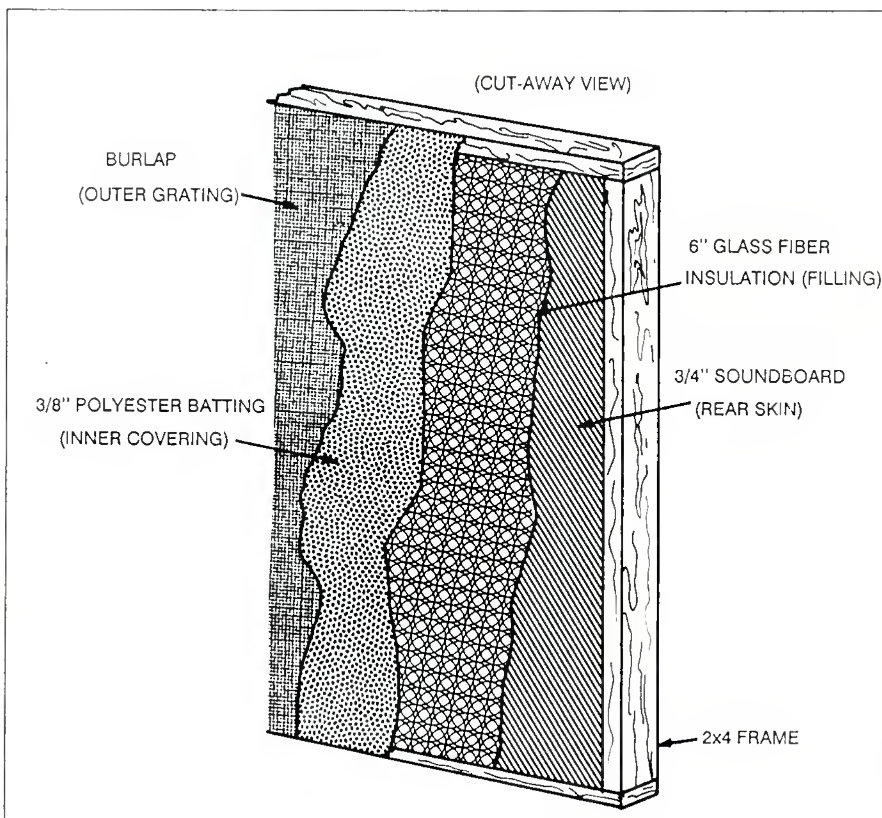
Before we get hung up trying to make, as it were, "a silk purse out of a sow's ear," let us be reminded that the electronic cottage is a populist—not an elitist—concept. Were we to take the high road, we would pick up the phone and call an acoustical consultant. (If I could afford *that* service, I would also be running a 24-track Studer rather than a 12-track AKAI!) Instead, most of us are looking for a prudent, common-sense approach that will *work* for us—not necessarily technical perfection.

In my case, I was allotted a small portion of a basement to set up my EC. My studio was to be essentially a one-man shop, with most activities centered in the control room. Not able to subdivide the basement in anything but the normal fashion, I constructed a 12x12 (you guessed it!) box, and called it home.

Now, during my construction process I learned a few valuable lessons that I would like to share with you. The principle of getting something useful from a small room is simply to diminish the acoustical contribution of that room until it no longer plays a significant part in the transmission of sound. The axiom most acoustical texts use is: *The smaller the room, the more sonically dead it should be.*

So the name of the game is to find cost-effective and aesthetically appropriate ways of diminishing sound reflection. While I am sure there are several good formulas for reducing bounce, I have stumbled on one that works rather well. (I admit, however, that it's not terribly effective on low-end boominess. Those long wavelengths—comparable to my 12-foot wall dimensions—are untouchable by any means except building bass traps. Unfortunately, I didn't have the space for that; neither did I have the time, money or inclination to wrestle with the finer parameters of acoustics. My aim was simply to cause reflected sounds to get fatigued and experience an early death, and by that definition what I achieved could properly be called a

Figure 1. Construction details of a non-stationary baffle.



"killer" method. As to the bass resonance problem, my solution is simply to monitor at low volumes for any critical application—like EQ'ing tracks or mixing. As long as there is not enough SPL present in the room to start the walls dancing, everything seems to work out just fine.

OFF THE WALL

You can use this simple method if you want to alleviate bounce in your EC. I chose to erect permanent walls, but if you don't, making a series of non-stationary baffles to cover critical areas will work quite neatly. They can then be supported from the floor, or flown from the ceiling (*Figure 1*). Make yourself a frame of 2x4s (use 1x4s if they are going to be hung), securing all parts together well. Unless it's a permanent wall, you really ought to glue and screw everything together to prevent any rattling. The outside skin should be something like 3/4-inch "soundboard." (Anything in the family of fibrous composite sheathing will do. Ask at your lumberyard.) The area in between the frame should be filled with 6 inches of "un-backed" glass fiber insulation (i.e., no foil, no paper binding). Just compress the 6 inches to 4 inches, securing it to the soundboard backing with a few staples.

Glass fiber insulation is known to be nasty stuff. You don't want to rub it into your skin or inhale the particles, but as most common building materials go, it has real value as a sound absorber. If we cover it over with another skin (as some people do), we have defeated that aspect of its use. If, however, you go to a fabric store and purchase some polyester batting (batting is the stuff used to fill quilts) and tack it onto the inside surface of the frame, you will be able to effectively encase the irritating particles of glass fiber, while still retaining all of its sound reducing qualities. A single layer of this batting (it usually comes in rolls three or four feet wide and 3/8-inch thick) strung over the frame will form a springy barrier to the glass fiber while allowing sound to enter the "death chamber." Cosmetically, a final covering of burlap (also available in a fabric store) will put a professional-looking finish on your sound absorbers. Burlap is sufficiently porous to let sound through, sufficiently strong to greatly reinforce the barrier, and has the acoustical qualities of a grating, somewhat diffusing sound and dispersing it from its incoming direction. The effect is to even out the sound a bit more than without it. Don't fear that your studio needs to look like a potato sack, though. Burlap comes in many different and fashionable colors.

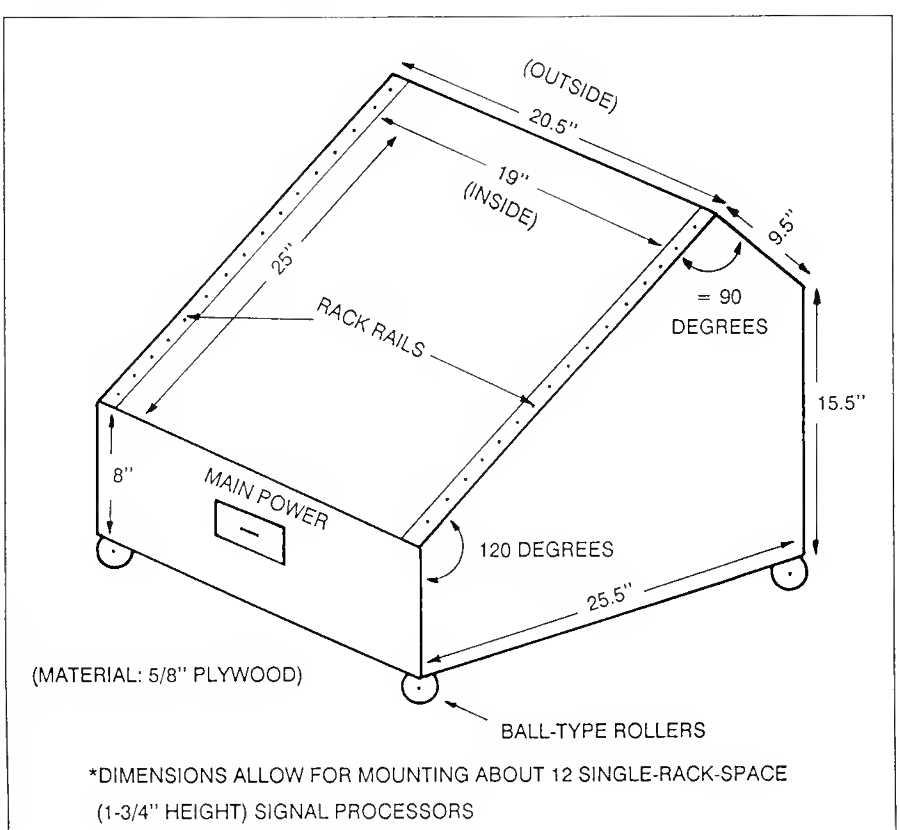
Cover as much of your studio as you can in this manner, but if you can't do it everywhere, at least distribute the sound absorbers symmetrically. Mixing would certainly be weird in a control room where the left side was covered with insulation, batting and burlap, and the right side featured concrete, metal and sheetrock. So even if your room has permanent fixtures on one wall and not the other, try to provide equal absorbency on both sides, even if it means having less coverage.

In a small electronic cottage, the areas of most critical concern with regard to sound absorbency are those directly behind the speakers and the wall opposing it. While in the acoustically-designed larger studio, a certain amount of reflected sound is found to be desirable, such pursuits usually have no purpose in the expedient world of the electronic cottage. We do what we must in order to get a good sound. That means restricting our hearing—to whatever extent possible—to what is coming from the speakers, with the least admixture of room ambience. It doesn't sound too awesome (or even natural) in that context, but you can learn to work like this to great advantage—in terms of final product. It is one of those necessary compromises that is well worth making. To sum it up then, a combination of 1) near-field monitoring, 2) low listening levels, and 3) efficient sound absorbency from both front and rear can help you create a reasonably accurate mixing environment—even in a box!

ERGONOMIC FACTORS

Most other aspects of room layout are simply discretionary. But one general issue should always be kept in mind: Take time to make it comfortable. It may not seem like such a big deal when you are high on your new gear, but after months of operation, working in a confined area can get *very* tiring. When you are ready to make permanent installations of your equipment, double check that your MIDI keyboards, recording console etc., are all at workable heights and angles. Ergonomics—human engineering—is a subtle but ultimately critical consideration in studio design, particularly on the level of the electronic cottage. (One studio owner suffered from a perennial backache

Figure 2. A roll-around equipment rack that is easy to build.



due to the contortions he had to put his body through to operate his system.) After all, an awful lot of equipment comes under the control of a single operator, so great care must be taken to keep all items neatly arranged, but close at hand. An appropriately sloped roll-around signal processing rack (*Figure 2*), can be wheeled to your center of activity and save you a lot of excessive movement. (Buy some rack rails and build it to your own specs.) Also, one of those neat pneumatic chairs that swivel, and go up and down at the touch of a lever, is an invaluable asset in alleviating studio fatigue. (You can check them out at an office or art supply store.)

PSYCHOVISUAL FACTORS

Finally, a word about color. Bright, saturated colors (much like intense upper mid-range EQ) tend to cause an initial titillation that is inevitably followed by sensory fatigue, and finally, annoyance. Similarly, a predominance of dark colors tend to bring about low energy, depressive states of mind, impede light dispersion and hence make visibility tedious. Life in the electronic cottage can indeed be stressful when the environment is not sensitively designed. For the best overall response, always choose light, unsaturated, neutral colors for your major wall coverings. If you do so, your ability to spend long hours in a confined space will be improved.

ELECTRICAL CONCERNS

For those who would build a safe electronic cottage, capable of delivering clean, hum-free audio, there are two main areas of concern with regard to electricity. Those areas are: 1) power consumption and distribution and 2) system grounding. I use the word concern rather intentionally because electrical concerns so frequently get neglected, only to crop up as mysterious problems later on. In their zeal to get started and be productive, EC owners sometimes look at an electrical outlet as an infinite power source. As modules are added to their system, extension cords proliferate in haphazard ways, while noise levels seem to rise, data glitches become more common, and circuit breakers pop off in the night.

Fortunately, most of these studio nightmares can be avoided with a little forethought. Understandably (and almost by definition), the electronic cottage is usually set up in available space and the hope is that available electrical circuits will be sufficient. Sometimes this is the case, but sometimes it is not. A little snooping will answer the question.

THE POWER STRUGGLE

Your first step is to find out what the power limits are of the available circuit(s), and compare that to the power requirements of your equipment (and lighting). It is always a good idea to keep the power for audio and for studio lighting on separate lines to avoid any possible interference between them, and also to avoid an excessive current drain on any one line. So let's examine the power needs of a typical audio system (mine), and see how it might be resolved in the context of the electronic cottage.

Now, some of you may be setting up in a spare room of your house or apartment. Fortunately for you, electricians

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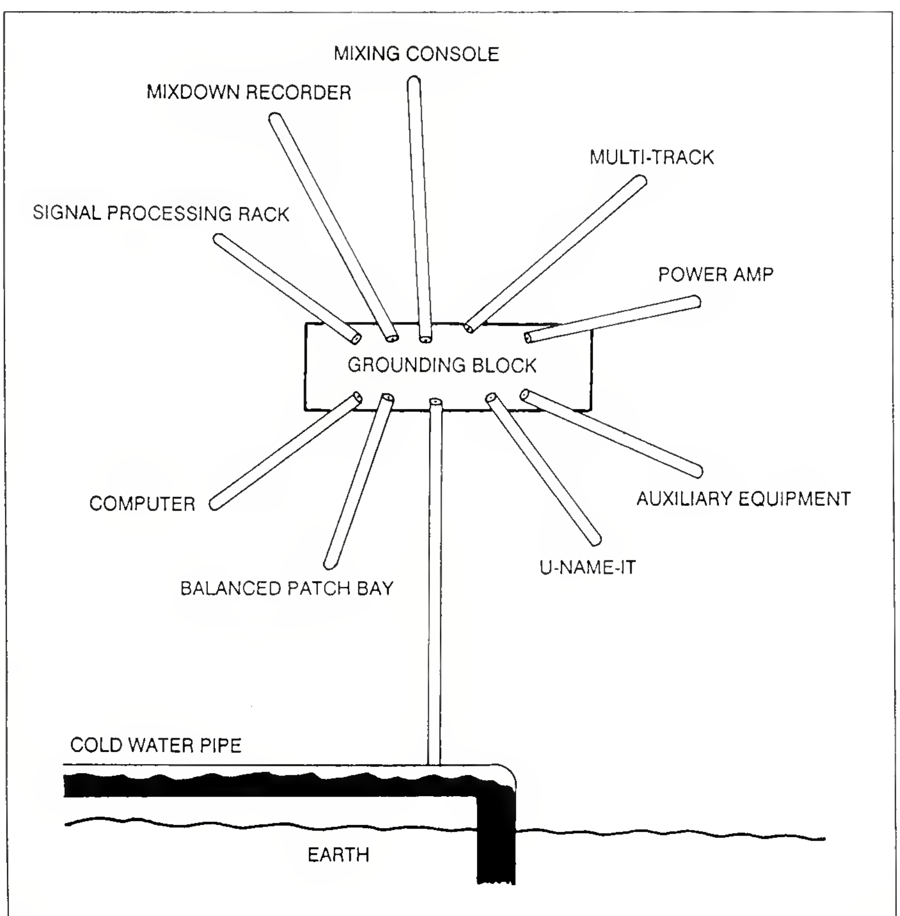
conventionally keep the room outlets separate from the overhead lighting circuitry. That seems hopeful enough, but the question remains, "What other loads are being powered by the circuit, which are not in the room?" In most residential installations, electrical circuit documentation is often poor or non-existent. In any case, don't necessarily trust what you see written on the main service box. It's best to test it yourself. Simply turn every light in the house on, and put some nightlights (or small electrical gizmos of any sort) in every spare outlet and turn them on too. Then go down to the main service box and turn all of the individual circuits off. When everything is dark and still, turn one circuit on at a time, note what lights up and write it down, along with the wattage of the bulb or other device that is normally operated from that outlet. From there, it is a simple matter to compare the *available* power with the requirements of your system (both present and projected) and determine if it is sufficient.

Now ideally, it is best to have a dedicated circuit installed for your audio power source. That's what the pros do. If you need to work with an existing circuit, let me give you an example of the computations. In my own case, I had determined that a circuit that was running through the cellar (near my studio door) was not getting much use. (Remember, I had built a room where there was none before.) After making a survey of the line, I found that it also powered one 60 watt and two 100 watt light bulbs upstairs. A few outlets were also on the line, but were hardly ever used—unless I decided to use them. So the total power requirement was 260 watts. Examining the circuit breaker, I found the current rating of the line was 20 amps. (Single-phase residential circuits are usually 15 or 20 amps.) At an assumed voltage of 115 volts, the power which the circuit could safely deliver turned out to be $P = VI$ or around 2300 watts. (I am told that today it is best to use 115 volts rather than the nominal 120 when figuring power, because due to fluctuations, the consumer always seems to get a little less.) So, subtracting the 260 already accounted for, my available power turned out to be around 2040 watts.

Figure 3. The "Star" system of grounding.



How did the power needs of my equipment stack up? Well, my mainstay, the AKAI 12-12 (a 12-track mixer/recorder in a single unit) logged in at 150 watts, my stereo mixdown recorder, a Tascam 32, came in at 70 watts, a Yamaha submixer was 15 watts, and an assortment of gear—synthesizers, sequencer, all manner of signal processors (the Yamaha SPX90 being typical at 20 watts)—17 pieces in all ranging from 5 to 30 watts totalled out to 340 watts. The surprise was the Yamaha P2150 stereo power amp. In order to pump 100 watts of power per side into my monitor system, it consumed 500 watts of power. Calculating for a headphone amp, dbx unit and a few other pieces of miscellaneous gear (such as my guitar amp)—if everybody was cranked up at the same time, my total power consumption would be less than 1500 watts. This was considerably under my 2040 watt limit, offering a bit of headroom for gear added later on. I was fortunate in being able to find a relatively idle circuit. Needless to say, if a television set (about 400 watts) and a toaster (1100 watts) were on the same line, it would have been stupid and dangerous to add anything else, and certainly time to call the electrician. But unless you checked, how would you know?

Another area that should be given some sober forethought is power distribution. Avoid skimping here. Almost every piece of gear in your electronic cottage is controlled by a microprocessor. The FCC calls them Class B computing devices. In other words, they act like computers, and so they ought to be treated as such. We know that computers don't like a) dirty electricity, b) spikes and surges, so look for fused power strips that pack a line filter, spike and surge suppressors. They are a little expensive, but the price has come down on these devices considerably in recent months (probably because so many manufacturers have jumped into the game). You can safely daisy-chain another less expensive power strip with one featuring all the protective circuitry—provided the total current draw of the devices you are powering does not exceed the rating of *either* power strip. Remember also that you will always find a need for another outlet, so install power strips liberally around the periphery of your EC.

"Grounding is an area of 'black magic' for many sound technicians and engineers..." So states an operating manual for Yamaha power amps, and how true it is indeed. Studio grounding procedures often end up being a confusing mixture of science and voodoo. Believe it or not, there is a rational approach to establishing a unified system ground.

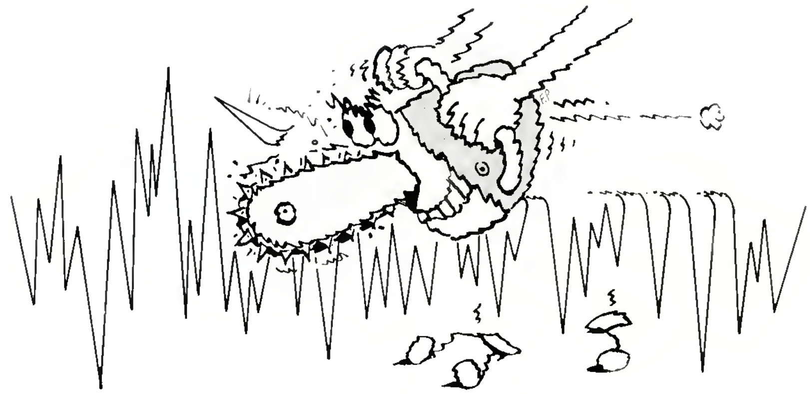
It's easy to get confused about these things, and "down to earth" advice on the matter is often difficult to come by. Most standard textbooks on studio practice give it only scant treatment, and more scholarly journals often sound like graduate electrical engineers addressing their peers. What seems to complicate things is the variety of grounding considerations for balanced vs. unbalanced audio, and those devices with a grounded chassis vs. those without. With the exception of items such as cassette decks, manufacturers have not yet joined in consensus about the best way to achieve their ends—ostensibly clean, quiet and safe electronic gear, so the most expedient method usually reigns. The more equipment you install, the greater your chance of error, unless you have a firm rationale planted in your mind.

When I first started planning my studio, I was determined to get the highest possible signal-to-noise ratio out of my system, and I knew that grounding was an important part of obtaining that goal. I had previously worked in some pro studios where the sound quality was uniformly excellent, and yet, in other (so-called) pro studios, seemingly unexplainable hums, noise and RF interference wafted in and out at will, terrorizing the engineer like a family of malicious poltergeists. What made the difference? Before firing up my soldering iron, I spent some time trying to find out. After talking with some of New York's best maintenance engineers, digesting articles from the available literature, and trying it out first hand in my studio, I have come to a comfortable awareness of effective grounding procedures, and I would like to pass some of it on to you.

SOLID GROUND

First of all, let's ask the question, "What is ground, and what purpose does it serve?" We know, for instance, that ground provides a reference from which we measure voltage—the unit of electromotive force. But all equipment grounds are not created equal. Unless we reference *their* grounds to the same point, they will all have their own "idea" of what absolute ground should be, thereby creating voltage differences in the audio signal when various pieces of gear are patched together. Because of the complex interaction of audio grounds and chassis (a.c.) grounds, oscillations will occur subtly, turning your system into a 60 cycle tone generator—replete with harmonics. For this reason alone, it is important to have a common ground between all elements in your system, but beyond this, it is also important to have the *best* possible ground reference available.

What is the best possible ground? Earth, of course, for a couple of really important reasons. Since the ideal ground path is one that has the least possible resistance, we should be aiming for a big, fat conductor between



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the "local" equipment grounds and the "absolute" earth ground. One of the benefits is that the earth is an abundant potential source of electrons paving a smooth highway for the disposal of spurious RF or magnetically induced signals. Considering that all studios have unwittingly built in antennas (in the hundreds of wires and metal cases it contains), you can see why it becomes very important to have an efficient means of diverting some of that RF garbage away from the audio signal.

More fundamentally, earth, beyond providing a drain for vagabond electrical currents, is also a storehouse for electrons. Since your body is mostly electrolytes, and walks on the face of the earth, it is well to keep this in mind. In the event of a short circuit in the power to any of your devices, that circuit will seek to maintain itself through the best available path to the ground. If your equipment ground is faulty, and *you* happen to be twiddling the knobs, the best source of electrons around will be those in your body! Hence, one cannot overstate the importance of a solid grounding installation in the electronic cottage.

DOWN TO EARTH

Now let's bring this discussion down to earth. When the utility company installs power in your dwelling, a single grounding point is established by running a very large wire from the breaker panel to a cold water pipe. It is firmly attached to the water pipe with a heavy-duty clamp. When the main power is subdivided into various lines, the ground is carried along with the hot and neutral wires to the individual power boxes throughout the facility. What makes this standard grounding procedure less than adequate for clean audio is *not* the grounding point itself (a cold water pipe is really quite excellent since it is a large underground conductor). Instead, it is the simple fact that the ground wire (being housed within the same casing) is running parallel with the hot and neutral wires, from the breaker panel to wherever you have installed your facility. In fact, bundles of power lines are conventionally paralleled together, furthering possible negative interactions such as increased capacitance and power-related signal inductions from devices in remote parts of the dwelling (your refrigerator, for example). Hence, what we want to do is run a dedicated ground connection from our equipment back to the cold water pipe (away from any a.c. power lines), providing our facility a safe and clean system ground.

INTERSTELLAR METHODOLOGY

The method which has proven most effective is known as *star-grounding*. If the various units of gear (console, recorders, equipment racks, etc.) were all considered points of a star, all of these peripheral points should be made to converge on a central grounding block, which in turn is connected to the cold water ground. This method is vastly superior to haphazardly daisy chaining grounds from one piece to the next in that it eliminates interaction between the pieces, essentially connecting each chassis directly to ground (Figure 3).

To star-ground your system, you will need a few items. For the central grounding block, get yourself a piece of solid copper. It usually comes around 1/8-inch thick, and it's not cheap. So buy whatever amount you actually need to install terminal posts for all your equipment (and maybe a couple extra). Solid copper plate is not your usual hardware store item, so you may have to search around for an industrial metal supply.

