MAY/JUNE 1986
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db

THE SOUND ENGINEERING MAGAZINE

serving: recording, broadcast and sound contracting fields

Featuring: 2 to 8 trk—The Smaller Recording Studio
Guides: Synthesizers and Computer Software

Circle 10 on Reader Service Card

MAY/JUNE 1986 VOLUME 20 NO.3

the recording engineer

2 to 8 the smaller recording studio

columns

ON THE COVER

● Tony Bennett with the Sony PCM-3324 digital multi-track recorder that was used to record his latest album, *The Art of Excellence.*

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Letters

ONE TOO MANY?

To The Editor:

Looks like we have a little problem. Again, I've received two copies of the same issue.

I hope this won't shorten my subscription period by the number of extras.

Victor Sutton
Hinkley, Ca

P.S. I've subscribed to *db* for some number of years now, and watched it gradually change. It keeps getting better. Thanks.

We hope by the time you receive the May/June issue we will have successfully merged the subscriptions for all who have received two issues. If, at this time, anyone is still receiving an extra copy of db *in place of* MR&M *please contact us.*

THREE CHEERS FOR JAN/FEB!

To The Editor:

The January/February 1986 issue of *db* is a gold mine!

The articles on Electro-Acoustics make this issue a text book and I can assure you that it has found a place on the book shelf.

I wish you could publish these articles plus others of the same trend, as well as articles on odd or unusual treatment of acoustical problems in a bound book form.

PLEASE continue to keep your future issues just as good as this one.

Robert C. Green
Milgren Acoustics
Milford, De

We'll certainly try!! Thanks for your encouragement.

db IS THE ONE!

To The Editor:

Howdy! Just a note—*db Magazine* provides such a rich resource of audio information that I can spend thousands of dollars and weeks of study, or I can get *db*, spend a matter of minutes, and have the basics of what I need. And the references listed at the end of articles give me the information should I wish to pursue a more in-depth study of a particular subject.

In a nutshell, *db* summarizes within a few pages what would normally take a week to read. My copies go in the library for future reference! Thanks and keep it up.

C.D. Wood
Kvll-Am
Woodville, Tx

Thanks for such a nice note.

A WEALTH OF INFORMATION

To The Editor:

I am sorry to hear that MR&M *will no longer be published as a semi-professional magazine, but happy to hear that the core of the publication will still be contained in the pages of db Magazine.*

I subscribe to db *and did subscribe to the now no-longer issued* MR&M *and will extend my subscription to* db.

As a sound reinforcement contractor I find a wealth of technical information dealing with the hard core facts of acoustics and how they pertain to the day-to-day life of the professional.

Speaking quite frankly, I will miss receiving *MR&M* since it started my career in audio and reminds me of the heyday of the industry.

Philip Bjorkman
Chief Engineer, Starr Sound Systems

We've gotten so many letters such as these above. It's nice to know that MR&M *was so loved (we loved doing it, too), and it's nice to know that you, the reader, likes what we're doing with* db Magazine. *Please continue to tell us all you feel—both good and bad.*

DISAPPOINTED BUT HOPEFUL

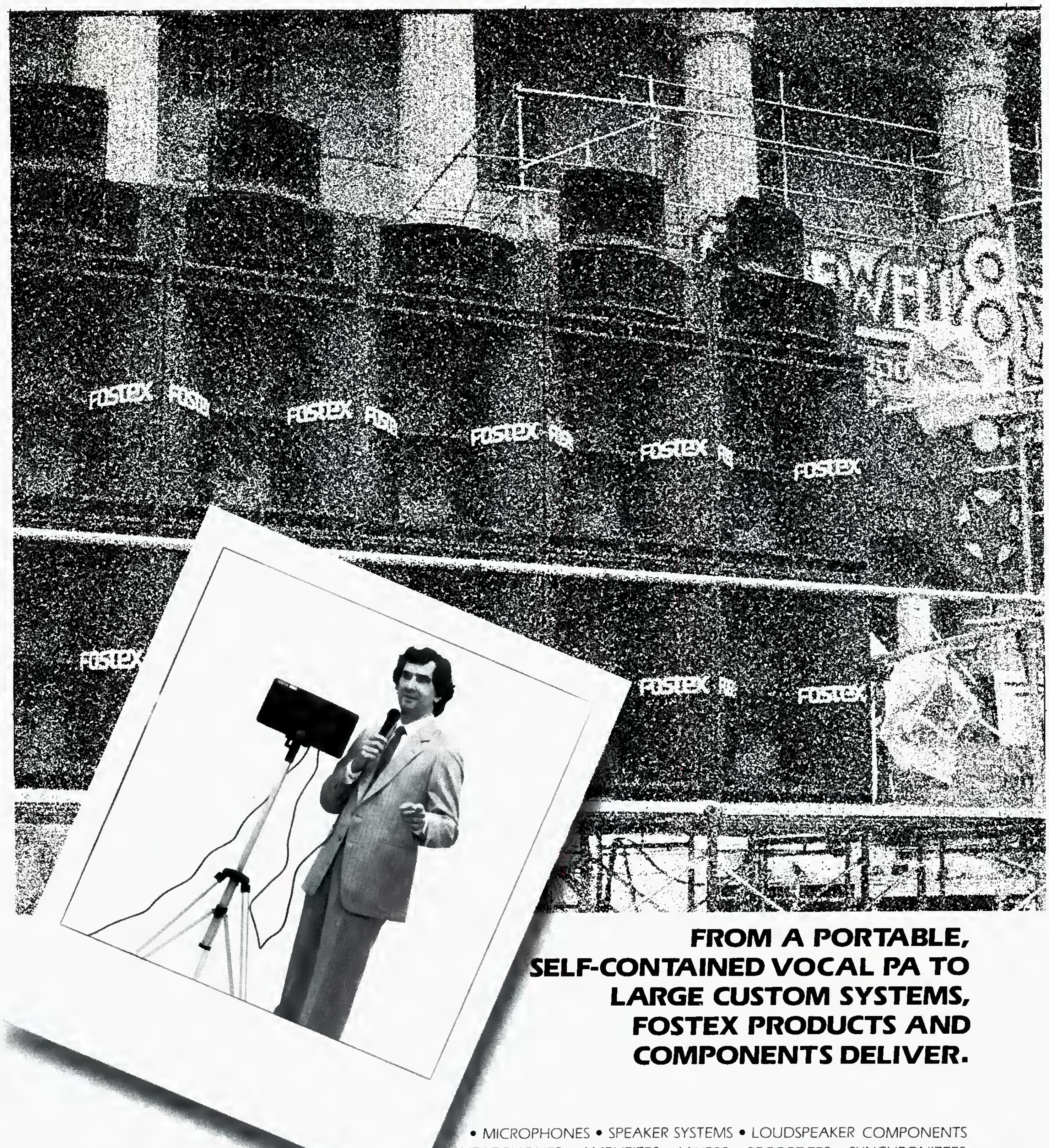
To The Editor:

I was very disappointed to find out that *Modern Recording & Music* is no longer. It was of immense help to me, one of the thousands of amateurs with little money and no formal training. In addition to being helpful, it was enjoyable to read. The attitude of the writers was not one of snobbery. I felt that *MR&M* really cared about the readers and, more importantly, they knew what kinds of articles the readers wanted.

I appreciate the transfer of my subscription to *db Magazine*—it is a classy way of dealing with *MR&M*'s demise. Again, I just wanted to let you know I really appreciated *Modern Recording & Music Magazine.*

Joan Bull
Germantown, Md

P.S. On a separate topic, I am a computer programmer who loves music (obviously). It would seem to be a good idea for me to combine my programming career with my love of music. Do you have any advice on



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TALKBACK

NEEDS SOME FEEDBACK

To The Editor:

I think your magazine is excellent. I am planning to build a small 8-track studio in my home. I want to use it to record small bands and to be able to do editing on master tapes. I would like to know two things: 1) some advice on what kind of equipment I would need (please take into account that I do not have a large budget), and 2) how do I get rights to the name I want to give to my studio. Keep up the good work!

George C. Davidson
Neptune NJ

John L. Murphy, Carvin's chief engineer, has been kind enough to provide an answer.

In order to set up an 8-track recording studio there are certain pieces of gear that are essential and other pieces that are somewhat optional depending on the goals you have established for your recording mixer (capable of supporting 8-track recording, several microphones and a monitoring system). For a low budget studio no headphone monitoring will probably be acceptable . Otherwise, monitor loudspeakers and monitor power amps will be required. Most recording enthusiasts will consider a reverb system to be a required item, expecially for recording vocals.

When it comes to purchasing a budget priced 8-track tape recorder you really don't have many units to choose from. The only two that come to mind are the Tascam Model 38 and the Fostex Model 80. I suggest you obtain the manufacturer's literature on these machines to become familiar with them and then discuss the pros and cons of each machine with the prospective dealers.

The choice of a mixing board will be more difficult since there are many more manufacturers of mixers than of recorders. Don't make the mistake of buying a pa (sound reinforcement) mixer for the purpose of doing 8-track recording since PA mixers generally have no provisions for setting up a control room mix which is independent of the mix going to tape. I can't emphasize the importance of independent control room mixing and soloing enough. The board you select should be specifically designed for recording with (at least) an 8-track tape machine. You will need no less than 8 input channels to allow mixing of your finished 8-track master tapes and you will probably want 12 or 16 channels to allow for more complex operations such as the multi-mic'ing of drum sets and the recording of larger groups. The mixer must allow you to send and return signals to and from outboard effects and must have provisions for setting up a cue mix for the musician's headphone mix. One such recording mixer that I can highly recommend is Carvin's model MX 1688 ($2995). This mixer has 16 input channels, 8 output channels, 4 auxiliary busses, two effects returns and was designed specifically for 8-track recording.

If you intend to monitor your recording activities over headphones then you will want to use highly accurate phones (units from Sennheiser or Sony come to mind) to insure that your recordings do not reflect compensation for problems in the monitoring system. The same applies to a loudspeaker monitoring system should you have the space and budget for one. Your monitoring system (phones or speakers) is no place for budget cutting.

For low cost but good quality microphones, I suggest you look at the lines available from Electro-Voice and Shure (the E-V PL-9 is a low cost but very accurate mic). Microphones (like monitors) are critical if you are going to produce