For the terminal posts, drill some holes in the copper plate, and permanently affix some small (#6) machine bolts, brazing them in place with solder and a propane torch. This grounding block should then be mounted at some central location, insulated from any other ground reference. (A wooden frame that supports my recording console seemed like a good place in my installation.) Next, find the nearest copper cold-water pipe. (Make sure that it is what it seems to be, by doing a continuity check between the pipe and the ground on the studio electrical outlet.) Then run an insulated solid copper wire between the pipe and the grounding block, permanently affixing it to the pipe with a hose clamp. At the other end, solder on a spade lug and tighten it onto the terminal post with a nut and washer. (Use heavier grounding wire than normal here. For 15 to 20 amp installations, use at least #10 AWG or better.)

Now connect the individual points of the star to the central block. Every piece of equipment whose chassis resides at ground potential should be connected to the grounding block with insulated heavy gauge copper wire (#14 AWG or better). For each piece of gear, prove to yourself that the chassis is actually at ground by doing a continuity check between the ground pin of the power cord and the chassis screw to which the grounding cable is to be attached. (While most equipment with a three pin power cable will meet this criterion, occasionally you will find one with no ground reference through the chassis. In such a case you will have to run the star configuration for that unit, from the power cord ground pin instead.) Chassis grounds from items mounted in equipment racks can be bused together in a smaller star pattern, with a single wire running from the rack to the grounding block. (If you are using a balanced patch-bay, a groundwire bus should also be connected to the grounding block, but not so with the unbalanced variety.) The last contingency are those devices that utilize a two-prong non-polarized a.c. cord. Whereas grounding through the power cord has been forsaken here, audio ground provides the reference and is usually tied to a chassis screw—even though the outer casing of the unit is insulated from ground. You can verify this yourself by testing continuity between these two points. Therefore, virtually every piece of gear in your EC can be referenced to the same unified ground.

Having *firmly* attached groundwires from chassis to central grounding plate to earth ground as previously described, the standard third-pin ground can now be safely and legally lifted at the power outlets. Our "replacement" ground fulfills all the safety requirements while providing us with a cleaner ground reference for audio. It is commonly reported that changing over to an efficient star-ground system often results in audible (and measurable) differences in the S/N ratio of the studio. This, of course, becomes more and more critical as you increase your arsenal of outboard gear.

Prove the system out by listening to it through the monitors with faders up all the way. Some minor accommodations may still further improve the quietness of the system. In the case of non-polarized equipment, reversing the plugs may help. Individual pieces in a rack may benefit from being lifted again at the power strip in the rack, even though the strip itself has previously been lifted at the outlet that feeds it. Quite fortunately, you can do all this with total impunity, since the grounding requirements of each piece have been adequately provided for with the star-ground system. db

Broadcast Audio

RANDY HOFFNER

• Television is well into its fourth decade in the United States, and its impact on our society may be fairly described using such words as "fundamental" and "profound." The changes that the past several years have brought to the television and ancillary video industries have been no less fundamental or profound.

From the inception of television broadcasting until the late 1970s, the producers, distributors, and consumers of television focused their attentions on the visual image, with only secondary consideration being given to the audio portion of this audio/video medium.

Even if a high-fidelity soundtrack did accompany a network television program, for many years the networking system often compromised the audio quality delivered to the local broadcaster. Even if the high quality audio was delivered to the local broadcaster and transmitted, the audio section of the home television receiver was not up to the task of faithfully reproducing it.

COMPETITION IMPROVES AUDIO

The 1980s have been years of change in the television and video industries. The commercial and public networks and a relative handful of independent broadcasters have been joined by a legion of new independent broadcasters and cable networks in competition for advertising dollars and the attention of viewers. The television consumer also has pay cable, videocassettes and videodiscs, and direct satellite reception to further expand the choices available.

All this competition, plus the conditioning of the American public over the last 10 to 15 years to expect high fidelity and stereo sound from movies, phonograph records, compact discs, car stereos, personal headphone equipment, and many other sources, have produced a keen interest in better television audio. The precursor to broadcast television stereo was the FM stereo simulcast of the audio portion of certain television programs.

Although Federal Communications Commission inquiries concerning stereo television had been made several times in the past, the industry responded with indifference each time until the beginning of this decade. This time, the electronic sophistication of the American public, the burgeoning competition in the video media, and the capability to deliver high quality multi-channel audio via satellite acted in concert to precipitate the FCC's approval of multi-channel television sound broadcasting, and the adoption of the BTSC system as the de facto United States standard for MTS transmission in 1984. These actions constituted watershed developments in television audio.

Regular network stereo television broadcasting began in the United States in July, 1985. In the three years since then, public awareness and acceptance of television stereo has been little short of phenomenal, with at least eleven percent of television households now having stereo television reception capability. The speed with which consumers have embraced stereo television can be more fully appreciated when one considers how long it took color television to attain this level of market penetration.

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On the transmission side, between 400 and 500 of the approximately 1300 television stations in the United States are stereo-equipped. The NBC Television Network, the leader in stereo television broadcasting, currently delivers stereo audio to almost 89 percent of the U.S. television audience via its 141 stereo affiliates. NBC's stereo schedule includes early morning, late night, and virtually all prime time programming. CBS has announced large-scale stereo broadcasting beginning with the new fall season, and ABC cannot be far behind. The Public Broadcasting Network airs about 30 hours of stereo programming each month. In the cable TV world there are now over 3500 BTSC stereo encoders installed in cable systems in the United States, permitting system operators so equipped to offer, in addition to off-air stereo TV sound, BTSC stereo sound on cable channels and premium services.

As important as stereo and high fidelity audio for television have recently become, the future will see them become even more important. The growing trend toward large-screen and projection television sets begs for a larger sound field to accompany the larger picture. Surround sound has become a staple in the movie theater, and the trend toward larger video screens will increase the demand for surround sound on television. There are currently about a million home surround decoders in the field, and many premium television receivers feature integral surround decoders. Surround sound not only delivers the excitement and sonic realism of the rear or surround channel. It also affords a convenient method of providing a "hard center" channel for dialogue and other monophonic elements of a stereo soundtrack. The center speaker is far superior to the "phantom" center channel created by equal-level mono signals in the left and right speakers, particularly for listeners sitting off-center. It conveys a much more accurate sonic "picture" of the front sound field. This stereo image accuracy becomes more important as the sound field is widened. Early experiments with stereophony generally employed more than two audio channels, and it may be argued that the two-channel stereo convention was born of convenience and cost-effectiveness rather than audio realism. It is also true that accuracy in sonic spatial location is much more important when sound is related to picture than when only audio is involved.

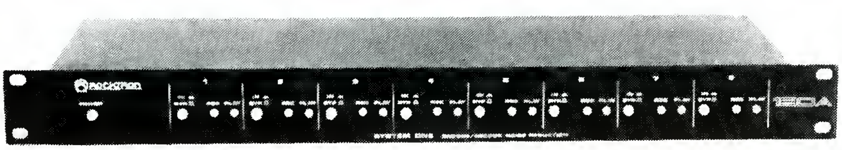
As everyone involved in the television industry is aware, there is a major thrust to develop high definition production and transmission systems. There are a number of ideas presently afloat to realize the goal of enhanced definition, and all of them include a wider aspect ratio in addition to greater resolution. It is a certainty that high definition, wide screen television will demand high definition, wide screen sound. As the picture is improved, the viewer will expect and insist on improvements to the sound that accompanies it.

MOVING TO DIGITAL

At this point, stereo and high fidelity audio for television are given, and major efforts will be expended to make substantive improvements beyond television sound's present capabilities. Television audio, like audio in general, is rapidly and inexorably moving into the digital domain. The attributes of digital audio: wide dynamic range, very low distortion, absence of interchannel stereo phase errors, and robustness under adverse conditions, make it very appealing for applications in television. These applications include processing, recording, routing, distribution, and eventually extend into the living room of the viewer. The transmission of digital audio to the home will assure that the highest quality sound is provided to the video consumer. The day will come when audio will be digitized prior to the recording process, and will remain in the digital domain through all the processing, post-production, in-plant and external distribution systems, and only enter the analog domain in the home receiver. This will solve a number of problems associated with analog audio channels, but will present its own challenges, not the least of which is the relatively large amount of bandwidth required by high quality digital audio signals.

These are the challenges presented to television audio's future. We have entered an era in which the sound that accompanies the television picture is as important as the picture itself. Surely this importance will continue to grow. It will be interesting and exciting to watch the future of television audio unfold! db

Goodbye Noise!



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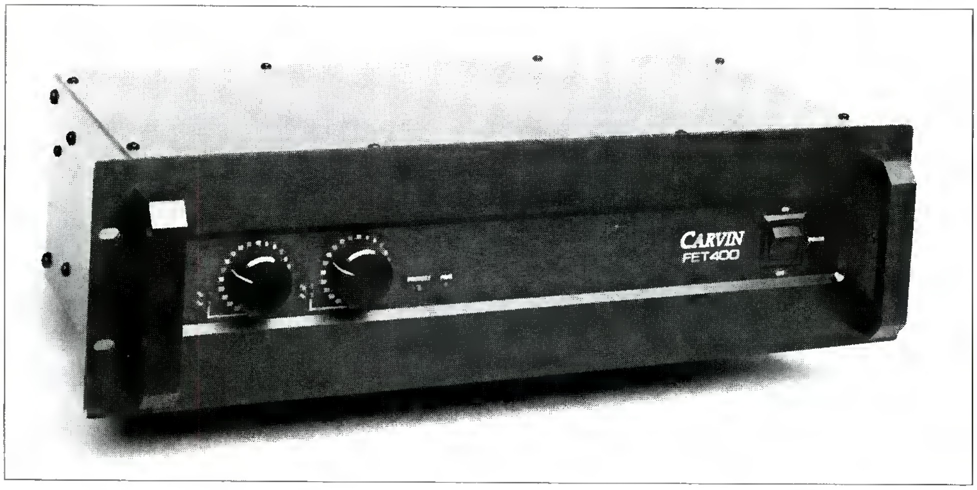
Lab Report

Carvin FET 400 Power Amplifier



GENERAL INFORMATION

Carvin offers two "FET" series amplifiers (FET-400 and FET-900). Both have the same front panel and rear panel features. The amps differ only in terms of power ratings and a few other performance specifications. As Carvin points out in their well written owner's manual, MOSFET devices, used in the FET 400 amplifier tested for this report, offer several advantages in performance and reliability compared with bipolar transistors. For one thing, MOSFETs have a negative temperature coefficient; their internal impedance increases with increases in operating temperature, thereby reducing current flow and preventing possible thermal runaway.

To protect against potentially damaging d.c. voltages reaching the speaker loads, Carvin also incorporates what they call a "Speaker Guard" protection circuit – electronic disconnection of the load when a fault is detected. The action is faster than would be possible using mechanical relays. The FET 400 also features short circuit protection and turn-on muting. The amplifier has a thermally controlled fan controlled by a circuit that provides variable fan speed depending upon the temperature of the amplifier.

The FET 400 amplifier incorporates a heavy duty linear power supply with a high current transformer. In examining the external and internal construction of this professional amplifier we found only premium grade components in all sections of the amplifier. All circuit boards were constructed of epoxy glass fiber with moisture-proof epoxy overplating. The chassis itself is constructed with 14-gauge steel side panels and with a standard 19-inch by 5-1/4 inch front rack panel of 0.187-inch thick aluminum. It is anodized in brushed black. As is evident from the claimed and measured specifi-

Figure 1. Frequency response.

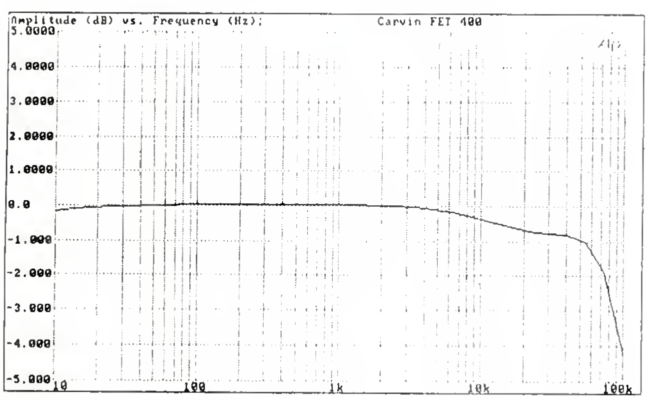
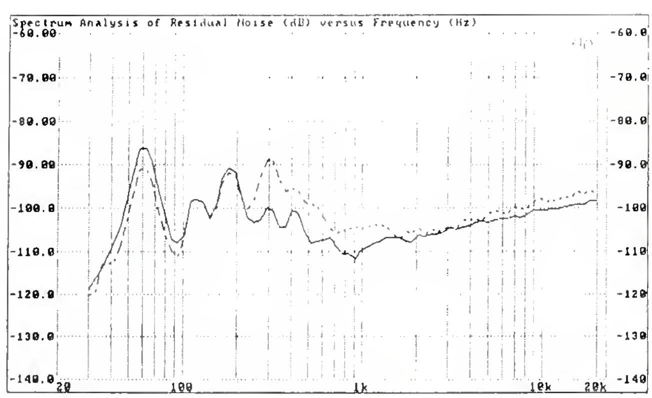


Figure 2. Spectrum analysis of residual noise illustrates that most of the significant residual noise is power-supply related.



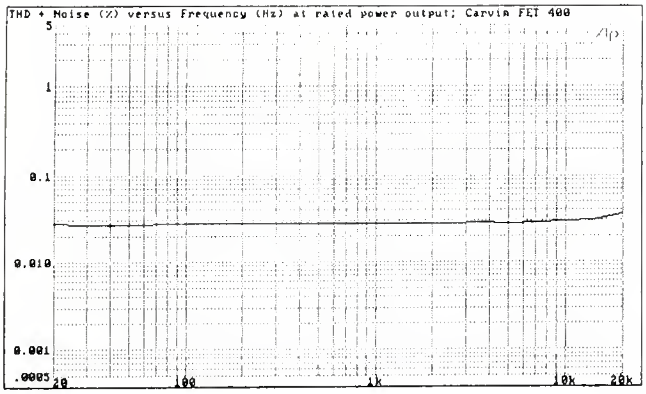


Figure 3A.Harmonic distortion plus noise vs. frequency, at rated output (100 watts/channel, 8Ωloads).

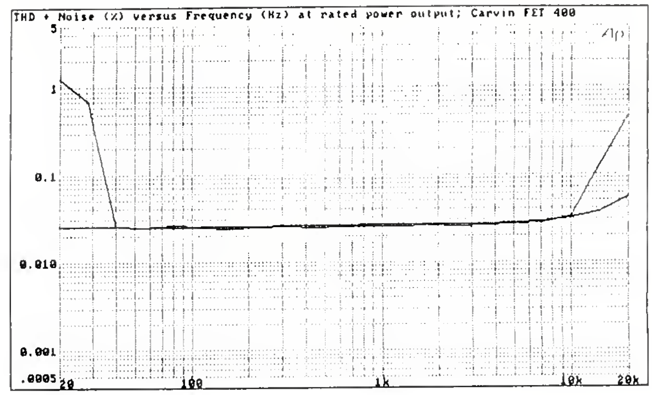


Figure 3B. Harmonic distortion plus noise vs. frequency at rated output (200 watts/channel, 4Ωloads).Improved readings are for 180 watts/channel (lower traces).

cations summarized in our VITAL STATISTICS chart at the end of this report, this Carvin amplifier, though intended for professional use and sound reinforcement applications, could serve equally well as the amplifier of choice where critical listening requirements are involved, such as recording studio monitoring applications.

A professional "accessory group" of features is accessible via a removable sub-panel on the rear of the amplifier. Removal of this plate or panel permits you to set various switches for such optional functions as high-pass and low-pass filtering, a limited function with four adjustable levels, and mono bridging of the amplifier.

CONTROL LAYOUT

A rugged rocker switch at the right end of the front panel serves to power up the unit, while at the left there are two input level attenuator knobs that have 41 1/2-dB steps for precise matching of the amplifier's sensitivity to that of your mixer or preamplifier outputs. A "Protect" light nearby illuminates only in the event that the "Speaker Guard" circuit is engaged, or, in some instances, briefly when the unit is first turned on.

At the left of the input level attenuator for each channel are a pair of LEDs that indicate the signal status of each channel. A green "Signal" LED indicates when a –30 dB signal, or greater, is present at the output terminals. The red "Clip" indicator LED lights when clipping distortion begins to occur. The clipping indicators feature a peak-hold circuit that provides a bright indication even when momentary clipping occurs.

The rear panel of the Carvin FET 400 is equipped with a heavy three-conductor line cord, adjacent to which is the main a.c. fuseholder containing a 5-ampere, type 3AGC fuse. Speaker outputs for each channel are available either from a pair of 1/4-inch phone plugs or from dual gold-plated 25-ampere color coded binding post banana jacks. Speaker fuses are also accessible from the rear panel. Both XLR connectors and 1/4-inch phone jacks are provided for input connection and each may be used either for balanced or unbalanced inputs. Removal of the plastic cover found on the rear panel reveals two banks of gold-contact mini switches that are used to select the various accessory functions of the amplifier mentioned earlier. A 120/220 VAC line switch is located near the a.c. line fuse. This switch normally comes covered with a protective label to prevent accidental shifting of the switch to an improper line voltage.

LABORATORY MEASUREMENTS

Frequency response of the Carvin FET 400 amplifier is plotted in *Figure 1.* If you interpret a " ± 0.5 dB" spec to mean that the response can exhibit a 1 dB spread over its stated range, then the response of this amplifier might be quoted as extending from below 10 Hz to 50 kHz (at which point response is down 1.0 dB.) If, on the other hand, you insist that "± 0.5 dB" means that response, referenced to 1 kHz, must not depart from "flat" by more than 0.5 dB in either direction, than we'd have to say that the response of this amplifier extends from below 10 Hz to around 16 kHz. The –3 dB point in the response curve (referenced to 1 kHz) occurred at 85 kHz.

Figure 4A. Harmonic distortion plus noise vs. power output per channel (8Ω loads, 1 kHz).

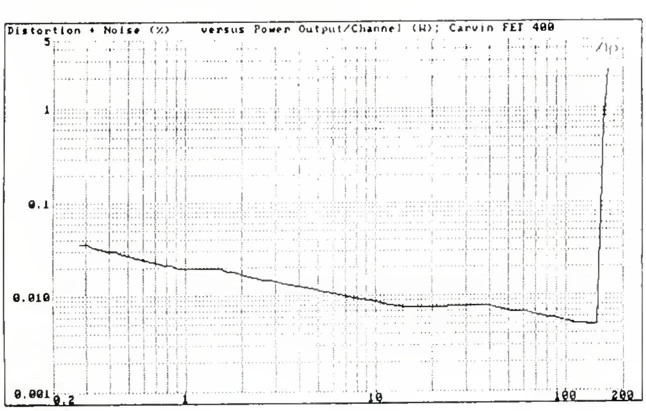
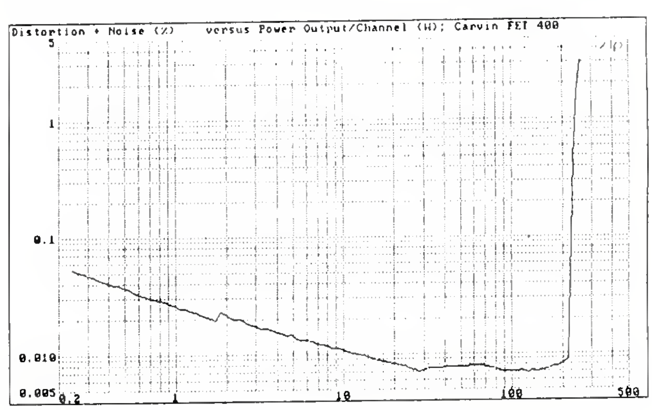


Figure 4B. Harmonic distortion plus noise vs. power output per channel (4Ω loads, 1 kHz).



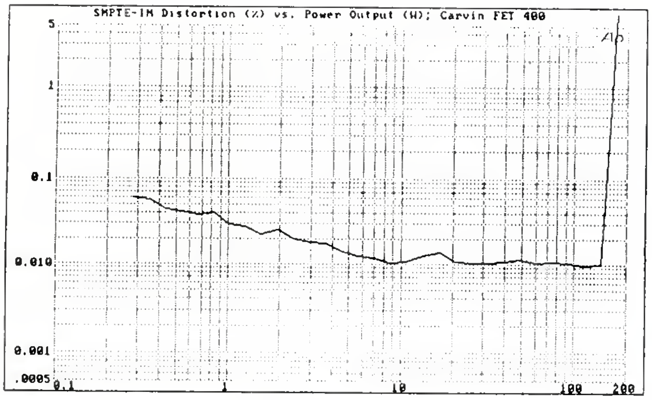


Figure 5A. SMPTE-IM distortion vs. power output. (8Ω loads).

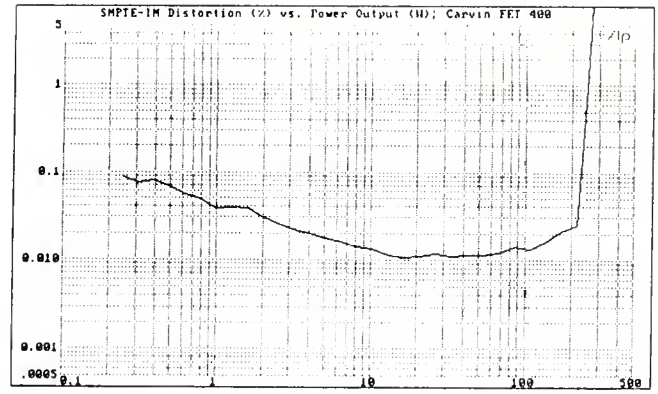


Figure 5B. SMPTE-IM distortion vs. power output. (4Ω loads).

Input sensitivity for rated output measured 1.1 volts rms. Signal-to-noise, measured using an A-weighting network, measured 105 dB below rated output for the left channel and 108 dB for the right channel. *Figure 2* is a spectrum analysis plot of noise as versus frequency and it is clear that most of the contribution to the overall S/N reading comes from the power supply frequency (60 Hz) and its harmonics (120 Hz, 180 Hz, 240 Hz, etc.) In any event, these hum and buzz contributions were so low as to be insignificant in audible terms.

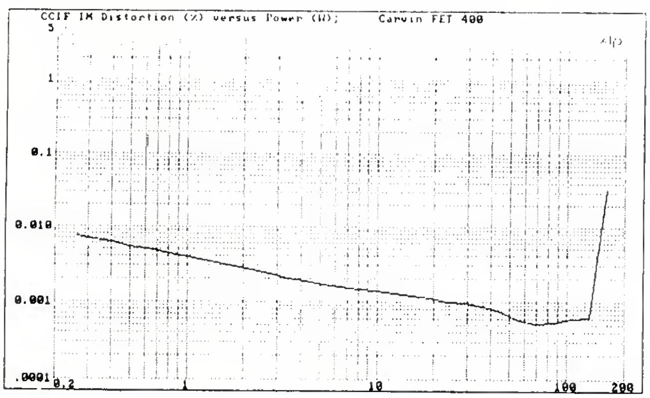


Figure 6A. CCIF-IM (two-tone) distortion vs. power output (8Ω loads).

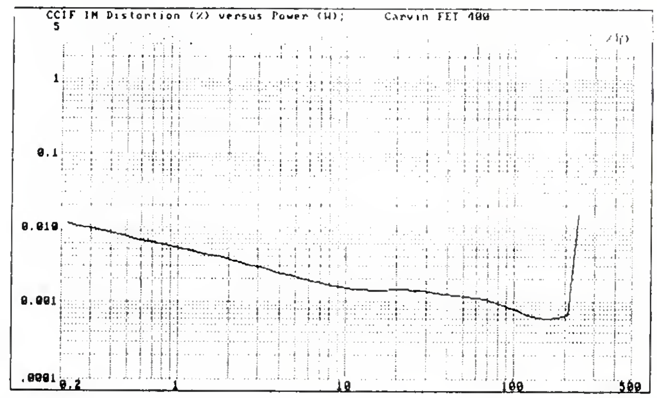


Figure 6B. CCIF-IM (two-tone) distortion vs. power output (4Ω loads).

Figures 3A and *3B* are plots of total harmonic distortion plus noise at rated output, using 8-ohm and 4-ohm loads, respectively. Using 8-ohm loads, as 1 kHz, THD plus noise measured only 0.028, rising slightly to 0.035 percent at 20 kHz. Using 4-ohm loads *(Figure 3B)*, the amplifier was not quite able to deliver the 200 watts per channel that it easily produced at mid-frequencies. Two sweeps were therefore made, the first at 200 watts per channel, the second at 180 watts per channel. In either case, at mid-frequencies, THD plus noise measured only 0.027 percent. Clipping occurred at the frequency extremes at the 200 watt per channel level, but when power level was reduced to 180 watts per channel, THD plus noise at 20 kHz remained low, with readings of only 0.052 percent.

Figures 4A and *4B* are plots of distortion plus noise as a function of power output, using a 1 kHz test signal. With 8 ohm loads *(Figure 4A)* the amplifier delivered 149 watts per channel before reaching the 0.1 percent distortion level, while with 4-ohm loads, power climbed to 218 watts per channel before reaching the claimed 0.1 percent distortion level *(Figure 4B)*.

Carvin does not specify values for rated SMPTE-IM or rated CCIF (twin-tone) Intermodulation distortion, but we measured those performance characteristics anyway. *Figures 5A* and *5B* show how SMPTE-IM varied as a function of power output per channel, using tones at 60 Hz and 7000 Hz in the usual 4:1 ratio. SMPTE-IM for 8-ohm loads hovered at around 0.01 percent over most of the measured power output levels, while for 4-ohm loads *(Figure 5B)* SMPTE-IM remained almost as low until power output levels exceeded the 200 watts per channel level. For the twin-tone IM measurements we used equal amplitude signals of 19 kHz and 20 kHz and results are plotted as a function of power output levels per channel in *Figures 6A* and *6B*. At rated output, for 8 ohm loads, (100 watts per channel) CCIF-IM measured an insignificant 0.0006 percent) while with 4-ohm loads the CCIF-IM at 200 watts per channel was only 0.0065 percent.

COMMENTS

Because Carvin FET series amplifiers are only sold directly to the end user by Carvin, the price of this excellent performing amplifier has been set at an amazingly low level of $449.00. When you compare the performance of this power amplifier with that of other amplifiers having similar power output ratings and features you begin to realize just how much of a bargain that price represents.

In our bench tests and subsequent listening tests, we subjected to amplifier to about as much abuse as it is likely to get in professional sound reinforcement installations. Yet, at no time did the amplifier's thermal protection circuits shut down the amp. After a particularly grueling session, the internal fan did go on, but it rotated at one of its slower speeds and never really was revved up to its maximum rpm. During more normal listening sessions such as might be used in the control room for monitoring of a recording session or of playback, the amplifier remained cool to the touch.

Internal layout of the amplifier was exemplary and was clearly the result of careful design and layout. Carvin, in the introduction to the amplifier found in the owner's manual, mentions the fact that this design was computer-aided, but even computer-aided designs have to be initiated and implemented by engineers and production personnel who know what they're doing. Carvin's design and production people obviously belong in that category. db

VITAL STATISTICS

SPECIFICATION	MFR'S CLAIM	dbMEASURED
Power Output		
Mono Bridged Mode		
8-ohm, 1 kHz, 0.1% THD	400 W	430 W
Both Channel Driven		
8-ohm, 1 kHz, 0.1% THD	100 W/ch.	149 W/ch.
4-ohm, 1 kHz, 0.1% THD	200 W/ch.	218 W/ch.
Rated THD (20 Hz to 20 kHz)	0.05%	0.035% @ 8ohms
S/N Ratio (re: rated output)	100 dB	105/108 dB
Frequency Response		
±0.5 dB	20 Hz to 20 kHz	10Hz to 50 kHz (See Text)
–3 dB	5 Hz to 100 kHz	Below 10 Hz to 85 kHz
Damping Factor	250	240
Input Sensitivity (4-ohms)	1.0 V rms	1.1 V rms
Input Impedance	10K ohms	Confirmed
Slew Rate (Volts/ μsec)	30	35
Max. Power Requirements		
@120VAC	4.5 Amps	5.0 Amps
@240VAC	2.25 Amps	N/A
Shipping Weight	28 lbs.	Confirmed
Dimensions (H x W x D, in.)	5-1/4x19x10	Confirmed
Price:	$449.00	

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Buyer's Guide

Consoles and Mixers

On the pages that follow, we present this issue's Buyer's Guide on consoles and mixers. The information contained is supplied by the respective manufacturers. Further, if a manufacturer that you seek is not listed, the chances are strong that, as many times as we tried, we could not get information from them.

ALLEN & HEATH BRENELL

SR series consists of the SR-8, SR-12, SR-16, SR-416, and SR-424 which are designed for sound reinforcement and basic 2 and 4-track recording applications. All units feature 4-band equalization and 4 auxiliary sends per unit, insert points on all input channels, sub-groups and left and right channels, and XLR outputs on left, right and mono outputs.

SRC Modular series consists of the SRC-416, SRC-424, and SRC-432 which are designed for professional sound reinforcement and 2, 4, and 8-track recording applications. Fully modular series comes standard with Alps faders and 8 channels of tape monitoring, and XLR electronically balanced outputs on 4 sub-groups, left, right, and mono channels. External power supply is rack-mountable.

SRM on-stage monitor mixers are available as the SRM-186 (18-in by 6-out matrix) and SRM-248 (24-in by 8-out matrix). Both units include a built-in passive mic splitting system and expanded facilities for the monitor mix engineer. Both mixers are built into road cases.

SYSTEM 8 series of 8-bus consoles designed for 8 and 16-track recording applications includes Alps faders as standard. Both the 1616D (16 input channels) and 2416D (24 input channels) feature 8 output subgroups with 16-channel tape monitoring and direct outputs on all input channels. An 8 input expander is available (EX-8).

CMC series of microprocessor controlled mixers available as the CMC-24 (24x16x2 with 16-channel tape monitoring) and CMC-32 (32x16x2 with 24-channel tape monitoring). Both muting functions (input and monitor) and bus assignment functions are under microprocessor control. 32 presets can be controlled from on board or an optional controller can be purchased that allows up to 2048 events to be controlled from SMPTE time code and also generates MIDI song pointer information derived form SMPTE.

Phantom series is designed in 8 and 16-bus formats for multi-track recording (up to 32-track monitoring depending upon modules chosen) and sound reinforcement applications. Customer choice of LED or mechanical meter bridge or LED metering built into sub-group modules for convenient transportation in sound reinforcement applications. All units are available with optional internal patch bay system and features as standard full MIDI muting functions on all input channels. Standard frames hold 36, 44, and 52 module positions.

Sigma split format console series is for sound reinforcement and recording applications. True 24-bus design in 2 frame sizes plus a 12 input expander frame gives unlimited size capabilities. Features include 4-band sweepable equalization on all input channels and choice of recording (dual) or sound reinforcement (with FX return section) sub-group modules, choice of mechanical or LED metering, and MIDI muting as standard. Can be configured for 8, 16 or 24-bus operation.

Sigma in-line console series with dual input capabilities per module, has 8 auxiliary buses that are supplemented by 24 additional post fader auxiliary sends during mixdown, MIDI muting as standard plus 32 muting presets, and choice of mechanical meters or LED bar graph meters (up to 32).

ALTEC LANSING CORPORATION

1674C (4-input) and the 1678C (8-input) microphone mixers are sound reinforcement "automatic" mixers (the main output is always equal to 1 open mic regardless of the number of mics in the system). Features include rack-mountability, link capability up to 40 mic inputs, gain sharing, transformer balanced mic inputs, phantom power, TTL compatible logic outputs for zonal speaker switching, channel line outputs for logging tape recorders, remote muting and priority override control, switchable 200 Hz high-pass filter on each mic input, manual mix capable, black anodized front panel.

AMEK SYSTEMS AND CONTROLS LTD. (TAC)

In the APC1000, each channel strip is freely assignable with reset of all switches and manual recall of all knob positions. A wide variety of frame sizes are available with up to 128 inputs. Standard automation is the GML (George Massenburg Labs) Moving Fader System.
Price:
from $200,000.00

G2520 is an advanced in-line console for 24 or 48-track operation, available in 2 chassis sizes, for 40 or 56 inputs. Suitable for recording or for use in video and broadcast production studios. Range of optional automation systems available including GML moving fader automation.
Price:
from $120,000.00

Angela is available in a wide variety of frame sizes and configurations for 24 or 48-track operation. Three chassis sizes are available with 28 to 62 inputs. The number of inputs can be doubled for mix-down. Optional d.c. subgrouping and fader automation.
Price:
from $47,000.00

Classic is an advanced broadcast and video production console system available in a wide variety of configurations featuring 8 auxiliary sends, 8 buses, 2 stereo outputs and optional VCA faders. Nominally available in 32, 48 and 64 input versions. Mono and stereo inputs are available. Optional automation.
Price:
from $80,000.00

BCII is a high performance flexible console system for the broadcast and post-production industries. Available in transportable, drop through or studio chassis. The configurations are diverse to suit a wide variety of applications. The system also features a post for AFV (audio follows video) which can be used in conjunction with the ESM1000 edit suite control interface.
Price:
from $13,000.00

TAC Scorpion is a multi-purpose mixing system with a multitude of possible configurations. Suitable applications include recording, sound reinforcement, broadcast and post production. Over 50 standard formats are available in 3 chassis sizes. Latest addition to the range is the S1200 stereo input module. Available with optional automation.
Price:
from $6,600.00

Matchless is a fully modular 24-bus multi-track recording console. Currently available in 2 frame sizes, formats of up to 36 inputs are available. In mix-down all monitor inputs can be utilized as additional line inputs totalling 72 channels. Available with optional automation.
Price:
from $25,000.00

SR9000 has been built for sophisticated sound reinforcement application. The console features a 40 input mainframe, 16 audio sub-groups, 8 mute groups, 8 VCA groups, 16 auxiliary sends, 16x8 matrix and parametric equalization. Expansion up to 72 inputs available.
Price:
from $70,000.00

AUDIO LOGIC

SC 601 is a rack mounted 6-input, 1-output mixer. Four inputs are mic level only, 2 input may be switched between mic or line level. Phantom power and XLRs on all inputs and an input level trim on each input are provided. Output is mic or line level switchable. Units may be cascaded. Dimensions are 1.75x19x8. Weight is 4.5 lbs.

AUDIO-TECHNICA U.S. INC.

AT4462 portable stereo field production mixer features 2 mono, pannable and 2 true stereo mic or line inputs, stereo limiter, 12 volt phantom power, and MODU-COMM, an IFB system which operates common mode over the first 2 mic lines with addition of optional MODU-COMM decoder.
Price:
$1,295.00

AUDITRONICS INC.

200 Series console has 6, 12, 18 and 24 in, 2 out. It is designed for broadcast production and on-air. Automation and signal processing are optional.
Price:
$12,000.00 to $30,000.00

400 Series console has 18, 24 and 32 in, 4 and 8 out. It is designed for broadcast production and multi-track. Automation and signal processing are optional.
Price:
$31,000.00 to $60,000.00

310 Series has 16, 24 and 32 in, 4 and 8 submasters and 4 sub-groups. Features include stereo and mono outs. Automation and signal processing are optional.
Price:
$35,000.00 to $75,000.00

BIAMP SYSTEMS

Rackmax is a high performance, high density 16-input rack mount mixer with studio console specifications. 48 volt phantom power is switchable on each channel, and there are 100 mm faders, a complete solo system, and 3 auxiliary sends per channel, with LED ladder metering. The mixer has available an optional Integral Digital Reverb with 16 programs including digital readout and control.
Price:
$2,099.00
$2,349.00 (with IDR)

883RX is a highly compact, ultra low noise mixer with 8 inputs. The output section is organized into main, monitor and 2 submasters, with 3-band equalization on each input, and 10 segment switchable LED ladder metering. Floating and balanced output is provided, as is a four transistor, discrete front end and 15 volt phantom power. Optional IDR is available with 16 programs and built-in digital readout and control.
Price:
$1,099.00
$1,349.00 (with IDR)

24 series is a highly flexible mixing console with switched signal routing and tape returns on all input channels for use in live or recording applications. It has high slew rate, and low distortion electronics, with 12 segment LED ladder metering, and 100 mm faders, 3-band equalization with sweepable mid-band, one stereo and 2 mono sends and standard 48 volt phantom power. Available in 12, 16 or 24 inputs mixing to 4 submasters, left, right and mono outputs.
Price:
$2,999.00 (1224)
$3,799.00 (1624)
$5,399.00 (2424)

28 series is a highly flexible mixing console with switched signal routing (SSR) and tape returns on all input channels for use in live or recording applications. It has high slew rate, low distortion electronics, 12 segment LED ladder metering, 100 mm faders, and 3-band equalization with sweepable midranges. There are one stereo and 2 mono sends, standard 48 volt phantom power and it is available in 24 or 32 inputs mixing to 8 submasters, left, right and mono outputs.
Price:
$5,999.00 (2428)
$7,699.00 (3228)

Legend series are a high performance, modular, in-line recording console series available in configurations ranging from 12 to 32 inputs and 8 to 24 submasters in 2 frame sizes,with 3-band equalization with all bands fully sweepable and true shelving filters on high and low filters. There are 12 segment, high intensity LED switchable meters on every channel, complete control room and studio communications, playback and tape monitoring systems, 48 volt phantom power switchable on each channel, 100 mm faders, with optional Penny & Giles 3220 conductive plastic faders, stereo monitor send and 4 effects/cue sends. Mixpak series are small powered mixers that feature all steel and high impact injection molded plastic design with a special high speed PLUS + channel for the special requirements of digital electronic instruments such as synthesizers and drum machines. They provide 250 watt power into 4 ohms with built-in auto limit for speaker protection, 9-band graphic equalization on output, and optional rack wings and travel cover. 6-channel (5 +), 7-channel (6 +), and 8-channel (7 +) models.
Price:
$699.00 (5 +)
$749.00 (6 +)
$799.00 (7 +)
$35.00 (rack wings)
$18.00 (cover)

Stereo DJ5001 is a flexible DJ mixer with 9 inputs, 6 line level and 3 phono, one of which is switchable to line. LED visual beat sync. 3-band of equalization on each program channel, rumble filter and switchable bass impact circuit, switchable LED ladder metering package are all standard. DJ mic talkover with separate equalization and automatic program attenuation are also provide as are one mono and 2 stereo outputs, effects loop on both DJ mics and main outs, and balanced and floating outputs.
Price:
$999.00

CALREC by AMS

UA8000 Music console is available with frame sizes of 32, 48, 56, 64, or 72 channels. AMS Studio Computer Automation is standard.
Price:
On application

M Series Mixer production or post production mixers are customized for particular applications. Features include low noise mic amplifiers, 3 or 4 band parametric equalizers, and dynamic modules based on the UA8000 system. Digitally Assignable Mixers have total instant reset for video production/post production. Features include free assignment of fader for unlimited flexibility, instant memory reset of all console settings, and multiple disc based memories.
Price:
On application

CARVIN CORPORATION —See our ad on page 13

MX22 series of stereo sound mixers are designed for live performance. Six models are available from 6 to 24 channels, featuring MOSFET technology, 800 watts RMS (powered models) and forced air cooling. It has XLR balanced inputs and outputs with channel patching for recording and effects. Dimensions are 8.5x26.5x 23 (6 & 8 channel), 29 (12 & 16 channel), 39 (24 channel) x26.5. Weight is 55 lbs. (6 channel), 59 lbs. (8 channel), 69 lbs. (12 channel), 73 lbs. (24 channel).
Price:
$999.00 (MX622S)
$1,149.00 (MX822S)
$1,449.00 (MX122S)
$1,489.00 (MX1622)
$2,099.00 (MX2422)

MX1644 is designed as 4-track recording and live sound mixer. Featured is 16-input channels including 4-band active equalization and 4 auxiliary mixing buses. Hammond reverberation system is standard. The unit has VU meters, talkback with built-in mic and monitor dimming +4 or –10 operating level with THD better than 0.03 percent, and totally modular internal construction. Dimensions are 8.75x35.13x29. Weight is 70 lbs.
Price:
$1,895.00

MX1688 and MX2488 are 16 and 24-channel full function recording mixers designed for 8-track recording studios, live sound mixing, production studios, broadcast and sound mixing for film. Input channels include 3-band parametric equalization with defeat switch, 4 auxiliary mixing buses with pre/post switching and solo/mute functions. The unit has highly modular construction. Dimensions are 9, 36 (16 channel) 46 (24 channel), 29. Weight is 80 lbs. (16 channel), 115 lbs. (24 channel).
Price:
$3,395.00 (MX1688)
$3,995.00 (MX2488)

MX621 offers 200 watts rms and the MX641 delivers 400 watts rms for concert applications. Featuring 6 balanced XLR inputs with low equivalent input noise of –126 dbm, 6 buffered 0.25-inch unbalanced inputs for high impedance instrumentation or mics. A built-in 48 volt phantom supply on the MX641 features the use of condenser mic. Dimensions are 9.8x19. Weight is 40 lbs.
Price:
$499.00 (MX621)
$599.00 (MX641)

FET series of amplifiers feature the latest in MOSFET technology. It features heavy-duty power supplies with high current transformers and filters. Performing beyond the requirements of FTC ratings, it utilizes a speaker guard circuit protecting from d.c. voltages. The amps can be switched to mono mode to double the output voltage and feature electronic speed controlled fan. Dimensions are 5.25x19. Weight is 28 lbs. (FET400), 35 lbs. (FET900).
Price:
$449.00 (FET400)
$599.00 (FET900)

DOD ELECTRONICS CORPORATION

R-855 is a rack mounted, 4-input, stereo output mixer, 1 rack space high, with pan pots, a headphone output, a master level control, an effects send/receive loop, and a clipping indicator. Inputs allow bridging another R-855 mixer or inputs from other sources. Dimensions are 1.75x19x8. Weight is 4.5 lbs.
Price:
$279.00
$299.00 (XLR version)

ELECTRO-VOICE

BK32 series stereo mixing consoles are available in 8 (rack), 12, 16 and 24-channel configurations. Applications include sound reinforcement and recording. Other features offered are 3 sends per channel (effects, monitor and auxiliary), phantom power, 3-band equalization, reverb to monitor, insert jacks on all inputs and subs.
Price:
$1,040.00 (BK832)
$1,250.00 (BK1232)
$1,495.00 (BK1632)
$1,931.00 (BK2432)

FOSTEX CORPORATION —See our ad on page 9

450-8 is an 8-track recording mixer with parametric equalization, 8 inputs, 4-channel bus, stereo bus, mono bus, switchable phantom power on each channel, and in-line monitoring.
Price:
$1195.00

450-16 is a 16-track recording mixer with parametric equalization, 4-channel bus, stereo, mono and solo buses, in-line monitoring, and switchable phantom powering on each channel.
Price:
$2195.00

260 Multitracker is a 4-track cassette mixer/recorder with 6 inputs. It has independent stereo bus, 2 mono buses, 3.75 in./sec. tape speed, Dolby C, parametric equalization, and true rolling punch-ins.
Price:
$1195.00

160 Multitracker is a 4-track cassette mixer/recorder with 4 channel simultaneous recording and accessory patch points.
Price:
$840.00

X-30 is a 4-track cassette mixer with Dolby B and C noise reduction. The mixer section is 4x2 dedicated sub-mixer for overdubbing and bouncing tracks.
Price:
$499.00

460 is a multi-track cassette mixer capable of synchronization with video recorders. The mixing section contains 8 inputs, 4 bus outputs, dedicated stereo mixer for the 4-channel bus, selectable monitoring, switchable LED bar graph metering and accessible patch points for flexible system interface. The recorder section features true 2-speed transport (separate record/equalization circuits for 1.87 and 3.75 in./sec.), Dolby B and C noise reduction, 2-position autolocate, search to zero, auto repeat and SMPTE/EBU synchronization capability.
Price:
$2,495.00

FURMAN SOUND INC.

Rackmount mixers (MM-4A and MM-8A) have 4 inputs, mono (MM-4A) or stereo (MM-8A) outputs and include pan pots on each MM-8A input, effects bus with send and return jacks, and stereo auxiliary inputs with RCA jacks and level control. "B" models feature balanced ins with both phone and XLR connectors, mic/line switches. "BP" models feature 48 volt phantom powering on all inputs and phantom power switch. Suitable for sound reinforcement or recording. Dimensions are 1.75x19x8. Weight is 6 lbs.
Price:
$335.00 (MM-4A)
$375.00 (MM-4AB)
$405.00 (MM-4ABP)
$395.00 (MM-8A)
$435.00 (MM-8AB)
$465.00 (MM-8ABP)

GOTHAM AUDIO CORPORATION (AUDIO DEVELOPMENTS, LTD.)

AD 160 is a mono engineering mixer with the following features: 3 mic inputs, 1 line input (all are balanced), 1 line output (balanced), limiter on line output, phantom/A-B powering on mic inputs, 1 kHz oscillator, talkback mic, VU or PPM meter, monitor output, battery or a.c. (with optional a.c. adapter). Dimensions are 2.2x8.5x6.5. Weight is 3.3 lbs.
Price:
$1,765.00

AD 260 stereo engineering mixer is the same as above with the following differences: 4 mic/line inputs with pan pots, 2 line outputs, 2 limiters on line output (linkable for stereo), 1 stereo auxiliary input. Dimensions are 2.2x11.9x7.9. Weight is 5.5 lbs.
Price:
$2,500.00

AD 145 Pico mixer has 4-8 mic/line inputs with pan pots (all are balanced), 2 line outputs (balanced), 3-band equalization on each input, phantom/A-B powering on mic inputs, 1 kHz oscillator, talk back mic, monitor output, cue input. Dimensions are 4.8x13.8x10.3. Battery or a.c. (with optional adapter) Weight is 13.2 lbs.
Price:
$4,400.00 to $6,775.00

AD 062 multi-mixer has 4-16 mic/line input modules with pan pots (balanced), 2 line outputs, 3-band equalization on each input, phantom/A-B powering on mic inputs, auxiliary send on each input module. Options include stereo line input module to replace mic/line input module, dual auxiliary return module, communications module, stereo compressor/limiter module. Standard features include monitor level, selector controls, master auxiliary sends, A+B mixdown, Battery or a.c. (with optional a.c. adapter). Dimensions are 16x16x5.4. Weight is 24.2 lbs.

Price:
$6,000.00 to 20,000.00

HILL AUDIO INC.

Multimix can be used in a 16x4x2x1, 12x4x2x1 or 16x2x1 configuration. Designed for recording, broadcast or sound reinforcement. The unit is an 8 space, 16 input, semi-modular rack mount mixing console, with the following features: 48 volt phantom power, RIAA equalized inputs, direct outputs on all 16 channels, 3-band equalization, 2 auxiliary sends, 100 mm faders. The console is 4 inches deep, and the power supply is rack mountable. Weight is 35 lbs.
Price:
$2,399.00

Soundmix semi-modular multi-purpose console for recording, broadcast or sound reinforcement is available in a 24x4x2x1 or 16x4x2x1 configuration. Standard features are: 4-band equalization, 2 pre-fade auxiliary sends and 2 post-fade auxiliary sends, 4 auxiliary returns, 100 mm faders (available with Alps or Noble), 48 volt phantom power, direct outputs on all channels and subgroups, insert points on all channels/subgroups and master outputs, PFL system with VU meter and 12 segment LEDs on the subgroups and masters, plus the power supply is rack-mountable. Dimensions are 4x41x24.5 (24 channel), 4x32x24.5 (16 channel). Weight is 75 lbs. (24 channel) and 55 lbs. (16 channel).
Price:
$4,999.00 (24 channel with Alps faders)
$4,499.00 (24 channel with Noble faders)
$3,899.00 (16 channel with Alps faders)
$3,499.00 (16 channel with Noble faders)

Stagemix is a 12x6 rack mount monitor console. This unit is a semi-modular rack-mount console featuring: built-in mic splitter using balanced mic inputs on transformer isolated, zero loss, parallel XLRs, transformer balanced group outputs, 3-band equalization on all inputs and 4-band equalization on all outputs at different center frequencies-giving 6 bands of equalization control, 12 segment LED group displays, PFL and AFL function on all outputs. This console is perfect for mixing your own monitors on stage. Fits in 8 rack spaces and is 4 inches deep. Weight is 35 lbs.
Price:
$2,999.00

Concept series 2200/3200/4400/5400/6400 and 8400 can be used in recording, broadcast or sound reinforcement. This is a modular console available in almost any configuration and comes standard with the Sidetraker equalization system. This system uses 6 fixed frequency active filters sweepable from 50 Hz to 5 kHz and adjustable from −24 dB to +6 dB. The 2200 and 3200 come with 8 auxiliary sends, 100 mm Alps faders and are available in 8 or 16 bus. The 2200 series is also available with an 8x8 matrix. The 4400, 5400, 6400 and 8400 have Sidetraker equalization plus 12 auxiliary sends, optional d.c. subgroups, 2 programmable mute systems, true solo-in-place and are available in 8, 12, 16 or 24 bus and up to 48 tape returns. The 4400 is available with an 8x8 matrix.

HM ELECTRONICS, INC.

MX55 4-channel stereo mixer has 1 mic, 1 line and 2 phono inputs with a cross-fader. Tape monitor function may be switched in and out. Features include preset guide for gain settings, cue/program headphone monitor, tape monitor and talkover mic. Dimensions are 5.25x19x7.4. Weight is 8.1 lbs.
Price:
$384.00

MX77 4-channel equalized stereo mixer has 1 mic, 2 line, and 2 phono inputs with cross-fader, 3-band graphic equalization and output selection. Tape monitor may be switched in and out. Features include preset guides for gain settings, cue/program headphone monitor and talkover mic. Dimensions are 5.25x19x6.3. Weight is 9.6 lbs.
Price:
$494.00

MX99 has 5 bands of graphic equalization and signal processor circuitry (both bypassable). Features include 4-channel inputs, cross-fader between 2 phono inputs, preset guides for gain settings, cue/program headphone monitor, and line/phono selection switch. Dimensions are 8.75x19x3.95. Weight is 9.6 lbs.
Price:
$534.00

MX1 6-channel stereo mixer has 6 channels of mixing with 9 selectable inputs: 3 phono inputs, 3 mic inputs, and 3 line inputs. The main output features a "pan" control for stereo balancing. Dimensions are 8.75x19x3.3. Weight is 9.8 lbs.
Price:
$749.00

MX56 4-channel stereo powered mixer features talkover mic, cross-fader between 2 phono inputs, line and tape monitor inputs and preset guides for gain settings. Dimensions are 5.25x19x11.75. Weight is 18 lbs.
Price:
$860.00

MX81 6-channel stereo mixer has three mics, 3 phonos, 2 lines, 2 tapes, and 1 CD provide flexibility in input sources. Stereo balancing for each channel, signal processor loop, DJ mic with pan, bass, and treble controls are a few of the features on this unit. Dimensions are 8.75x19x5. Weight is 11.9 lbs.
Price:
$899.00

MX10 is an 8-channel stereo mixer. There are 8-channel inputs with eleven selectable sources. Individually controlled channel input attenuators, pan controls, tone controls and high pass filters make this a versatile mixer. Both signal processor sends can be used simultaneously. Dimensions are 8.75x19x5. Weight is 12.3 lbs.
Price:
$950.00

INNOVATIVE ELECTRONIC DESIGNS, INC.

IED 4000 series automatic microphone mixer is modular, allowing its components to be used to configure systems from the smallest to the very largest. Mainframes may be linked for even greater capacity. Linking of mixer cards or mainframes may be fixed, or may be varied under manual or computer control. The unit also features programmable gain control.

IED 5000 series is a modular structure consisting of one 3.5x19 rackmount mainframe which holds up to 13 modular plug-in printed circuit cards. Utilizing the 5000 series cards, an engineer can "building block" a system to suit any specific application.

KLARK-TEKNIK (DDA)

DCM-232 in-line recording/production console has a dynamic range in excess of 100 dB with 22 dB of headroom at any stage, incorporates split equalizer design, Central Automation Terminal (C.A.T.), and optional fader automation. Maximum configuration is 56/32, VU meters, and dimensions are 42x49.2x122.
Price:
$146,520 (other configurations available).

AMR024 recording console uses split console design with modules having full equalization, mute and auxiliary facilities and can handle two 24-track machines simultaneously for mixdown, balanced mix buses reduce crosstalk and noise. Maximum configuration is 44/24/2, and dimensions are 41x45.5x122.
Price:
$79,750 (other configurations available).

D-Series consoles utilize electronically balanced inputs and outputs. The multiple earthing system and quasi-differential mix buses result in a quiet mixer. Versions are available for recording, theater, PA, monitor and broadcast. Maximum configuration for recording version is 42/8, and dimensions are 14.4x38x85.75.
Price:
$25,740 (other configurations available)

S-Series consoles utilize a multiple earthing system and electronically balanced inputs/outputs for noise and crosstalk specifications. System headroom is typically 22 dB at any point. It is available in standard, PA and monitor version. Maximum configuration for standard version is 32/4/2, and dimensions are 11.5x29.5x61.25.
Price:
$11,250.00 (other configurations are available).

Q-Series is able to access a direct output from each channel via a "direct" switch and the Auxiliary 1 level control. The matrix module provides an 8x4 matrix system, in addition to an auxiliary return section with 3-band equalization and subgroup routing. Maximum configuration for PA version is 40/8/2, and dimensions are 10.5x29.5x70.
Price:
$20,750.00 (other configurations available)

MITSUBISHI PRO AUDIO GROUP

SuperStar recording console has dual in-line I/O modules, 44 to 84 inputs, and 32/64 centrally assigned mixing buses. There are also 16 auxiliary send buses, 2 stereo outputs, modular bolt together aircraft frame, field expandable, optional meter overbridge for peripheral and in-line devices,with VU, or 60 segment LED bargraph meters. Selectable top panel plug-in equalizers, selectable top panel plug-in preamplifiers, selectable plug-in VCA. Audio, VCA, automated and intelligent digital faders are all standard as is Compumix PC automation, 20 Mbyte, and moving fader automation, 30 Mbyte.
Price:
from $139,000.00 to $354,000.00

Westar 8000 recording console has dual in-line I/O modules, 20 to 52 inputs, 24 mixing buses, 8 auxiliary send buses, 2 stereo outputs a modular bolt together aircraft frame, field expandable, and VU, or 60 segment LED bargraph meters. Selectable top panel plug-in equalizers, selectable top panel preamplifiers selectable plug-in VCA, Audio, VCA, automated and intelligent digital faders are all standard, as are Compumix PC automation, 20 Mbyte moving fader automation, 30 Mbyte.
Price:
from $70,000.00 to $225,000.00

Westar 8300 film re-recording console is available in 16 to 72 inputs. Features include 8, 16, or 24 mixing buses, 10 auxiliary send buses 3-channel pan bus, and a 2-channel pan bus film monitor system with 8x4 to 24x8 matrix select systems with dedicated monitor format buttons, recorder and bus/film pushbutton control panel(s) 8 to 24 tracks, reassign, transfer key, and mono and stereo composite modules, multi-track pre-dub input modules, a modular bolt together aircraft frame, field expandable, VU, or 60 segment LED bargraph meters, selectable top panel plug-in equalizers, and selectable plug-in VCA. Audio, VCA, automated and intelligent digital faders, are standard as are Compumix IV film automation, 85 Mbyte, 1 to 4 sections, moving fader automation.
Price:
from $89,000.00 to $600,000.00

NEVE

"V" series console is a multi-track recording console for the music, video post and film industries, available in 36, 48, 60 or 72 inputs, including the Formant Spectrum Equalizer, mic/dynamics unit and 8 mono/4 stereo auxiliaries. Additional benefits include a centrally positioned monitor path status indication to enable rapid console status checks and choice of metering options,

DTC (Digital Transfer Console) for compact disc mastering and is used for the preparation of compact disc master tapes, the unit also provides total digital mixing and processing capabilities. All console parameters can be instantly reset under SMPTE control, also permitting the user to select or mix either of 2 stereo digital inputs and 1 stereo analog input with manual or auto crossfade from AES/EBU or 1610/1630 inputs to compatible outputs.

8232 console for TV production, post-production and multi-track recording has 32 mic/line input channels with 24 mixing buses and optional stereo reverb returns. Each channel features the Formant Spectrum Equalizer, 4 mono auxiliary sends and 1 stereo cue send.

542 console comprise a range of small compact audio mixing consoles for broadcast and post-production applications. With 8, 12 or 16 input channels and 2 playback inputs, the console provides 2 main stereo program outputs with a mono facility, 2 auxiliary outputs and a comprehensive stereo/mono monitoring system.

Necam 96 is a computer-assisted moving fader automation system that is also available for fitting to non-Neve consoles, controlling up to 96 faders, and instinctive mixing, completely free grouping, unlimited mute groups, unique intelligent rollback. It keeps the pass as a virtual mix for review or update. Auto merge and merge-off-line capabilities.

Prism series is a range of rackmount units derived from the V series console comprising a 4U 19-inch rack with capacity for 10 modules that may be powered from an existing console or by a 2U power supply. The 2 modules are the Formant Spectrum Equalizer and the mic amp/dynamics unit comprises compressor/limiter/gate/expander.

51 series console is for stereo broadcast, they comprise models that are suitable for studio and vehicle installations. The console is designed for use in either mono or stereo configuration and incorporate the Formant Spectrum Equalizer, available in 12, 24, 36 or 48 input channels with 8 sub-groups and 4 main outputs, high and low pass filters and limiter/compressor.

PEAVEY ELECTRONICS

MDII 8/12/16 sound reinforcement consoles are available in 8, 12 and 16 channels. Features include XLR balanced inputs, –133 dBv input stage S/N ratio, 100 mm faders, pre-monitor send, post effects send, built-in reverb, stereo left and right outputs and master (mono) output. Dimensions are 5x19.88x25.75 (8), 5x27x25.75 (12), 5x32.75x25.75 (16). Weights are 24, 28 and 31 lbs. respectively.
Price:
$749.50 (8)
$949.50 (12)
$1149.50 (16)

MDIIB 12/16 sound reinforcement consoles feature XLR balanced inputs, –133 dBv input stage S/N ratio, 100 mm faders, pre-monitor send, post effects send, built-in reverb, transformer balanced outputs for left and right, monitor and master. Dimensions are 5x27x25.75 (12), 5x32.75x25.75 (16). Weights are 28 and 31 lbs. respectively.
Price:
$999.50 (12)
$1199.50 (16)

Mark III 16/24 sound reinforcement consoles come in 16 and 24-channel versions. Features include transformer balanced XLR inputs, 2 pre-monitor sends, 2 post effects sends, cue system, internal reverb, pre and post channel patch, 4 10-segment LED arrays, flight case construction. Dimensions are 6.75x35.63x26 (16), 6.75x47.63x29 (24). Weights are 71 and 90 lbs. respectively.
Price:
$1999.50 (16)
$2499.50 (24)

MS-1221/1621/2421 sound reinforcement consoles come in 12, 16 and 24 channels versions. Features include balanced XLR inputs, 2 pre-monitor sends, 2 post effects sends, built-in reverb, built-in electronic delay, 4 9-band graphic equalizers, transformer balanced outputs for mains and monitors, 48 volt phantom power. Dimensions are 8.06x30.19x29.25 (12), 8.06x36.25x29.25 (16), 8.06x48.5x29.25 (24). Weights are 41, 43 and 58 lbs. respectively.
Price:
$1799.50 (12)

$2099.50 (16)
$2699.50 (24)

Mark IV 16/24 sound reinforcement consoles are available in 16 and 24-channel versions. Features include transformer balanced XLR inputs, channel assignment, 4 submasters, internal reverb, 2 pre-monitor sends, 1 post effects send, intercom system, cue system, 4 outputs, transformer balanced outputs for subs and monitors, flight-case construction. Dimensions are 6.75x38.38x29 (16), 6.75x50.38x29 (24). Weights are 74 and 95 lbs. respectively.
Price:
$2499.00 (16)
$2999.00 (24)

Mark IV 24x8 sound reinforcement monitor console is a 24-channel version. Features include transformer balanced XLR inputs, input splitter (low Z), 4-band equalization, solo, mute, 8 separate transformer balanced XLR outputs, headphone system, intercom system, flight-case construction. Dimensions are 7x54.13x29.13. Weight is 103 lbs.
Price:
$3499.00

SRC-421 16/24 sound reinforcement console is available in 16 and 24-channel versions. Features include balanced XLR inputs, 4 separate balanced outputs (XLR), channel assignment, auxiliary send (pre), 2 post effects sends, 3-band equalization with sweepable midrange, 100 mm faders, PFL, pre-send and return patch. Dimensions are 5x36.25x25.25 (16), 5x48.25x25.25 (24). Weights are 40 and 44 lbs. respectively.
Price:
$1699.50 (16)
$2099.50 (24)

RAMSA-PANASONIC

WR-8616 is a broadcast and recording mixing console with modular construction so it can be customized. It includes versatile monitoring, 3-band sweepable equalization, and 100 mm faders.
Price:
starts under $10,000.00

WR-T820B is a recording console with 48 input capability for multi-track applications. It has up to 20 mic/line and 28 line inputs in mixdown mode, 8 addressable auxiliary outputs, MRP 300,000-operations faders, and high-performance electronics.
Price:
$8,500.00

WR-S208, WR-S212 and WR-S216 are 8, 12, and 16-input channel by 3-output consoles that feature 3-band equalization and bar graph style LED metering. WR-S208 is rackmountable.
Price:
$1,600.00 (WR-S208)
$2,200.00 (WR-S212)
$2,700.00 (WR-S216)

WR-8112 and WR-8118 are 12 and 18-input channel consoles that are for recording and sound reinforcement. They include 3-band equalization, 12-point LED bar graph metering, and 7 outputs.
Price:
$2,900.00 (WR-8112)
$3,900.00 (WR-8118)

WR-S840 series of professional mixing consoles are modular and features include Fiberglas-epoxy circuit boards throughout the unit, comprehensive 4-band equalization, sweep-frequency high-pass filters, long-life MRP TM controls, ribbon wire busing and gold contacts for reliability and low noise, and stage monitor modules that turn the unit into a 40x18 stage monitor board.
Price:
$31,500.00

WR-8428 is a post-production and recording console with dual 24-track capabilities, low noise and distortion, and 75 dB of crosstalk rejection.
Price:
$8,180.00

WR-8210A is a compact recording console for 4 and 8-track recording and features switch-selectable mixdown (without re-patching), a sub-mix for headphones, each channel has 3-band sweepable equalization, and there are 12-point bar graph-type LED meters.
Price:
$2,500.00

WR-133 is a portable 8x2 audio mixer for recording or reinforcement and has re-configurable pre/post effect and monitor sends, balanced mic/line inputs, phono inputs, and balanced and unbalanced outputs.
Price:
$1,200.00

SHURE BROTHERS INC. —See our ad on page 29/31

Audiomaster 1200 Powermixer is a 6-input, 1-output 200-watt sound reinforcement mixer, expandable up to 10 inputs. Rack-mountable; portable road case available. Each transformer-balanced input has input attenuation, overload LED, 2-band equalization, reverb send, and simultaneous low and high Z inputs. 3100 (permanent installation) and 3200 (portable) speakers are available. Dimensions are 7.5x19x13.5. Weight is 27 lbs.
Price:
$930.00

FP51 gated compressor-mixer is a portable 4-input, 1-output mixer for broadcast, recording, and sound reinforcement. True average-responding compressor has 40 dB range, adjustable response rate, and a gated memory function. Features include phantom power, tone oscillator, and a triple-function VU meter. Fully mic/line switchable and transformer-coupled; may be powered via a.c. or built-in battery pack. Dimensions are 3.13x12.22x9.03. Weight is 6.06 lbs.
Price:
$940.00

FP42 stereo mic mixer is a portable 4-input, 2-output stereo mixer for broadcast, recording and sound reinforcement. Inputs are transformer coupled, mic/line switchable with active gain controls, low-cut filters, detented pan pots. Outputs are transformer-coupled, with mic/line and mono/stereo switches. Features include phantom power, pull-pot cuing system, tone oscillator, limiter, dual VU meters and a.c. or battery operation. Dimensions are 3.13x12.22x9.03. Weight is 6.5 lbs.
Price:
$990.00

FP32 stereo engineering mixer is a lightweight, compact 3-input, 2-output stereo mixer for electronic news gathering and field production. All inputs and outputs are transformer-coupled XLR-type, mic/line switchable. This unit features active gain controls, detented input pan pots, phantom power, adjustable limiter, dual VU meters, slate mic and tone. It runs 6 hours on 2 9-volt alkaline batteries. Removable shoulder strap, carrying case included. Dimensions are 2.31x7.25x6. Weight is 2.5 lbs.
Price:
$1,350.00

FP31 is a compact and lightweight 3-input, 1-output engineering mixer for electronic news gathering and field production. Fully transformer-coupled and mic/line switchable, this unit features phantom power, slate mic and tone, selectable low-cut filters, adjustable limiter and VU meter, and 1 kHz test tone. It runs 8 hours on 2 9-volt alkaline batteries. Dimensions are 1.88x6.31x5.31. Weight is 2.2 lbs.
Price:
$990.00

M267 mic mixer is a portable 4-input, 1-output mixer for broadcast, recording, and sound reinforcement. All channels are fully transformer-coupled and switchable for balanced mic-level or line-level operation. Features include adjustable peak limiter, phantom power, active gain controls, tone oscillator, built-in battery pack, multi-range VU meter, and a mix bus jack for stacking multiple units. a.c. or battery operation. Dimensions are 2.97x12.16x9. Weight is 5.13 lbs.
Price:
$520.00

M268 mic mixer is a portable 5-input, 1-output mixer for sound reinforcement, recording, and audio-visual applications. All inputs and outputs are fully transformer-coupled with active gain controls. Four inputs accept both high and low Z mics, and 1 input accommodates high-level auxiliary sources. Features include 30Vdc phantom power and a mix bus jack for stacking multiple units. a.c. or battery operation. Dimensions are 2.97x12.16x9. Weight is 5.13 lbs.
Price:
$290.00

SOLID STATE LOGIC

SL 4000 G series master audio system is for multi-track music recording and mixing. In-line console available with 24 to 72 input/output channels. Four-band parametric equalization and dynamics on all channels. Features patch-free audio subgrouping, 8 VCA control groups, and the series studio computer featuring fast processors, vast on-board memory and high capacity disc cartridges. Options include total recall computer, events controller and synchronizer.

SL 5000 M series audio production system is for broadcast and film applications. Features include a modular console architecture with totally electronic switching, complete custom configurability and a wide range of Eurocard-style audio and control cassettes. The system is capable of supporting stereo broadcasting and all current film release formats. The system will handle large numbers of outputs and features a highly flexible internal bus structure, sophisticated routing assignment and panning systems, and a versatile monitoring system. Options include instant reset computer.

SL 6000 E series stereo video system for audio production and post-production applications. In-line system with 3 stereo mix buses plus a main stereo programme bus. Available with 24 to 72 I/O modules and featuring 4-band parametric equalization and dynamics in all channels.

SONY PROFESSIONAL AUDIO

MXP-3000 series console has been designed to meet the demands of digital audio recording. The series includes the MXP-3020 (20 I/O modules), MXP-3036 (36 I/O modules), and the new MXP-3036 VF. Features common to all 3 include user configurability with a choice of 5 different equalizers, 4 different mic preamps and hard disc automation.
Price:
$47,000.00 to $99,000.00

MXP-3036 VF is designed with a vacuum fluorescent (VF) light meter that displays various selectable scales including VU, BBC peak, DIN peak, Nordic peak, and a d.c. scale which indicates fader position in the automated version of the MXP-3036 VF. The automated version includes newly-developed version 2.0 software and optional wild faders that permit a user to increase the number of effects in a mix.
Price:
$117,000.00

MXP-2000 series audio consoles is designed for broadcast and post-production applications. Included in this series are the MXP-2016, featuring a 20-module frame size; the MXP-2026, which has a 30-module frame; and the MXP-2026, whose frame can hold as many as 40 modules. For each console, the user can install up to a maximum of 4 group modules with 12 channels, 22 channels or 32 channels, respectively. Additional features include 2 independent stereo outputs, 4 stereo external monitor inputs, 5 external talkbacks, and a user-assignable 4-channel dynamics processor and video editor interface capability.
Price:
$20,000.00 to $38,000.00

MXP-29 is an 8-channel audio mixer and can be controlled from the BVE-900 editing control unit. The unit offers trim control for each balanced mic/line input, built-in 3-band equalization, and VU meters with 15 segments of LEDs.
Price:
$3,849.00

MXP-21 8-channel audio mixer is designed for audio-for-video applications. It features built-in 3-band equalization and low-cut filter, and 2-way operation (a.c. or external d.c. 12 volt). Dimensions are 5.13x19x17.75. Weight is 27.81 lbs.
Price:
$1,899.00

MXP-61VU is an audio mixer with 12 mic/line inputs and 4 line outputs. The compact unit features high-cut and low-cut filters, a.c./d.c. operation, and built-in 1 kHz test tone for precise level setting.
Price:
$10,675.00

SOUNDCRAFT ELECTRONICS

Series 200B is suitable for live sound, recording, broadcast and post-production. Available in 8, 16, 24 or 32 input frames with a 19-inch rack mount 8-channel version. Routing facilities include direct assign to both subgroups and stereo buses. The 4 auxiliary sends are selectable pre or post for both fader and equalization. Choice of 3 input modules with sweep equalization and stereo options. The output section comprises 4 subgroups, the stereo mix and 8-track monitoring/effects returns.

Series 500 is suitable for live mixing. Available in 16, 24 or 32 inputs with space for a further 4 on each. Stereo line input modules may also be fitted when stereo sources are required. The output section provides 8 discrete subgroups and 8 independently equalized effects returns/monitor channels. 4-band equalization with 2 sweepable mid bands, 6 auxiliary sends, insert send and returns, –10 dBv/ +4 dBu operating levels, 8 subgroups, 8 effects returns, stereo input option.
Price:
$8,950.00 (16 channels)
$11,850.00 (24 channels)
$14,850.00 (32 channels)
$17,500.00 (40 channels)

Series 600 features 8 bus, 16 track monitoring (expandable up to 24 or 32 track) 16, 24 or 32 input models available, sweep equalization, 6 auxiliary sends, peak or VU LED metering, patchbay option.
Price:
$9,450.00 (16 channel)
$12,250.00 (24 channel)
$13,350.00 (16 channel with patchbay)
$15,950.00 (24 channel with patchbay)
$19,250.00 (32 channel with patchbay)

TS12 console is suitable for recording, live sound, broadcast or post production purposes. Features include in-line design, 8 separate subgroups, 4-band parametric equalization, versatile routing matrix system, 6 stereo returns, optional mono effects returns module. Also featured is a separate 12 group output section which can also be used to create stereo audio subgroups into the mix bus. Complete MIDI automation is available for the TS12 as well.

Price:
$29,950.00 to $46,250.00 (depending upon configurations and automation options)

Series 6000 is suited for recording, live sound, broadcast and post-production applications. The preamplifier for each input module accepts 68 dB of continuously variable gain and a low noise floor. Capable of up to 24 buses and can be expanded to 32 track monitoring. 4-band semi-parametric equalization and a phase reverse switch on individual input modules. Each of 6 discrete auxiliaries send selectable pre or post fader with additional pre or post equalization settings. MIDI automation is available.
Price:
$14,250.00 to $31,250.00

200 B/VE is an 8 input version of the series 200B, most suited to work closely with a video editor. Auxiliary sends normally follow the fading action of VCA. Auxiliaries 1 and 2 may be switched pre-equalization and pre-VCA. Either standard or sweep equalization input modules may be specified.
Price:
$4,370.00 to $7,645.00 (16 channel with SEQ)

SAC 200 features crosstalk isolation of over 100 dB, a range of mono and stereo modules, Telco system which generates mix-minus feeds within each module, master stereo and monitor modules are fitted as standard with subgroups being optional, meter options include PPM, VU and plasma displays, choice of frame sizes. Ideally suited for all types of broadcast applications.
Price:
$5,150.00 (16 channel)
$5,750.00 (24 channel)

Series 8000 is suited for live sound applications. Selectable features include VCA subgrouping, LED input metering, 3 way panning. Other options include stereo input modules, an 8x8 matrix with parametric equalization. Also featured are 4-band parametric on each input channel, 8 bus design. Available in 24, 32 and 40-channel sizes.
Price:
$24,675.00 (24 channel)
$29,350.00 (32 channel)
$34,125.00 (40 channel)

SOUNDTRACS PLC/SAMSON TECHNOLOGIES CORPORATION

Eric is a mixing console with up to 48 inputs for multi-track recording, video post-production and film facilities. Automation features include digital routing, muting on all inputs, auxiliaries, monitors and groups, 32 external event controllers, and optional linear and VCA fader automation. Other features include 8 auxiliary sends, solo in place, soft muting, 5-band parametric equalization, and Mosses and Mitchell patchbay. Dimensions are 48x42x144.
Price:
$100,000.00 to $140,000.00

CP6800 is a recording console with up to 44 inputs. Automation features include digital routing, muting on inputs, monitors and groups, and 8 external event controllers. Standard features include 6 auxiliaries sends, solo in place, 4-band equalization and patchbay. Dimensions are 45.6x38.4x102.
Price:
$53,000.00 to $60,000.00

CM4400 is a recording/production console with up to 44 inputs. Features include digital routing with manual patch storage and recall, solo in place, 6 auxiliary sends, 4-band equalization. Options include CMS-2 automation system and patchbay. Dimensions are 13.2x33.6x86.4.
Price:
$19,300.00 to $35,000.00

PC MIDI is a 24x16x2 mixing console used for recording, keyboard workshops, and live sound. Features include dual line inputs on each channel which may be used simultaneously, split equalization on each channel dividing equalization between input and monitor, soft muting, 16 subgroups. MIDI features include 100 muting patch combinations, external mode for outboard communication. Options include 2 effects return modules adding 8 line inputs. Dimensions are 9.6x34.8x45.6.
Price:
$9,300.00 to $12,000.00

M series mixing consoles are designed for 8-track recording and live sound reinforcement. Features include 6 auxiliaries, 4-band equalization, 4 matrix sends, direct output on each line in. Options include up to 32 inputs, transformer balanced group and master outputs. Dimensions are 9.6x32x51.6.
Price:
$10,335.00 to $14,000.00

MRX series is designed for multi-track recording and is available in up to 3x8x2 configurations. Features include 6 auxiliaries, 4-band equalization, insert points, direct outputs on each channel. Options include patchbay. Dimensions are 9.6x32.4x51.6.
Price:
$11,000.00 to 15,000.00

FME/FMX/FM series comprises a group of modular mixing consoles designed for customized applications for multi-track recording, broadcasting, and live reinforcement. This series offers input modules for a variety of needs including mono, mono with remote, stereo including RIAA, stereo with remote, monitor with 8 sends. FME frame size fits 22 or 30 modules. FMX/FM frame size fits 14 modules (rackmountable).
Price:
$3,995.00 to $10,425.00

IL3632 and IL4832 consoles are available with 48 channels (104 inputs in remix) or 36 channels (80 inputs in remix), both with 32 buses and a mix noise with 32 channels assigned of better than −82 dB on the masters and groups, inter-channel line crosstalk at 10 kHz better than −85 dB, and inter-monitor crosstalk at 10 kHz better than −85 dB. Features include dual line inputs in addition to a microphone input on each channel, a 4-band fully parametric equalization that may be assigned or split between either the monitor or the channel, 8 auxiliary sends (6 mono and a stereo are provided on each channel), monitoring that may be either PFL or in place solo, and the monitor fader may be assigned as a subgroup fader. A comprehensive TT jack patchbay is standard equipment. The distortion figures at 1 kHz are better than 0.007 percent and at 10 kHz are better than 0.001 percent.
Price:
Available upon request

SOUND WORKSHOP PROFESSIONAL AUDIO PRODUCTS, INC.
Series 34C is a recording/mixing console with broadcast applications. Versions include 12 to 56+ inputs in a variety of mainframe sizes. Arms II/Diskmix and Diskmix moving fader automation available.
Price:
$33,900.00 to $75,000.00

STUDER/REVOX
Series 900 mixing consoles are designed for TV broadcast and production/multi-track recording. They have in/out channels up to 52 inputs, 24 bus available, 4-band equalization, high and low pass filters, fully modular, digital-ready with long work-life, full studio muting logic, mix-down capabilities, and monitor mixers. Flexible with many custom features including Massenberg automation.
Price:
$50,000.00 to $250,000.00

961/962 professional mixing consoles are designed for field recording, broadcast production, and sound reinforcement. Available in 8 to 16 inputs; 2, 3 or 4 outputs,with 3-band equalization, compressor/limiter. This is a fully modular, digital-ready console, other features include fader-start logic, expandable full monitoring features, configured as either portable or studio version. Dimensions are 19x20 (961), 26x20 (962). Weights are 55 and 75 lbs. respectively.
Price:
$14,000.00 to $23,000.00

963 is a full-featured broadcast and recording console with up to 52 inputs, 8 groups and 4 master outputs. Features include 3-band equalization, compressor/limiter, compact and modular design, mixdown capabilities, full-level balanced outputs from groups as well as masters, balanced inserts, full patch standard, complete studio muting signal logic. Dimensions are 15x33x60.
Price:
$45,000.00 to $90,000.00

970 on-air/disk jockey console is specially configured for radio broadcast and production. 12 to 22 inputs are available with stereo master out, stereo out, optional equalization input selectors, control logic with on/off/audition switches that can be re-configured to meet individual needs. Compressor/limiter is standard, as are remote control logic for CD player, cart machine, reel-to-reel and turntables can be configured as an on-air console and/or production room control, work-surface on center module. Dimensions are 37x50.
Price:
$27,000.00 to $42,000.00

ReVox C279 compact mixer has 6 balanced inputs, each mic or line selectable with line input channel selectable stereo (unbalanced) or mono (balanced), stereo balanced and unbalanced outputs. Other features include built-in phantom power, peak reading LED bargraph, monitor speaker output, built-in phase meter, auxiliary send and receive. Options are dbx Type II noise reduction, test generator, fader start logic. Dimensions are 5.25x18x14. Weight is 24 lbs.
Price:
$2,799.00

TASCAM —See our ad on Cover IV
M600 series has 24 or 32-channel input configurations , 16 program buses, 8 auxiliary sends, dual effect returns, 4-band equalization with shelving and sweepable peaks, 16 or 32 monitor returns, insert points on all inputs, fader reverse feature, PFL/solo monitoring, talkback/slate with mic, 4 frequency test oscillator, optional stereo input modules, automation ready and a separate power supply with voltage and thermal indicators on the consoles master section.
Price:
$10,750.00 (M-600/16/16)

$11,250.00 (M-600/8M8S/16)
$13,750.00 (M-600/32/16, 32/32)
$14,250.00 (M-600/24M8S/16, 24M8S/32)

M-520 is equipped with the necessary features for all production modes-record, overdub, remix, or assembly. This is a multi purpose audio mixing console, designed to function as multiple, independent primary mixing system. It comes equipped with 8 balanced outputs, allowing the console to be used as either balanced or unbalanced, or both simultaneously. Dimensions are 9.44x42.94x31.44. Weight is 103.63 lbs.
Price:
$6,999.00

M-308 is designed for recording and sound reinforcement. Features include 5 submix systems (main mix, 2 auxiliary, effects, monitor), 8 channels, phantom power (48V), 3-band equalization, 4 program group/buses, stereo and mono master section, solo system with PFL and AFL capability, insert points on each channel, complete talkback system, transformerless balanced differential input design, 8 tape return jacks, trim and pad controls, 1.5 watt/channel stereo headphone amp.
Price:
$2,299.00
$4,599.00 (20 channel)

M–200 series features main, stereo, foldback, effects and solo sub-systems for recording or sound reinforcement. Features include 8 or 16 tape input jacks, 4 program buses each with master fader and pan control with a choice of XLR and RCA outputs, trim and pad controls to accommodate signals from –70 to +4 dBv (mic in) or from –50 to +24 dBv (line in), 3-band equalization (shelving and sweepable peaks), switchable VU meters with peak level indicators, LED overload indicators, 1.5 watt/channel stereo headphone amp and completely modular construction.
Price:
$1,199.00 (8 channel)
$1,849.00 (16 channel)
$2,599.00 (24 channel)

M-106 is a modular mixer designed for music recording, video post-production, audio sweetening, live sound reinforcement or DJ mixing. Features include a ±15Vdc power supply, 6 channels with mic and line connectors, 4 phono inputs, trim and pad controls, 2-band shelving type equalization, 5 submix systems (main mix, monitor, auxiliary, effects send and return), 6-bus operation provided by 4 program bus outputs and master controls for the auxiliary and effects systems, monitor select switches, 1.5 watt/channel stereo headphone amp, LED overload indicators on each input channel, 19-inch rack mount option.
Price:
$699.00

UREI ELECTRONICS

1650 series is designed with 5 mixers, 2 inputs per mixer, signal to noise ratio program and audition is better than 70 dB below +4 dbm output with –50 dbm, 15.7 kHz noise bandwidth, 3 independent muting buses; each mixer position may be independently assigned to any or all of the buses; each bus may drive 1 of the 3 mute relays.
Price:
$3,046.00 (1651)
$3,796.00 (1652)
$4,096.00 (1653)

1680 series is designed with 8 mixers, 2 inputs per mixer. Signal to noise ratio program and audition is better than 70 dB below +4 dbm output with –50 dbm mic input. Equivalent input noise is better than –124 dbm, 15.7 kHz bandwidth, 3 independent muting buses, each mixer position may be independently assigned to any or all of the buses, each bus may drive 1 of the 3 mute relays.
Price:
$3,796.00 (1681)
$5,046.00 (1682)
$5,446.00 (1683)

1690 series is designed with 12 mixers, 2 inputs per mixer, signal to noise ratio program and audition is better than 70 dB below +4 dbm output with 50 dbm mic input, equivalent input noise is better than –124 dbm, 15.7 kHz noise bandwidth, 3 independent muting buses, each mixer position may drive 1 of the 3 mute relays.
Price:
$6,846.00 (1691)
$7,146.00 (1692)
$7,346.00 (1693)

WHEATSTONE CORPORATION

A-20 on-air console is a 10-input console employing module construction. Available modules include mono/mic, stereo line, control room, studio and full function machine. Other features included standard are program and audition meters, digital timer, and remote starts and external input controls.
Price:
from $10,000.00

Recording Techniques

BRUCE BARTLETT

A COMPARISON BETWEEN TAPE RECORDING AND SEQUENCER RECORDING

• The tape-less studio (synthesizer/sequencer studio) is becoming more popular every day. If you haven't learned how to use one yet, it's time. A basic sequencer studio can be purchased for as little as $650, assuming you already have a home computer system.

This new kind of recording system uses one or more synthesizers and a drum machine linked by MIDI cables to a personal computer running a sequencer program. With this program, the computer records the performance done on the synthesizer keyboard, and plays it back through the synthesizer exactly as performed—or with specified changes.

Sequencing has much in common with tape recording. Since many readers of this magazine are familiar with tape-recording procedures, making the transition to sequencer recording isn't too hard. This article compares the two styles of recording.

First, here's a brief review of how sequencer recording works (see *Figure 1*):

1. You play notes on a synthesizer keyboard.

2. MIDI data about the keys you pressed passes through a MIDI cable into a MIDI computer interface plugged into your computer.

3. The interface converts MIDI data to computer language.

4. The computer memory stores an indication of which keys were pressed, their duration (note on/off), voice (timbre), and sometimes key velocity and pitch-wheel data.

5. During playback of this recording, the data is read out of computer memory.

6. The data is converted to a MIDI signal by the MIDI interface plugged into the computer.

7. The MIDI signal passes through the MIDI cable to your synthesizer.

8. The synthesizer sound generators (or samples) are made to play the same notes you played during recording.

Figure 2 shows a more complicated MIDI studio. It adds a drum machine, other keyboards, a sound generator, and a MIDI thru box. It works as follows:

The clock output of the drum machine drives the computer-sequencer, which in turn controls the keyboards and a multi-voice sound generator through a MIDI thru box. This box feeds MIDI signals to all the keyboards simultaneously. This arrangement is better than a daisy-chain connection, which slows the transfer of MIDI data.

If the various synthesizers and the drum machine require different synchronizing signals, they can be accommodated by a synchronization adapter, such as the Garfield Electronics Dr. Click 2. It can be driven from steady or varying click tracks, drum-mic signals, MIDI clocks, and other sync sources. The unit provides a variety of synchronization, trigger, and click-track signals.

The master keyboard controls all the others. It is plugged into the MIDI computer interface and the MIDI thru box.

After all the tracks are recorded, the computer-sequencer plays the multitrack recording, which activates all the keyboards and the sound generator. The outputs of the keyboards, sound generator, and drum machine are mixed through a mixer (not shown), and the mix is monitored with speakers or headphones. The mix also can be recorded with a stereo tape deck.

The remainder of this article compares tape recording to sequencer recording. That is, we'll compare a tape studio to a tape-less studio. We'll cover each procedure in a recording session, step-by-step from start to finish, contrasting the two systems as we go.

STEP 1. COLLECT EQUIPMENT

Tape studio: You need a studio, control room, multi-track tape recorder, 2-track tape recorder, noise-reduction system, tape, console, effects devices, monitor speakers, power amplifier, cue headphones, microphones, mic stands, audio cables, and materials for editing, cleaning, and degaussing.

Tape-less studio: You need a personal computer system, sequencing software, blank disks for data storage, MIDI-equipped synthesizer, MIDI computer interface, MIDI cables, 2-track tape recorder and tape, monitor speakers, power amplifier, and audio cables. Effects devices and a mixing console are optional.

An alternative to a computer and sequencer program is a stand-alone sequencer, or a sequencer built into a synthesizer.

Such a system can be quite affordable. If you already have a home computer and a stereo system, you can assemble a basic tape-less studio (without vocal recording) for as little as $650. That price includes $300 for a synthesizer (discount price), $150 for a MIDI computer interface, and $200 for sequencing software to run on the computer. The sound quality of this system is excellent.

STEP 2. SET UP MUSICAL INSTRUMENTS

Tape studio: Place instruments and gobos in the studio. Mike the instruments or use direct boxes. Adjust equalization and vary mic choice and placement for the desired sound.

Tape-less studio: Adjust the timbre of each voice with the sound controls on the synthesizer, or use pre-programmed voices. Select the voices to be used in the composition. There's no concern about microphones, mic placement, leakage, isolation, or room acoustics. On the other hand, you're stuck with the sounds your keyboard can play.

ADDRESSES

Allen & Heath Brenell
Five Connair Rd.
Orange, CT 06477

Altec Lansing Corporation
10500 W. Reno
Oklahoma City, OK 73128

Amek Systems and Controls Ltd.
10815 Burbank Blvd.
N. Hollywood, CA 91601

Audio Logic
See DOD Electronics

Audio-Technica US, Inc.
1221 Commerce Dr.
Stow, OH 44224

Auditronics, Inc.
3750 Old Getwell Rd.
Memphis, TN 38118

Biamp Systems
14270 NW Science Park
Portland, OR 97229

Calrec by AMS
PO Box 31864
Seattle, WA 98103

Carvin Corporation
1155 Industrial Ave.
Escondido, CA 92025

DOD Electronics
5639 South Riley Ln.
Salt Lake City, UT 84107

Electro-Voice
600 Cecil St.
Buchanan, MI 49107

Fostex Corporation
15431 Blackburn Ave.
Norwalk, CA 90650

Furman Sound
30 Rich St.
Greenbrae, CA 94904

Gotham Audio Corporation
1790 Broadway
New York, NY 10019-1412

Hill Audio Inc.
5002 N. Royal Atlanta Dr. #B
Tucker, GA 30084

HM Electronics, Inc.
6675 Mesa Ridge Rd.
San Diego, CA 92121

Innovative Electronic Designs Inc.
9701 Taylorsville Rd.
Louisville, KY 40224

Klark-Teknik (DDA)
30 S Banfi Plaza
Farmingdale, NY 11735

Mitsubishi Pro Audio Group
225 Parkside Dr.
San Fernando, CA 91340

Neve
Berkshire Industrial Park
Bethel, CT 06801

Peavey Electronics
711 A St.
Meridian, MS 39301

Ramsa-Panasonic
6550 Katella Ave.
Cypress, CA 90630

Shure Brothers Inc.
222 Hartrey
Evanston, IL 60204

Solid State Logic
Begbroke
Oxford, England OX5 1RU

Sony Professional Audio
1600 Queen Anne Rd.
Teaneck, NJ 07666

Soundcraft Electronics
8500 Balboa Blvd.
Northridge, CA 91329

Soundtracs/Samson Technologies
485-19 S. Broadway
Hicksville, NY 11801

Sound Workshop Pro Audio Prod.
50 Werman Ct.
Plainview, NY 11803

Studer Revox
1425 Elm Hill Pike
Nashville, TN 37210

Tascam
7733 Telegraph Rd.
Montebello, CA 90640

UREI Electronics
see Soundcraft Electronics

Wheatstone Corporation
6720 VIP Parkway
Syracuse, NY 13211

Yamaha Corporation
6600 Orangethorpe Ave.
Buena Park, CA 90620

M916 is a 16 input mixer with selectable mic or line in. 3-band, 9 frequency equalization with post insert jacks. Other features include 11 mix buses, 2 program, 2 foldback, 2 echo, 4 mix matrix and 1 cue, 5x4 mix matrix, cue switch on each input for preview/solo via headphones, headphone feed is stereo in PGM and mono in echo/foldback, 5 illuminated VU meters with peak overload, switchable to indicate stereo program, matrix, foldback, echo and cue levels. Main input/output are transformer-isolated XLRs, switchable phantom power on each mic input.
Price:
$6,595.00

MC1204/1604/2404 are 12, 16, 24 input consoles. Features include 4 program mix buses, 2 effects buses, 2 foldback buses, and a cue bus. Each input has a pad, gain control and peak LED for precise gain matching. It has 4-band equalization, with the 2 mid-bands featuring quasi-parametric control, foldback 1 and 2, and echo 1 and 2 strappable pre/post equalization. Channel On (mute) is post cue for proper cue monitoring. The unit also features complete talkback system, illuminated VU meters (each with peak LEDs), and phantom power is available for mic inputs.
Price:
$2,695.00 (MC1204)
$3,295.00 (MC1604)
$4,395.00 (MC2024)
$4,395.00 (MC2404M stage monitor)

MC802/1202/1602 are 8, 12 and 16 input stereo consoles with 3-band input equalization with sweepable midrange, 3 auxiliary sends per channel, 2 stereo auxiliary returns, input and master headphone cue system, talkback microphone input (assignable to all outputs). The unit also features electronically balanced XLR, 1/4-inch inputs, metering for all outputs, and switchable phantom power on each mic input.
Price:
$845.00 (MC802)
$1,145.00 (MC1202)
$1,345.00 (MC1602)

DMP7 digital mixing processor features all digital mixing and signal processing, 3 on-board DSPs (digital signal processors), digital 3-band parametric equalization on each channel, preset memories (32 internal, 67 external via supplied RAM cartridge), motorized multi-function faders, digital stereo output compressor, MIDI control of preset changes and parameter manipulations, 4 bar-graph meters and LED parameter read-out, digital cascade input/output (ties multiple units together).
Price:
$4,225.00

M406 professional sound mixer is 6-channel, with 3-band equalization, 6-position input level controls, high gain (84 dB) for full output, stereo program output with left and right master controls, echo/effects send bus with master send control, 2 effects inputs (each with level and pan control), dual illuminated VU meters with peak indicators, and right VU meter and headphone output switchable to monitor program or echo output. The unit is rack mountable, with front panel power switch, and security cover available.
Price:
$1,375.00

MV802 8x2 rack mount mixer has 8 inputs, stereo outputs, 2 auxiliary buses, 2 stereo auxiliary returns, balanced and unbalanced outputs, and VCA master volume controls (with optional foot controller).
Price:
$495.00

EM1400/1600/1800 monaural powered mixers have 4, 6 and 8 inputs (each with built-in 150 watts at 4 ohms/100 watts at 8 ohms power amplifier), master 6-band graphic equalizer, 3-band input channel equalizers, 2 auxiliary sends per channel with built-in reverb on auxiliary send 2, electronically balanced XLR and 1/4-inch inputs, complete master section patching in and out, 20 dB pad and variable gain control on each input, and illuminated VU meter.
Price:
$595.00 (EM1400)
$795.00 (EM1600)
$975.00 (EM1800)

EMX200 professional mixer has 8 inputs, stereo power amplifier (250 + 250 watts at 4 ohms/170 + 170 watts at 8 ohms), 8 input channels (each with 3-band equalization, panning, 2 foldback and 1 echo/reverb send), dual 9-band graphic equalizers, electronically balanced low-Z XLR and 1/4-inch input, switchable pad and gain control on each input, pre-fader, pre-equalization sends plus auxiliary returns to foldback buses, built-in analog delay line (200 ms) for echo/reverb and effects, complete patching facilities on rear panel, and illuminated VU meters. Also available as EMX300, with 12 inputs.
Price:
$2,195.00 (EMX200)
$2,545.00 (EMX300)

TV-500 MTS master control console includes 4 stereo subgroup buses, 2 stereo master buses, a mono bus for SAP and mono sum outputs. Also included are 4 stereo auxiliary buses, comprehensive control room and multiple studio communication and muting systems. The unit is available with VCA subgroup functions and external VCA control ports for editor control. Mainframes are available in 16 to 56 input configurations.
Price:
from $45,000.00

SP-6 radio production console features stereo and mono input modules with machine control and remote on/off capabilities, control room and studio muting and tally system. The unit is available in 4 or 8 track configurations and can mix down to stereo and composite mono, and also features 4 auxiliary send buses, equalization, clocks, timers, and tape recorder controls. Available in 4-56 input configurations.
Price:
from $24,000.00

SP-5 stereo production console has mono and stereo inputs, stereo subgroups, multi-track routing and matrix submixing. Master stereo and mono outputs, and 4 auxiliary send buses are provided. Additional features include controls for clocks, timers, tape machines and user specified frame widths.
Price:
from $17,000.00

MTX-1080 sound reinforcement consoles have 32-52 mic/line inputs with LED level indicators, 18 outputs with VU meter level indicators, 8 effects sends/electronically balanced inputs and outputs, 4-band variable equalization, 8 programmable meter presets, modular construction and external power supply.
Price:
from $32,000.00

MTX-88 sound reinforcement consoles have 32-52 mic/line inputs with LED level indicators, 10 outputs with VU meter level indicators, 8 effects sends/electronically balanced inputs and outputs, 3-band variable equalization, modular construction and external power supply.
Price:
from $27,000.00

MTX-40 sound reinforcement console has 24 mic/line inputs with LED level indicators, 7 outputs with VU meter level indicators, 4 effects sends, electronically balanced inputs and outputs, 3-band variable equalization on all inputs, modular construction and external power supply.
Price:
$11,350.00

M-16 sound reinforcement console is available in a 16 to 52 mic line input configuration with LED level indicators, 17 outputs with VU meter level indicators, 16 monitor sends, 4-band variable equalization on all inputs, 4 programmable mute presets, modular construction and external power supply.
Price:
from $16,000.00

YAMAHA CORPORATION —See our ad on page 11

PM3000-24/32/40C comes in 24, 32 or 40 input configurations, 8 group buses, 8 auxiliary buses and separate stereo bus. VCA assignable grouping with 8 submasters with automation interface. 8 bus muting master system with safety override. XLR inputs are differentially balanced with 34 dB trim and 5 position pad for optimizing gain structure. 4-band parametric equalization with variable high pass filters on each input plus 2-band equalization on the 4 stereo auxiliary returns. 11x8 mix matrix. Insert point selectable in/out on each input. Extensive cue and solo system. Comprehensive talkback system with full intercom capability. Phantom power individually selectable on each mic input.
Price:
$33,500.00 (24 channel)
$38,500.00 (32 channel)
$44,500.00 (40C)

PM1800-16/24/32/40C are configured as 16, 24, 32 or 40 channels. Features include 8 mixing buses, 4-band sweepable equalization with variable high pass filters, 6 mute groups, 8x4 mix matrix, 4 stereo auxiliary returns.
Price:
$13,900.00 (16 channel)
$16,900.00 (24 channel)
$19,900.00 (32 channel)
$23,900.00 (40C)

M508/512 is an 8 or 12 input console with 3-band equalization on each input, 4 mixing buses, selectable input sensitivity with overload LED, VU meters with peak indicators, 3 effects returns and 4 sub inputs, stereo headphone feed in PGM and mono in echo/foldback, talkback mic input assignable to all outputs, transformer isolated XLR inputs and outputs, 0.25-inch input/output jacks for effects and sub inputs, phantom power on each mic input.
Price:
$2,045.00 (M508)
$2,825.00 (M512)

STEP 3. SET LEVELS

Tape studio: Set master faders; set trim and input fader for each instrument; set up subgroups; set monitor mix; set cue mix.

Tape-less studio: There's no level setting, unless you want to specify a different volume (key-velocity scaling) for each track.

STEP 4. RECORD

Tape studio: Set the tracks you want to record to "record-ready" mode. Hit the record button. Watch levels as the musicians play.

Tape-less studio: Hit the record key(s) on your computer keyboard. This key is specified in the sequencer-software instructions. Play a musical part on your synthesizer, either in real-time or in step-time.

STEP 5. REWIND TO THE BEGINNING

Tape studio: Enable the return-to-zero function. Hit the rewind button. Wait several seconds.

Tape-less studio: Hit the "to the top" key on your computer keyboard. You're there in an instant.

STEP 6. PLAY THE RECORDING

Tape studio: Hit the play button. The recorded tape tracks play through the monitor mixer into the power amplifier and monitor speakers.

Tape-less studio: Hit the "play" key on your computer (specified in the sequencer instructions). The computer data plays the synthesizer, which you monitor with a power amp and monitor speakers.

STEP 7. PUNCH IN/OUT TO CORRECT MISTAKES

Tape studio: Start the track running in play mode several bars before the part you want to correct. Next, manually punch in the record button just before the correction. Then record the corrected musical part. Finally, punch out of record mode just after the correction.

Tape-less studio: Use the above procedure for manual punches. Alternatively, you can use AUTOPUNCH. With this feature, the computer automatically punches in and out at pre-set times; all you have to do is play the corrected musical part. Perform an autopunch as follows:

1. Using the computer keyboard, set the punch-out point (the measure, beat, and pulse where you want to go out of record mode).

2 Set the punch-in point (just before the part you want to correct).

3. Set the cue point (where you want the track to start playing before the punch).

4. Hit the "play" key on your computer.

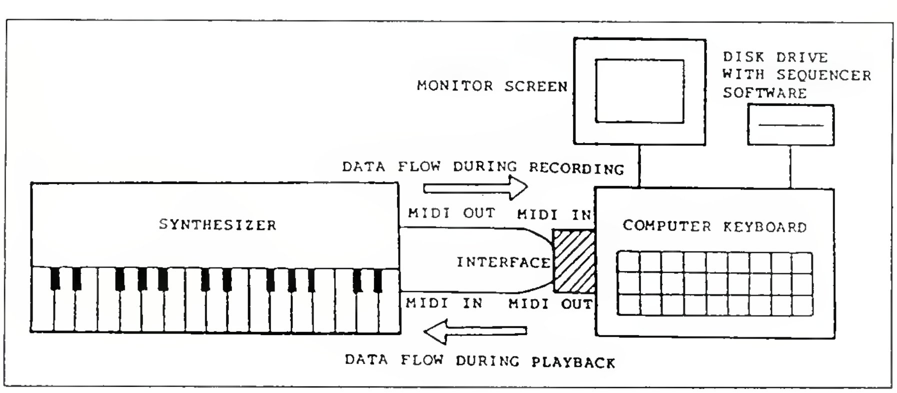
5. When the screen indicates punch-in mode, play the corrected part.

STEP 8. DO OVERDUBS

Tape studio: First, set the tracks you want to record to "record-ready" mode, and set the other tracks to "safe" mode. Then set up the musical instrument and its microphone. Get a level. Next, play the tape while setting a cue mix and monitor mix. Rewind the tape. Finally, hit the record button. Have the musician play a new part while listening to the recorded tracks.

Tape-less studio: First, use the computer keyboard to select the track you want to record. On your synthesizer, select the voice (sound patch) you want to use. Then hit the "record" key(s) on your computer keyboard. Play the new part on the synthesizer while listening to the recorded tracks. (These tracks are also played by the synthesizer.)

Figure 1. A basic MIDI sequencer studio(tape-less studio).



STEP 9. BOUNCE TRACKS

Bouncing tracks is the process of copying two or more tracks onto another track. Then the original tracks can be erased, freeing them up for recording more instruments.

Tape studio: The following is an example of bouncing tracks. First, assign tracks 1, 2, and 3 to track 4. Next, play the multi-track tape and set a mix for tracks 1, 2, and 3. Then rewind the tape. Finally, record the mix of tracks 1, 2, and 3 onto track 4.

Tape-less studio: Hit the "bounce" key(s) on your computer keyboard. Type in the source track and destination track (indicate which track you want to bounce to). In a few seconds, the bounce is accomplished.

With a tape studio, you can bounce two or more tracks at a time. But with a tape-less studio, you can bounce only one track at a time. However, you can bounce a track into a pre-recorded track without erasing the pre-recorded track. The pre-recorded track and bounced track will mix together.

With the tape studio, there is a generation loss (loss of sound quality) each time you bounce. This does not occur with the tape-less studio.

STEP 10. MIXDOWN

Tape studio: To begin, monitor the 2-track stereo mix bus. Next, play the multi-track tape. Adjust the level, EQ, panning, and effects for each track. Then rewind the multi-track tape and repeat the above procedure to perfect the mix. When your mix settings are ready, play the multi-track tape and record the mix onto a 2-track tape deck.

Tape-less studio: Connect the synthesizer audio output to a 2-track tape deck. If your synthesizer allows, adjust the volume (key-velocity scaling) of each track by hitting the appropriate computer keys. Record the program changes (voice changes) on a separate track. When all the tracks are recorded, hit the "play" key on your computer keyboard and record the mix off the synthesizer audio output.

If your song data plays several keyboards and a drum machine, plug all their audio outputs into a mixer. Set up a mix with panning and effects, and record the mix on a 2-track tape deck.

Note: Although you can record effects on a multi-track tape recorder, you can't do that with a sequencer; you can add effects only during mixdown. That's because effects are audio signals, which sequencers can't record.

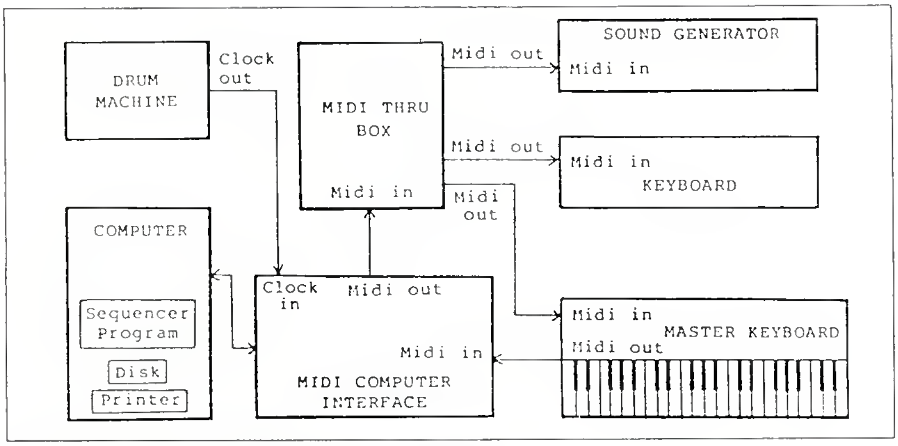


Figure 2. A more complicated MIDI studio.

This can be a limitation. If you play a sequencer recording through a multi-timbral synthesizer, and add an effect to that synth, the effect will be on all the voices which that synth plays. However, if each track plays through a different synth, you can add a different effect (or different amounts of effect) to each synth.

There's a way to have different effects on different voices in a multi-timbral synthesizer. If the synth is a sampling keyboard, each sample could have reverberation or some other effect already on it; in which case each sample can have a different effect. The effect is not recorded in the sequencer; rather, the effect is part of the sampled sound. Note that the sampled reverberation will cut off every time you play a new note. This sounds unnatural but can be used for a special effect.

When you're doing a mixdown, it can be hard to remember all the fader changes. But there is help. A MIDI-equipped mixer permits computer-controlled (automated) mixdowns via a MIDI signal between the mixer and computer. The computer remembers your fader changes and resets the faders automatically as the song plays. Two such automation systems are the MegaMix system by Musically Intelligent Devices, and the MidiMation system by J.L. Cooper Electronics. You can even buy an automated mixer in a single package, such as the MixMate by J.L. Cooper Electronics.

STEP 11. EDIT

Tape studio: Obtain splicing tape, a grease pencil, an editing block, and leader tape. As an example, splice section A of Take 3 onto section B of Take 6. Add leader tape between selections on the 2-track master tape (if you're using an open-reel recorder).

Tape-less studio: Using the computer keyboard, append section B onto section A. Add leader tape between selections on the 2-track master tape.

With a sequencer program, you can rearrange song sections by pressing a few keys on the computer. You also can have any section replayed wherever desired in the song. Thus, you might build up a song by having the computer play sections A, A, B, A, B, A in that order. To do this with a tape studio, the musicians must play all the sections in order. Or you must copy each section and splice in the copies where desired.

MAJOR CONCERNS WITH EACH SYSTEM

Tape studio: Audio quality is a primary concern. You must calibrate and clean the tape recorders, use noise reduction, and set levels properly to obtain a good signal-to-noise ratio, wide flat frequency response, and low distortion.

Tape-less studio: Audio quality is hardly a problem. The recording plays back through the instrument you recorded on, (or another one) with perfect fidelity. The main concerns are to:

1. Fluently get around the various menus in the program.

2. Understand the program and hardware well enough to figure out, and correct, confusing results. For example, "Why is track 3 playing a flute sound when I asked for a harpsichord?" "Why is that note stuck on? (you lost a note-off message). "Why didn't the punch-in record what I played?"

It helps to read the instruction manuals thoroughly, and simplify them into step-by-step procedures for various operations. If you have questions, call the technical service people at the manufacturers of your equipment—there may be errors or omissions in the instructions.

FEATURES FOUND ONLY IN THE TAPE-LESS STUDIO

Sequencers (or sequencer programs) permit you to do things that are next-to-impossible in tape-based studios. For example, you can change the key of a musical passage without changing its tempo. Or you can change the tempo without changing the pitch. You can record the music at a very slow, easy-to-play tempo; then play it back at a faster rate. All you have to do is type the appropriate keys to specify the tempo and key of a particular sequence of notes.

It's also possible to record voice changes (timbre or program changes) on a sequencer track that is separate from a music track. While a music track is playing, you press the buttons on your synth for the desired voice changes. These changes are recorded on a track of the sequencer.

For example, to change from a flute to bells partway through a melody, simply press the flute button on your synth at the desired point in the song. Program changes can be punched in, entered in step mode, or edited just as notes can.

If you're playing a multi-track sequence through a multi-timbral synthesizer, you can change the voice of any track after recording by pressing the appropriate buttons on the synth. For example, if you don't like your music performed by a synthesized trumpet, you can have it performed by a synthesized organ.

Another advantage of sequencer recording is automatic punch in/out. It can be much more precise and controllable than a tape punch in/out. There's no worry about punching out too late and accidentally erasing part of a track.

As mentioned before, a sequencer program also permits step recording. You enter notes one-at-a-time at your own pace. When you're done, you hit the "play" key on the computer keyboard, and the notes play back at the tempo you specified.

You might specify the duration of each note by holding down the appropriate piano-style key, and hitting [RETURN] on the computer keyboard once for each 1/8th-note duration. For example, an A# half note is four 1/8th notes long, so you'd hold down the A# key and hit [RETURN] four times.

Then play and hold the next note in the song, hit [RETURN] as many times as required, and continue until the entire song is entered.

COMBINING TAPE RECORDING WITH SEQUENCER RECORDING

With a tape-less studio, you're limited to the voices you can achieve on your synthesizer(s). If you want to add a vocal or other microphone signal, you must record it onto a multi-track tape recorder or a hard-disk recorder.

In other words, you often need a system that combines multi-track tape recording with multi-track sequencer recording. The two systems—tape and MIDI—can be synchronized with a special sync tone that you record on one track of the multi-track tape recorder. The sync tone is controlled by the sequencer software, and is taken from the sync connector (if any) on the MIDI/computer interface.

This system allows you to record many more tracks. You might have just four tape tracks available, but you can synchronize them with up to sixteen virtual tracks recorded with a sequencer. When you play the tape, the sync tone forces the tape tracks and virtual tracks to play together in synchronization.

To do this, you might proceed as follows:

1. Record a sync tone on one track of the recorder (usually track 4 of a recorder/mixer with a sync input). Record the tone at a level that produces correct synchronization (usually around -4 VU). If you can switch off the noise reduction for the sync track, do so.

2. Set up the musician and microphone to do an overdub on multi-track tape.

3. While the tape is rolling to record the overdub, the sync track activates the synthesizer sequence at the proper time.

4. The vocalist listens to the live synth sequence and sings along, while being recorded on an open track. That's how the overdub is done.

5. During mixdown, the sync track keeps the recorded voice track and live synth signals in synchronization. There's no need to record the synth signals onto multi-track tape—these signals just play "live" along with the recorded vocal track.

If you need more parts in your composition, you can record the synth parts onto tape, then overdub more synth parts. The sync track will keep them all synchronized.

A disadvantage of this system is that you must rewind the tape to the beginning for each overdub—you can't start in the middle of the song. To overcome this problem, you need a sync-to-tape converter with song position pointer. Then, tape tracks and virtual tracks can be synched anywhere in the song. This works only if your sequencer and synth implement song position pointer.

SUMMARY

As we've seen, both tape and tape-less studios have advantages and disadvantages. Since tape-less studios are becoming popular, it's to your advantage to understand both systems of recording. It's also great fun to learn the new technology and do things you never could do before. db

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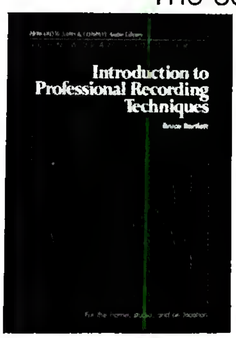

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THE ELECTRONIC COTTAGE

Cantrax Recorders

RICHARD CANNATA

If you start dreaming correctly when you are a child, you may well have a studio of your own when you grow up.

From the young age of nine, I have been fascinated by tape machines. This fascination has led to the creation of the facility I now own and operate.

In addition to the recording equipment, Cantrax Recorders offers a full range of musical instruments. Guitars, basses, drums, drum machines, and synths are available for the asking. I play all of the aforementioned instruments, and in fact, I have composed quite a bit of material for some of the industrial and commercial clients that have booked the room.

The type of services offered are basically all pertaining to audio recording. The clients are generally corporate accounts, radio stations, schools, public service agencies (announcements), and musicians. Recent jobs include jingles, audio post video, voiceovers, demos, slide-show sound tracks etc.

A NEW LOCATION

I started this business initially by doing remote recording with a ReVox A700 and a Teac 3440. I would go to various locations throughout Southern California. Eventually this type of recording became too tedious and time consuming. At approximately the same time, my wife Nancy and I were preparing to purchase a home.

I remember as we were looking at different homes, the first thing that would look for would be an ideal area within the house for a production-type studio. This did not come easily. I think we searched for over a year! Being very particular finally paid off. We found a home with an add-on that had an outside entrance. The room turned out to be 15 x 20 feet.

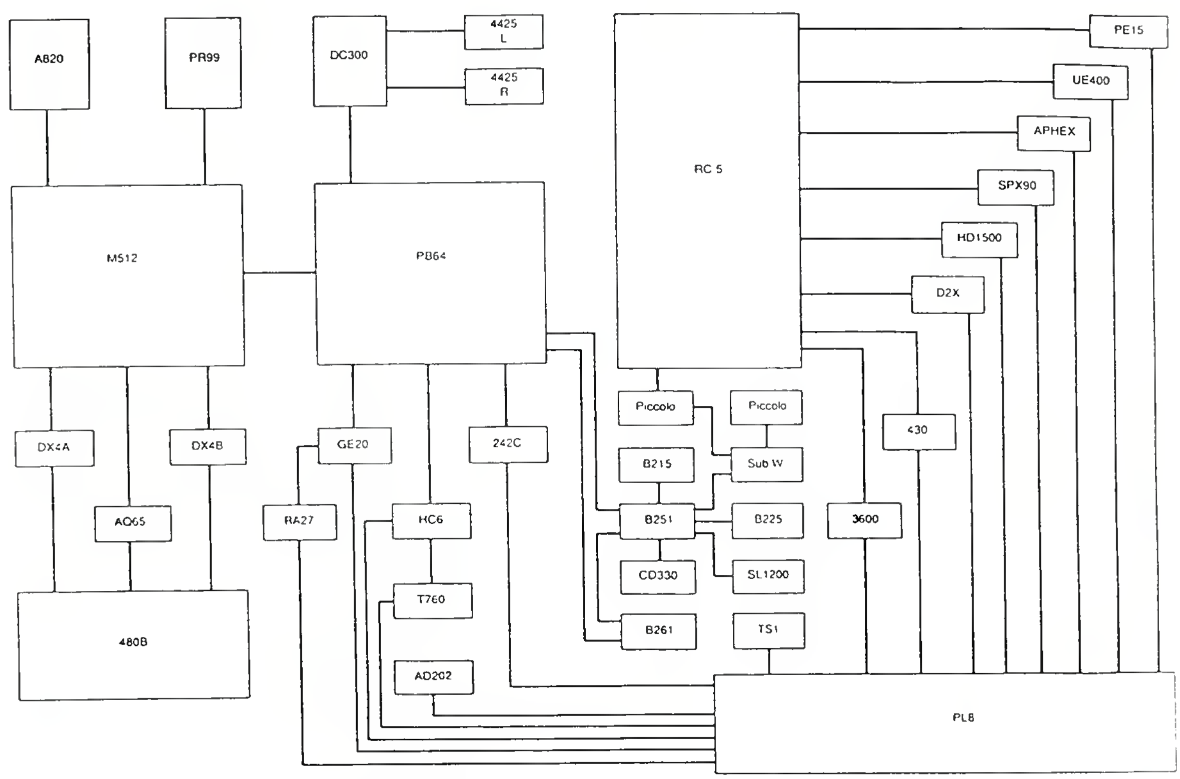


Figure 1. Overall simplified wiring diagram of the studio.

Cantrax Recorders, located in Long Beach, California, is a full production facility headed by Richard Cannata.

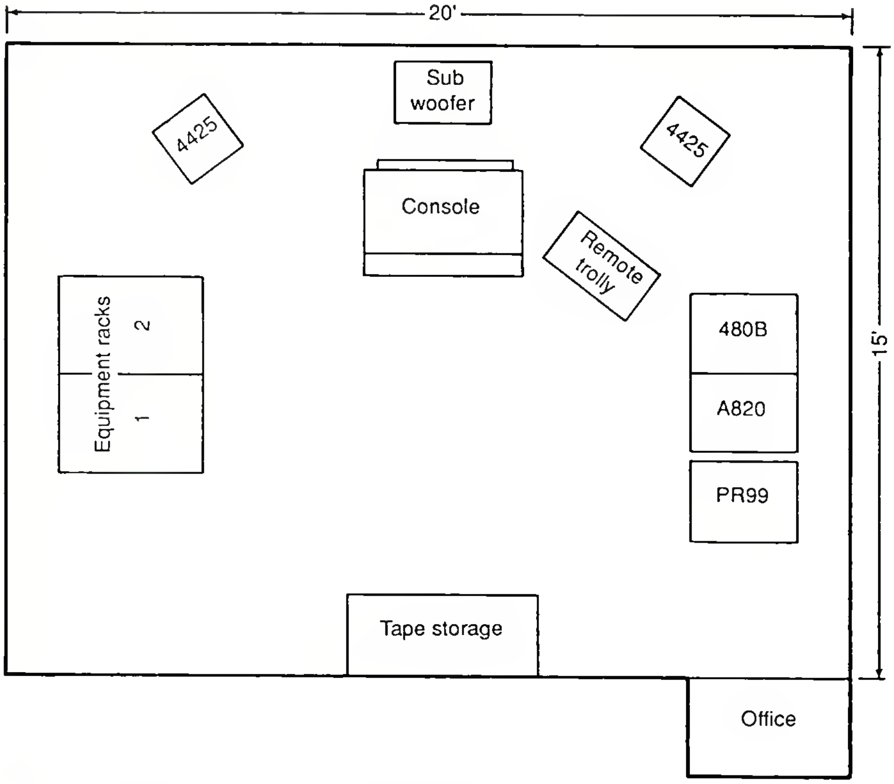


Figure 2. The present room equipment layout.

The first thing we did was to scrape off the old "cottage cheese" type material that covered the ceiling and replace it with an acoustically-treated ceiling. The whole process took several days and a lot of hard work. Upon completion of the ceiling, my wife and I stripped and re-finished the floor to its natural state, which is solid oak planks.

I drew up rough sketches of where the equipment was to be placed. The guidelines for equipment placement necessitated that all equipment be within an arm's length. If this was not possible, the units were either remote controlled or put in 19-inch roll-around equipment racks. After I decided on the final layout, equipment was purchased and placed in the designated areas.

Wiring the studio was a real nightmare. Ground loops, hum, buzzes etc. took a lot of trouble-shooting to find the problem. The layout of the room incorporates two patch bays, hence many feet of cable. The cable that we chose is the highest grade, from Belden. Before we were done, we had spent five days and used over 1000 feet of cable to complete the job. All cables are identified by tags and logged. In addition to this, I drew up a schematic drawing of the entire installation (*Figure 1*). The AC is completely filtered and grounded to eliminate any hums or buzzes.

Our equipment is purchased through local area recording and pro sound dealers. All repair work is done in-house by myself. I also maintain full service documentation and spare parts for all the studio equipment.

BUDGET WELL SPENT

Equipment choice was decided upon sonic excellence, durability, serviceability and company support. The budget was originally estimated at $50,000 but ended up somewhere around $73,000 (probably something to do with those few extras I deemed necessary). $10,000 of the budget was spent on our Studer A820 which we feel is money "well spent" and a sound investment (this machine is a dream to edit and master on, and the sound is excellent).

Once the equipment had been purchased and installed, we purchased a real time analyzer and Sonex acoustical foam. We did numerous tests moving the foam to different locations on the walls to obtain the flattest results and to eliminate any standing waves. After approximately four hours we reached satisfactory results, confirmed by the real time analyzer.

Monitoring is accomplished with JBL's Bi Radial System. Also, we incorporate the ReVox Piccolo System, which is a satellite subwoofer system. Our clients always have very positive comments about the sound of the room, and the sound that we capture on tape.

Advertising is generally done through trade magazines and local music stores. Fifty percent of our clients are return business, which is to me a sound indicator that we are doing something right. We are also affiliated with "Music City Songwriters Festivals Inc." We offer the people that they send to us a 10 to 15 percent discount. They are usually young songwriters starting out and this is their first foray into the recording environment. Once in a while we see a seasoned veteran come in through M.C.S.F. Inc.

As for the future, we are going to move into a new and larger room which will have a separate studio area. The present house layout maintains a large garage which will be the shell for the new facility. The dimensions for the control room will be 20 x 20 feet and the studio portion will be 35 x 35 feet. Plans are to incorporate a floating hardwood floor and a 20-foot high-angled ceiling. The walls will be made of double construction lined with an acoustic treatment. The control room window will be 1/2-inch double pane glass set in an angle. No stone shall be left unturned.

After we complete the new room expansion, we will be going to 16-track. I've got an eye on Studer for their unveiling of a multi-track line. This would make us a total Studer ReVox facility.

TIME TO EXPAND

The expansion was originally scheduled for the first quarter of 1988, but has been postponed until June. Construction of the walls and floor should take about three full weeks with another week to ten days for installation of the wiring and patch bays. Transfer of the equipment from the old room to the new room should take about two or three days. Finally, the tuning of the room and studio with the real time analyzer should be completed in a day. The way that we have things planned, we shouldn't be down more than three or four days. Just long enough to transfer the equipment and tune the rooms. I'm very anxious to get in there and do what we do best: recording!

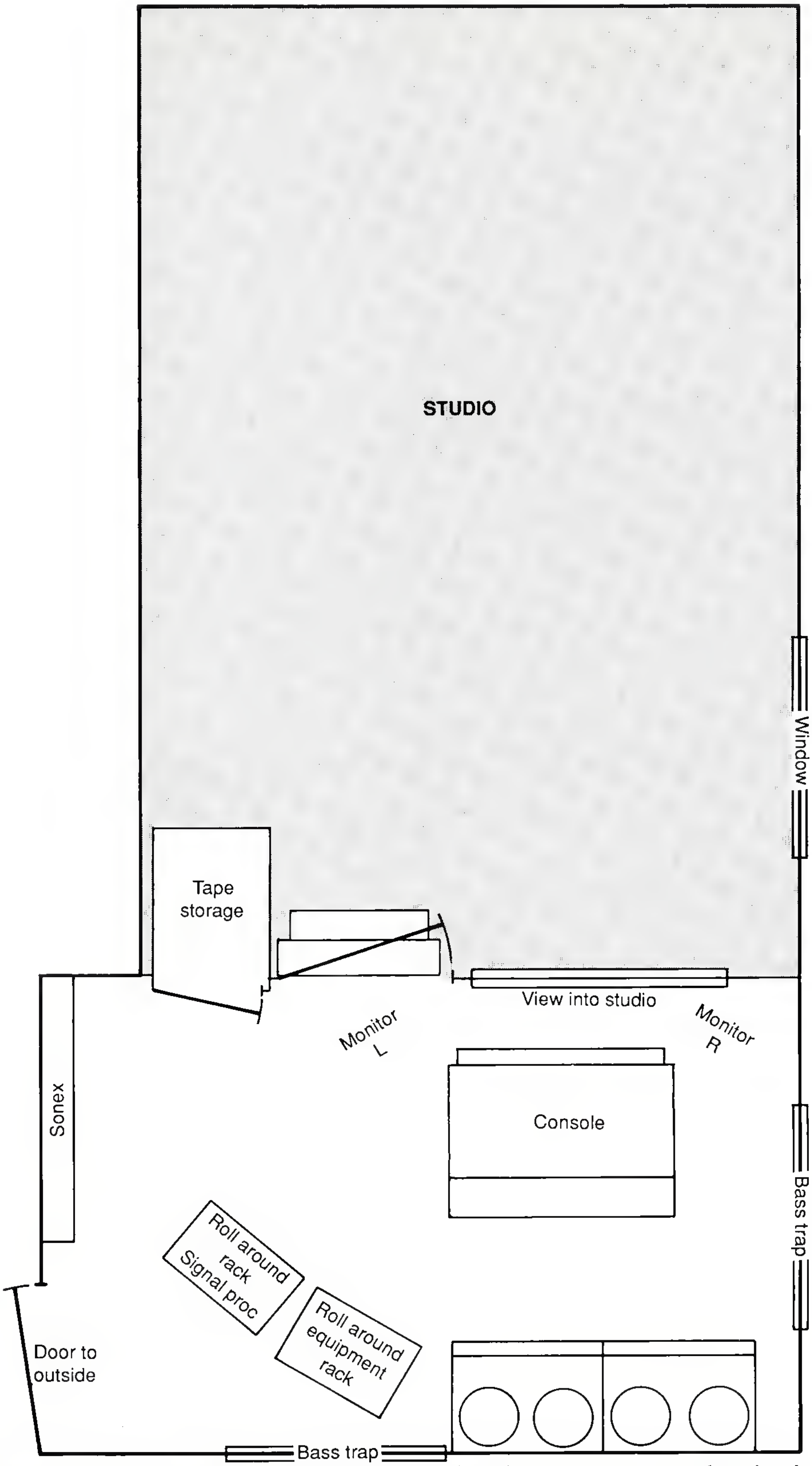


Figure 3. This is what Cantrax will look like when the move to our new location is done.

In the meantime, Cantrax Recorders will be keeping abreast of new developments which, if they apply, will be incorporated into the new rooms. We are hoping to take on more musicians with the separate control room/studio layout. We have dreams of pleasure pursuing the recording world. db

EQUIPMENT LIST

Tape machines:
Studer A820
ReVox PR99II, B215 cassette
Tascam 480B
Superscope CD330 field recorder
Console:
Tascam M512, 12 x 8 x 2
Monitors:
JBL 4425
ReVox Piccolo & subwoofer system
Amps:
Crown DC300A, series 2
ReVox B251
Noise reduction:
8 channels of dbx Type 1
Dolby B & C
Dynafex
Reverb and delays:
Yamaha SPX9011 (X2)
Digitech RDS3600
Sound workshop 242C reverb
Ibanez AD202 analog delay, HD1500 harmonics/delay
Other effects:
Valley People Dynamite (X2)
Aphex Type C
Rane PE15 parametric eq, HC-6 headphone console, RA27 analyzer
Ibanez UE400 multi-effects
Teac GE20 graphic eq
Furman PL+8 line filter
Other equipment:
ReVox 261 digital tuner, 225 CD
Yamaha T760 tuner
Loft TS-1 test set
Technics SL1200 MKII turntable
Teac PB64 patch bay
Tascam AQ65 autolocator
Microphones:
Electro-Voice PL20, 635, DS35
Sennheiser 421, S
Shure SM 57, 58
Headphones:
AKG K240DF
Sennheiser 414 (X3)
Musical instruments:
Fender Strat, amp
Guild F44 acoustic
Hofner Beatle bass
Roland TR505 rhythm composer
Yamaha DX100 synth
Full set of maple Ludwigs with Zildjian cymbals

Ad Ventures

BRIAN BATTLES

• This issue's installment is not for you crusty old-timers with the razor blade scars on your fingertips and permanent grease pencil stains under your nails. If you're a grizzled veteran of long nights in a dimly-lit studio filled with overflowing ashtrays, cups of congealed coffee, and ankle-deep in shards of tape, you've probably learned most of the following tricks of the trade on your own. These are the techniques not found in the attractively-bound instruction manuals packaged inside the cardboard boxes that contain shiny new digital tape decks exuding their musky scent of freshly-soldered circuit boards. You've seen those manuals, hastily translated by Nippon's finest dropouts from English-language curricula. They often contain passages like this:

"1. For obtain maximum output in reproduced mode, thread tape onto left of head block through to takeup reel, depress REPRO button, and adjust trim pot R16 until VU meter read 0 dB of test tone of 1000 Hz at 230 nanoweber flux level, unless the switch S44 setting to –4 dB output. Now to remove six retaining screw and exposing front panel to access reproduce trimmers."

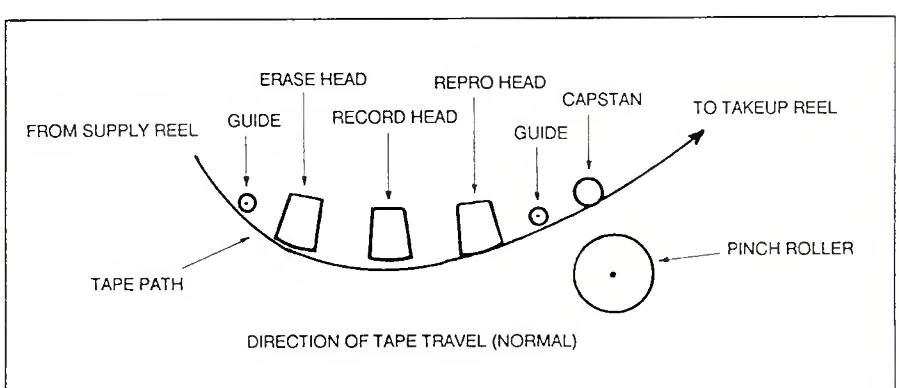
No, those of us who truly grew up with the 47th chromosome—also known as the Recording Gene—found out all the dirty, non-factory-approved methods of Getting the Job Done without concern for warranties and recommended operating procedures. We pioneered "sound-on-sound" by jamming a piece of cardboard between the tape and the erase head, then recording over a previously taped sound. (Remember when all tape machines with more than four tracks were the exclusive property of places like Electric Ladyland and Deutsche Grammophon studios?) We created 15-second repeats with 9-foot loops of tape strung through the capstan and wending its way around wastebaskets, alcohol bottles, mic stands, and back to the heads. We fabricated the sound of Martian voices by wrapping capstans with uneven strips of masking tape; we flipped tape end over end and dubbed voices from one deck to another to make bizarre "pre-echo" and "preverb." Nowadays, these artifices are easily preformed by investing a few thousand dollars in a rackful of nifty processors and specialty devices. This is not a column for you if you've already "invented" techniques like these while hunched over the knobs in a grubby Jorma Kaukonen tour T-shirt. This is for all the kiddies who have got into the business recently; you who have never owned a tape deck without full-logic braking and solenoid feather-touch controls.

For some special effects, you may want to run some tape through a machine in reverse.

We "old-timers" can vividly recall the days of quadraphonic stereo (with joystick balance controls), 8-track cartridge recorders, Burwen noise reduction and the first cassette decks. Remember the prehistoric days of those weird levers and bars and high-tech "piano key" push-button? Remember accidentally stretching two feet of a master tape into stiff Mylar twine by accidentally shifting from rewind to play without waiting for the reels to stop turning? Ah, the good old days, when the only thing digital in the studio was an early Hewlett-Packard scientific pocket calculator.

I don't mean to imply that every pimply-faced punk who gets into recording today just naturally lays out the wampum for a tape-less digital studio and hires away the chief engineer of a major research lab to handle studio maintenance, but it is much easier to learn recording using equipment that performs functions we only fantasized about in the days when transistors were the latest electronic innovation and every serious audiophile had his trusty Webcor open reel deck. Believe it or not, there are still a few unfortunate souls who have yet to take delivery on a Lexicon or a New England Digital Direct-to-Disk system, and it's for you poor slobs that I list a few of the handy tips and slapdash fixes used by hoary

Figure 1A. Normal tape travel on a recorder.



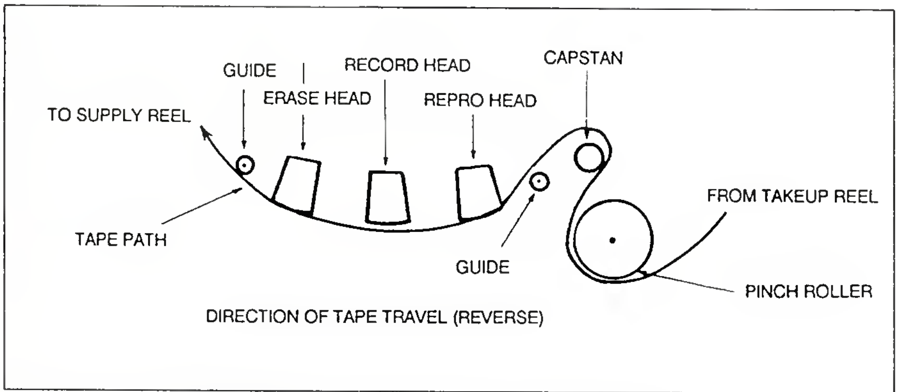


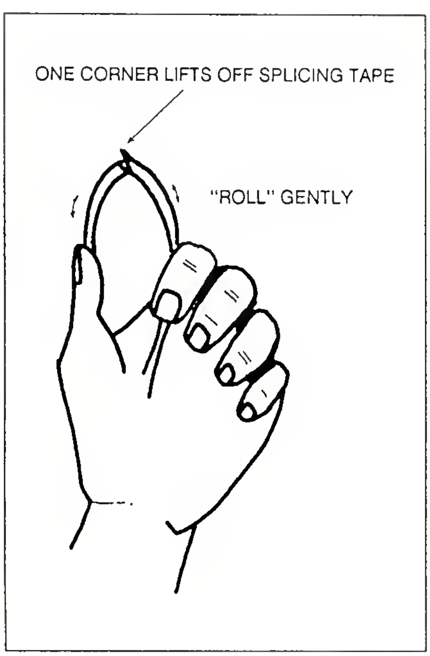
Figure 1B. This technique reverses the direction of tape travel

experts from audio recording's antediluvian epoch (roughly pre-1984):

1) For some special effects, you may want to run some tape through a machine in reverse. The obvious way to do this is to pull off both reels, flip them over, and re-thread the tape. A quicker way is to leave the reels where they are and simply wrap the tape around the capstan from the outside and bring it around so it goes through the capstan/pinch roller contact point backwards (*Figure 1A & 1B*).

2) Very full reels can be a real bother; you've probably experienced the hair-pulling frustration of watching a few turns of tape unravel from a nearly overflowing reel of a threaded tape. You might routinely fix this by taking off both &*%$#@!! reels and twisting one over and over to untangle the tape. Yet, if we put our advanced brains to this problem by applying principles of geometry, physics, and a few of Newton's laws, we can discover a brilliant solution. Hmmm...Aha! Here's a much easier way: just grab the tape on the opposite reel and pull an equal number of turns off it as well. Then manually rotate one reel to take up the slack, and *voila*! No more twisted tape.

Figure 2A. Bend the tape at the splice but no more severely than this.



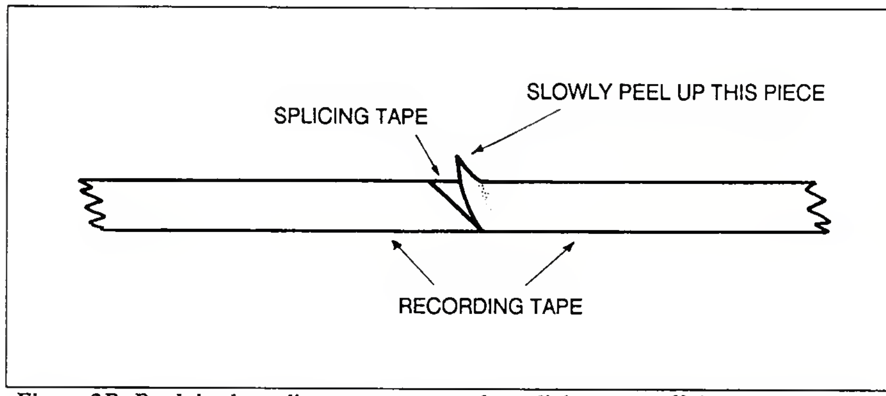


Figure 2B. Back in the splicer, you can get the splicing tape off the tape this way.

3) How many times have you had to redo a splice several times in one location? You could repeatedly cut through the same spot, splicing tape and all, and add another layer of splicing tape on top each time until the edit point gets almost as thick as it is wide. To avoid this situation, try removing the old splicing tape instead. The better way to do this is to bend the tape sharply at the joint and roll the tape between your fingers until one or both corners of the cut edges of recording tape begin to pull away from the splicing tape. This technique works best if your cut is at a 30 or 45 degree angle; it's hard to lift a corner of a 90 degree cut (*Figure 2A & 2B*).

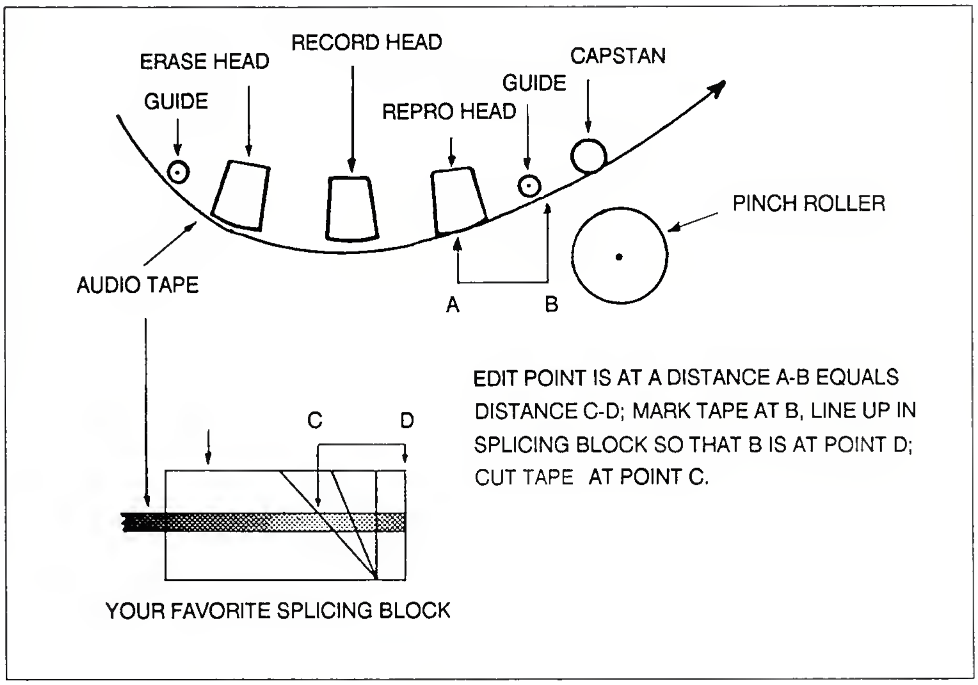
Be careful not to bend the tape sharply enough to fold it to a crease. It's rather like removing the backing from a bumper sticker. Incidentally, if you just peel away the recording tape you're discarding, you can probably reuse the splicing tape edge left hanging from the other end when you make your new splice. By the way, I prefer Scotch brand 41 splicing tape (the white stuff, not the blue), because it sticks nicely, and when you want it to, it comes off much easier than some of the other kinds that I've tried. It also doesn't stretch.

4) To keep wax pencil marks off your tape head surfaces, establish an alternative mark point. Manually rock the tape to your precise edit point, mark the tape at a spot an inch or so away from the head (at a tape guide, for instance), and put the tape in your splicing block with the mark lined up to a point exactly the same distance to the cut as the guide is to the head (*Figure 3*).

A tape dispenser's serrated edge can stretch the stuff, or cause a slight bulge at the edit.

5) Use your razor blade to cut off each piece of splicing tape. A tape dispenser's serrated edge can stretch the stuff, or cause a slight bulge at the edit. Also, carry the cut piece by sticking it lightly to the blade itself. This keeps your oily fingers off the stickum, and makes it easier to apply the splicing tape accurately when you lay it in the groove (*Figure 4*).

Figure 3. Marking the tape so that the splice will go exactly where you want it to be.



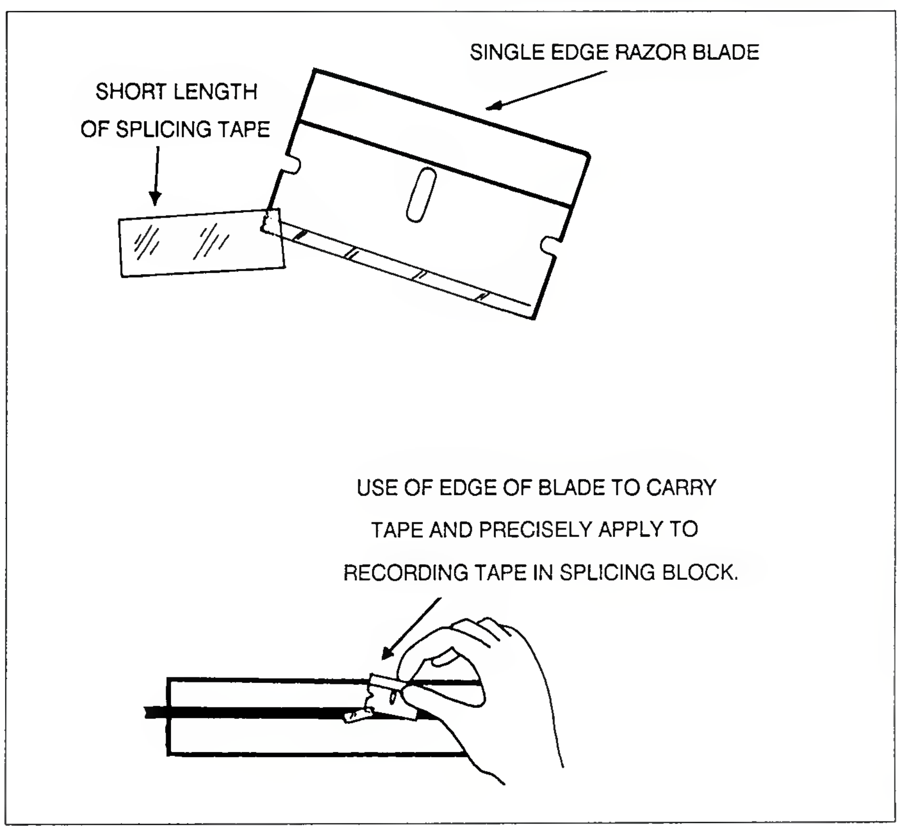
6) If you use tiny scraps of paper stuck between layers of tape on a reel as "bookmarks" to find spots that you want to locate later, scribble a key word or two on the scrap that shows what is there.

7) If you have a project that requires you to assemble a piece from many loose lengths of tape, use a masking tape tab on the end of each piece, and identify each with a key word and the running time of each strip. You may even be able to perform a "tape-less edit" by indexing each scrap to a transcribed script and reading, humming or singing the proposed new arrangement aloud.

Always mark your reels just before you begin a recording. As the room fills with anonymous reels of bits and pieces of a project, you'll lose track of what's what.

8) Never toss away a piece of edited tape until you are absolutely sure your edit works. Drape the cut out piece of tape over your lap or the top of the tape machine until you're thoroughly satisfied that you can safely discard it. It beats fishing through the spaghetti in

Figure 4. Appying splicing tape to the tape as it is held in the block. Note that the razor blade can help carry the tape and that the splicing tape should go onto the tape in a parallel position to the tape's edge.



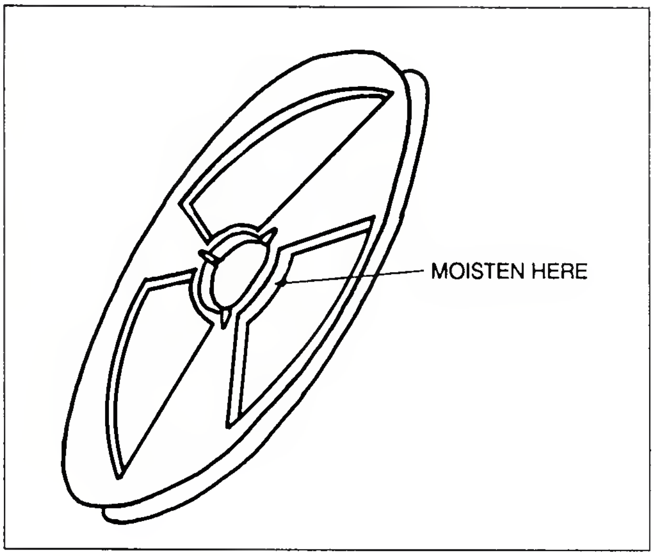


Figure 5. Get the hub slightly moist so it is tacky and grips the tape.

your trash can looking for what you thought was a worthless scrap.

9) Always mark your reels just before you begin a recording. As the room fills with anonymous reels of bits and pieces of a project, you'll lose track of what's what. It's much easier to use masking tape to identify each reel's contents. Stick a 2- or 3-inch strip right on the reel and write the contents on it. Don't write directly on the reel, since you may wind tape on and off that reel during the session, and you can simply peel off the masking tape off one and move it from reel to reel as necessary. Also, the masking tape comes off when you're done, and you can use the reel over again in the future without getting it covered with illegible scribbles.

If you'd like to get in touch, write to me in care of db, or to PO Box 17386, Boulder, CO 80308-7386.

10) Use your head demagnetizer or bulk eraser on your razor blades. This prevents much gnashing of teeth by preventing unaccountable clicks or dropouts from appearing at each of your splice points.

11) When trying to locate a splice in fast-wind mode, hold your finger lightly against the outside of the tape as it zips by between reels. You'll feel the splicing tape go by, and you don't have to keep your eyes riveted to the tape (and risk blinking just as the splice goes through).

12) Thoroughly scrub your hands before an editing session. This keeps skin oils to a minimum and keeps you from getting fingerprints all over the tape. I've cringed while watching someone edit who's just eaten a bag of potato chips.

13) When threading a tape, lay the end flat against the reel hub and let the tension of the tape's subsequent layers provide the pressure to hold it in place as you wind on more turns. If you can't seem to get the end to "catch," and the reel spins wildly without tightening up, hit the stop button and pull it back out. Then lick your fingertip, rub it on the hub surface to moisten it slightly, and the faint tackiness will help the end of the tape grip the hub (*Figure 5*).

TALKBACK MIC

I've been getting a lot of cards and letters from readers who are eagerly awaiting the release of my audiocassette program on *How To Produce Great Radio Commercials* (due out in the fall), and I want to let you know that I appreciate the comments and compliments on my column. A few specific thanks: Anthony Marrapese of Reel to Real Recording Studio in Cranston, Rhode Island...As proof that there is more to Orlando, Florida than just some big amusement park, I received some nice words of praise from Ray Olech of Ray Olech Productions...Greetings to Robert S. Dire of Winetree Productions in Rancho Cucamonga, California. Sheesh, Bob, a *plain* card? Couldn't you find anything with a picture on it?...Hello to Jeffrey P. Hedquist of Hedquist Productions, Inc. in Fairfield, Iowa. I'll consider your proposal...Welcome words of praise come from Rev. David K. Chase of The Community of Jesus, Incorporated in Orleans, Massachusetts... Techies: an indispensable new tool is now available—Allen Schultz, super-technician for ListenUp Audio/Video in Denver, Colorado has created an amazing gizmo in a handy little portable box. It's an incredibly versatile audio signal tracer that has every kind of input and output you can imagine, including XLR, 1/4-inch, RCA, as well as oscilloscope outputs, and it accepts levels from millivolts to a hundred volts. There's a built-in amp and speaker, headphone jack, and it runs on a.c. or batteries. The ideal test tool for field or bench testing. If you'd like to get more details, construction plans, or maybe coerce Allen into hand-building you one of these babies, drop me a note and I'll make sure he gets it. Also, watch for his article in a forthcoming issue of *Syn-Aud-Con Newsletter* (Norman, IN).

If you'd like to get in touch, write to me in care of **db**, or to PO Box 17386, Boulder, CO 80308-7386. Send a letter, a card, or perhaps a demo cassette of your best radio production efforts. I'd love to share your creative techniques with other readers of this column.

DID YOU KNOW?

You, our readers, are very important to us. Should you ever need to contact us about your subscription, please do not hesitate to write to us at:

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203 Commack Rd. Suite 1010
Commack NY 11725.

HOW TO READ THE LABEL

The top line of your label contains information that enables us to find you on the computer. Without that we can't effect address changes or find your entry. The sample below is what most labels look like.

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Sagamore Publishing Co
1120 Old Country Rd
Plainview, NY 11803

The first sequence of characters is how we locate you. The next four numbers indicate the issue you have just received. Finally, the last four numbers represent the date of expiry *reversed*. Just read backwards: 9309 is Sept/Oct 1993.

New Products

PORTABLE R-DAT

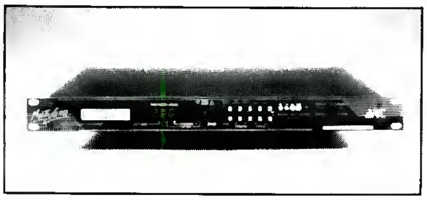
• Ramsa-Panasonic announces the Model SV-250 portable R-DAT recorder which features balanced XL-type input connectors and a 2.2 hour record capacity from its rechargeable NiCd battery pack. Weighing less than 3.2 pounds complete with battery pack, the unit measures 1 5/8-inch by 8 7/8-inch by 5 3/8-inch, and includes peak-level metering, headphone monitoring, switched 14 dB microphone attenuation, and a 60-times high-speed search mode. The recorder can be powered from rechargeable batteries, external d.c. supplies or main power. Record sampling frequency is 16-bit/48 kHz. The unit incorporates dual MASH (Multi-Stage Noise Shaping) analog-to-digital converters and 64-times oversampling digital filtering. As well as dramatically reducing the amount of signal distortion caused by conventional filters used in digital recorders, these latest generation circuit components also lower the amount of zero-cross distortion, thereby producing a clean, clear audio signal at low as well as high recording levels.

Mfr.- Ramsa-Panasonic

Price- $3900.00

Circle 52 on Reader Service Card

MULTI-EFFECTS UNIT



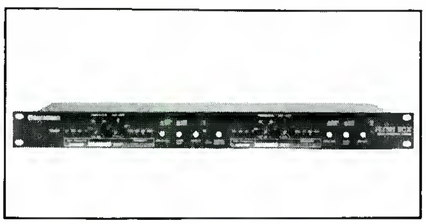
• Applied Research & Technology introduces the MultiVerb, a low-cost, multi-effects unit which allows four simultaneous effects from one single rack. Reverberation, arpeggio effects, reverse gates, pitch shift, doubling, image doubling, digital delay, chorusing, and equalization multi-effects can all be programmed into the unit's 200 memory locations, or alternatively, selected from the 100 on-board presets and subsequently stacked. These multi-effects may then be random accessed at a later date, in groups of four if required. The unit incorporates several features such as battery back-up for full memory protection, remote footswitch jack with preset increment, a level selector, 16 bit digital processing, and full MIDI compatibility. A random access keypad and 32 character LCD display for simple operation are further features. The inclusion of pitch transposition is standard.

Mfr.- Applied Research & Technology

Price- $575.00

Circle 53 on Reader Service Card

NOISE REDUCTION



• Rocktron announces the Hush IICX. It is a full-rack, two-channel, stereo version of the patented Hush II single-ended noise reduction. The Hush IICX offers effective noise reduction up to 50 dB. It also provides full metering for filter bandwidth and expander gain reduction. The visual indication of the bandwidth of the dynamic filter allows the user to see the effectiveness of the filter operation. The expander gain reduction meter gives a visual indication in decibels of the amount of expansion taking place. The unit also includes a stereo master/slave feature. When using the stereo master, the controls on channel one will also simultaneously control the settings for channel two. This two-channel unit also provides use with PA systems, monitor systems, and recording systems. By selecting a slow release in the expander section, the device is optimized for use with composite music.

Mfr.- Rocktron Corporation

Price- $409.00

Circle 54 on Reader Service Card

Classified

STUDIO EQUIPMENT for sale. (MegaMix 24 Track e.c.t.). Please send Name and Address for Info sheet to: **Hughes, 496 LaGuardia Place, Apt. 169, New York City, NY 10012.**

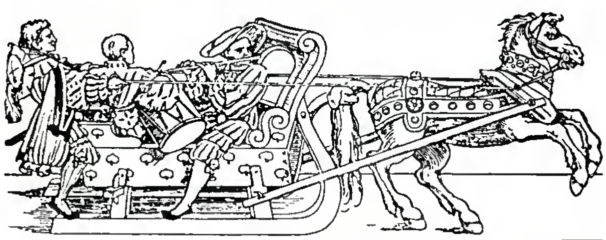
OTARI 5050 TAPE DECK
For sale, one Otari 5050 2-channel tape deck. In perfect operating condition. Meets all original specs. Heads, good. Complete with looseleaf instruction/service manual.Full price: $750.00, FOB New York. **Write Dept. 38, db Magazine, 203 Commack Road, Suite 1010, Commack, NY 11725.**

WANTED: Analogue Vocoder, Neve input modules, Tascam 32 or similar Revox. Contact **John (516) 483-9747.**

MIXING BOARDS
Factory direct, models for home recording and live sound. Quality boards priced to move. Quantities are limited, call while supplies last. **For info and prices call (201) 423-2176. Ask for Dave Fox.**

WANTED: Pultec EQ's. We will pay $1,000 for almost any Pultec program EQ modes EQP1/EQP1A /EQP1A3. Also wanted EQH2/ EQH3/MEQ5/MAVEC/MB1/ITI and Sontec EQ's. Any tube or ribbon microphones and limiting amps. Please call or write to: **Dan Alexander Audio, 2944 San Pablo Ave. Berkeley, CA 94702. (415) 644-2363.**

WE'VE MOVED!



db Magazine

ELAR Publishing

There's a new address:
203 Commack Road, Suite #1010
Commack, NY 11725
The telephone number is now:

516-586-6530.

Closing date is the first of the second month preceding the date of issue.

Rates are $1.00 per word with a $25.00 minimum. Boxed ads are $40.00 per column inch. db Box Numbers are $8.50 additional for wording "Department XX" plus $1.50 additional for postage and handling.

Quantity discounts are: 3X – 15%; 6X – 30%.

ALL CLASSIFIEDS MUST BE PREPAID

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People, Places... & Happenings

• **Synergetic Audio Concepts** has moved their Bedford office to their farm. Now their office, research lab and home are on the same acreage. The new address is: R.R. #1, Box 267, Norman, IN 47264, (812) 995-8212.

• **Zero Corporation** announced the acquisition of **Anvil Cases, Inc.** Anvil is a producer of high-quality carrying and transit cases and will become a part of the Zero Halliburton Division of Zero Corporation. Zero Corporation is engaged in the design and manufacture of enclosures and accessories for the electronics industry, including cabinets, cases, cooling equipment and packaging hardware.

• **Otari Corporation** now occupies a new facility which is located at 378 Vintage Park Dr., Foster City, CA 94404. The two-story facility houses combined office and warehouse space. New additions include an acoustically designed listening room by **RLS Acoustics** of San Francisco, and customer training facilities. The architectural firm involved with the project was Leason Pomeroy Associates of Santa Clara.

• The **Summa Music Group**, a music publisher and artist management company, has opened a new state-of-the-art recording facility overlooking Sunset Strip. Studio A's control room monitor system was designed by acoustician George Augspurger; Mogami and Monster Cable wiring is used throughout the facility. Tape machines include Studer analog A-820 and A-800 multi-tracks, as well as a Mitsubishi X-850 PD-format digital 32-track. Mastering is to Ampex ATR-102 two-tracks. Studio A was built as an adjunct to a newly remodeled 24-track MIDI/synthesizer and overdub room, which features a 36-input Demideo-API custom console. All three rooms are laid out in a line, with excellent visual communications.

• **Auditronics, Inc.** announces the appointment of **Murray Shields** to Director of Sales. Prior to this, Mr. Shields was Vice-president of Sales for ADM Technology in Troy, MI, and before that was Vice-president of Eastern Sound in Toronto, Canada. He has an extensive background in the broadcast and production industries.

• **Full Sail Center for Recording Arts**, Altamonte Springs, FL, has recently had two industry professionals, **Bruce Swedien** and **Tony Bongiovi**, as guest teacher/lecturers. Bruce Swedien, engineer for most of Quincy Jones' productions and winner of this year's Grammy for Best Engineer, gave a series of twelve lectures covering various aspects of engineering and recording. Engineer/producer, Tony Bongiovi, who is co-owner of New York's Power Station Recording facility, gave a series of four lectures on The Business of Music, covering such areas as owning and operating a major label studio, how record deals are made, and the job market for the recording engineer.

• **The University of Miami**, Coral Cables, FL, has installed a **Sony** MXP-3036 automated mixing console and two APR-5002 recorders in its recording studio at Gusman Concert Hall. The new equipment was acquired to provide music engineering students the opportunity to handle top-line recording and mixing gear. Another reason for acquiring the MXP-3036 is for use by the university music engineering program which is getting more involved in digital recording and automated sessions. The Sony equipment is used by the university's 100 music engineering students to record recitals at the school as well as their own sessions.

Sony also has announced that it has named **James M. Frische** president of **Digital Audio Disc Corporation**, its Terre Haute, Indiana, compact disc manufacturing subsidiary. Mr. Frische joined DADC in 1983 and supervised design, construction and staffing of the facility which opened in mid-1984. In 1987, Mr. Frische was promoted to executive vice president, adding product development and strategic planning responsibilities.

• **Soundtracs PLC** announces a joint venture with **Samson Technologies Inc.** for the sales and marketing of Soundtracs mixing consoles within the USA, effective June 1, 1988. **Todd Wells** of Soundtracs confirmed that he was delighted with the professionalism and determination shown by the Samson management, not only in the commercial area where the discussions started, but in the technical back-up which **Doug Bryant**, Vice President, is installing in the Hicksville facilities.

• The **Apollo Theater Investor Group** unveiled a $17 million multipurpose audio/video/stage production facility. ATIG Chairman **Percy Sutton** re-opened the world-famous Apollo Theater with assistance from New York State and the City of New York. He remarked that with these new facilities, they have produced the only landmark theater of its kind in the country that can handle production of commercials, films, music videos, and recordings on a competitive, state-of-the-art basis.

What is below, is, with minor revision, based on a recent p.r. release from Sony Corporation, and has been also approved by U.S. representatives of Studer Corporation in Nashville, and TEAC/TASCAM in Montebello, CA. We present it because we believe it to be of industry-wide importance.

• At a press conference held in Tokyo on June 20, 1988, and hosted by TEAC Corporation, the three professional audio companies currently involved in DASH products (Sony Corporation of America, Studer Revox of America and TEAC Corporation of America) gave an update on their plans to support their common format.

The DASH format, originated by Sony Corporation, was proposed for worldwide tape interchange jointly with Willi Studer AG and Matsushita Electric Industries Corporation in 1982, when the first DASH multi-channel recorders were introduced to the market.

Today, the DASH format is supported by the engineering work, product development and marketing of three major professional audio companies: TEAC Corporation and Sony Corporation of Japan, and Willi Studer AG of Switzerland.

TEAC CORPORATION'S ANNOUNCEMENT

TEAC Corporation announced its intention to display its prototype DASH 24-channel recorder before the end of the year. The recorder is entirely based on TEAC Corporation's proprietary development, with the exception of DASH key devices such as heads and LSI which were developed in common by all DASH companies. This announcement confirms TEAC Corporation's intention to extend its product line-up to include the highest market segment, and illustrates TEAC Corporation's confidence in the present importance and growth potential of the DASH format. The TEAC Corporation's DASH 24-channel recorder will support full digital audio interchangeability with the present and future 24-channel recorders of other DASH companies.

SONY INTRODUCES THE PCM-3324A

Sony Corporation announced the introduction of the PCM-3324A, an upgraded version of its widely successful DASH 24-channel recorder PCM-3324. While preserving the external appearance and function of its predecessor, the upgrade product features advanced DASH circuitry, reduced power consumption, and state-of-the-art A/D and D/A converters.

SONY AND STUDER CONFIRM HALF-INCH 48-CHANNEL

Today's DASH multi-channel recorders are based on 24-channels recorded on half-inch tape. Sony Corporation and Willi Studer AG announced the successful completion of all joint engineering work necessary for establishing an upward-compatible DASH 48-channel machine, based on the same half-inch tape width. Both Sony Corporation and Willi Studer AG have confirmed their intention to introduce half-inch 48-channel recorders at the nearest possible date; at least one product announcement will be made before the end of this year.

Forty-eight channel recorders will be introduced by all three DASH companies, and will support full digital audio interchangeability with present and future 24-channel DASH recorders.

DASH 24-CHANNEL AND DASH 48-CHANNEL

All three DASH companies stressed that, while they will introduce DASH 48-channel to cover the top segment of the professional market, DASH 24-channel remains the best solution for a majority of users in terms of flexibility, universality and overall cost.

The introduction by both TEAC Corporation and Sony Corporation of second-generation DASH 24-channel recorders illustrates their present and future support for the 24-channel format. The upward-compatible extension of the DASH format to 48 channels, the highest track number available today, ensures that the large investments of the audio industry in today's 24-channel multi-track equipment will remain fully productive and profit from technological development. As a result of upward compatibility, all 24-channel DASH tapes in use today can be extended to 48 channels on the forthcoming 48-channel recorders, and all 24-channel installations in use today can be considered as part of a forthcoming worldwide 48-channel DASH system with full interchangeability.

ENGINEERING COOPERATION

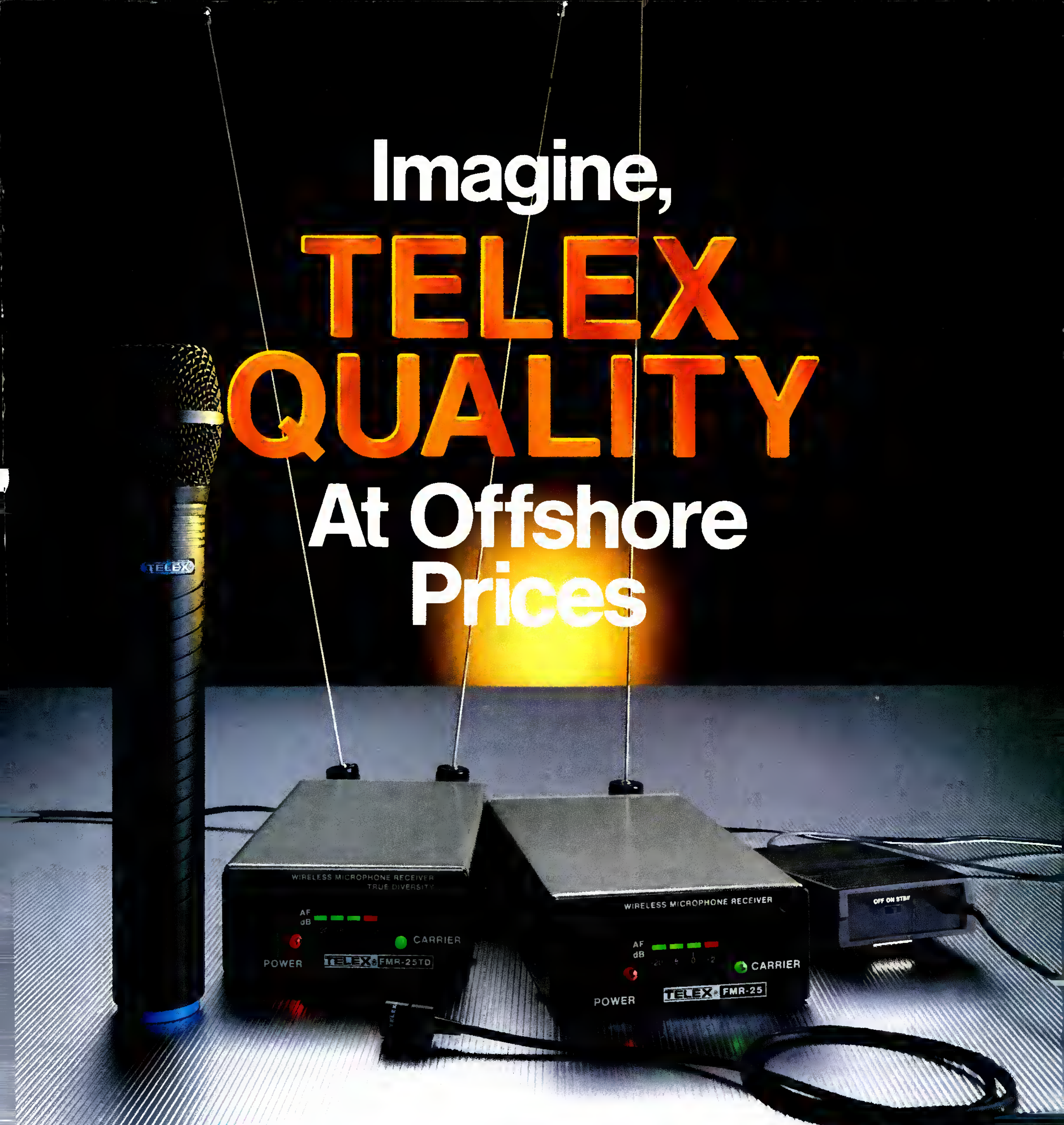
The introduction of second-generation 24-channel DASH recorders, and the technical development of DASH 48-channel, would have been impossible without a very strong management and technical commitment by all three companies.

In order to ensure maximum compatibility among all DASH recorders, the common development and use of key DASH components (heads, signal processing and interfacing LSI) was decided at an early stage, while all other areas of product development remained strictly independent.

As a result, fully compatible recorders with radically different design characters will compete in the marketplace, to the best benefit of professional users. This illustrates the policy of the DASH group to avoid an artificial second source based solely on OEM relationships; it is felt that this kind of defacto monopoly is not to the advantage of the professional audio industry.

DASH'S FUTURE

This year and next year, several new models of second-generation DASH multi-channel recorders guaranteeing full digital audio interchangeability and covering both 24-channel and 48-channel will be introduced. The continuing worldwide dominance of the DASH format is secured. The engineering-level confirmation, by Sony Corporation and Willi Studer AG, of the 48-channel format, and their clear commitment to follow up with products, provide the definitive answer to any controversy on the ideal track number, while the decision by TEAC Corporation to enter the DASH 24-channel market further reinforces the present DASH format. db



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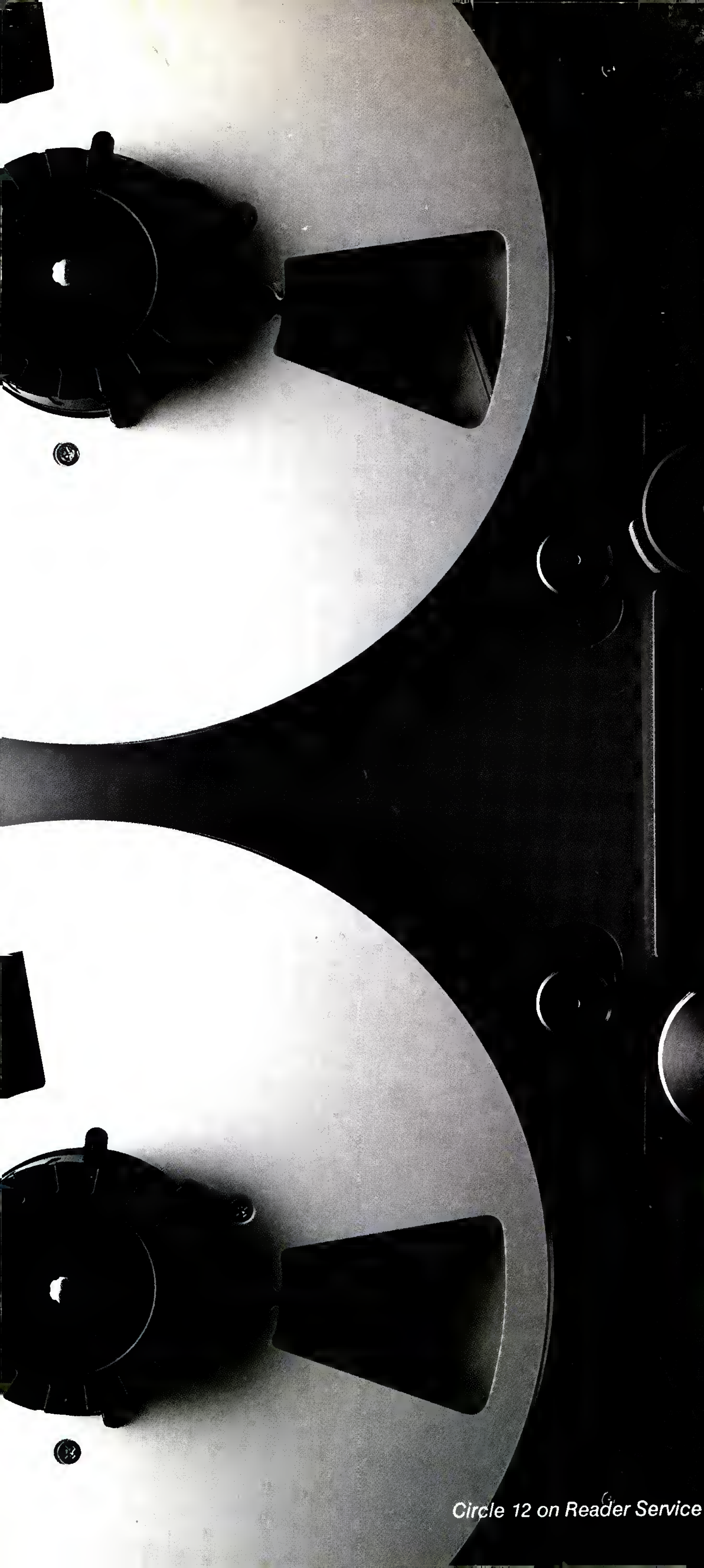

STACKED DECKS.

Production is a high stakes game. When played with skill and speed, its rewards are substantial. And it doesn't hurt if the decks you play with are arranged in your favor.

That's where the ATR-60 Series comes in.

We've designed this series of professional recording consoles to give you the features you need when the chips are down. For instance, proprietary Tascam head technology is so refined that you can make final EQing decisions right in the sync mode, without having to rewind and check the sound from the repro head.

The Omega Drive found on all the ATR-60s virtually eliminates tape stress. With their rock-solid deck plates, flex-induced post-production wow and flutter becomes a thing of the past. Lightning-fast lockup, time code lock and easy top panel source monitoring make the ATR-60s almost magically easy to use.

Sit down at the table and play with the ATR-60/8 half-inch production-quality 8-track; the ATR-60/2T center track time code deck; the ATR-60/4HS half-inch 4-track high speed mastering or multitrack; the ATR-60/2N quarter-inch mastering deck; the ATR-60/2HS half-inch high speed mastering deck; or the ATR-60/16 one-inch 16-track.

The ATR-60 series from Tascam. When you play the production game with an ATR-60, it's almost like cheating.

TASCAM

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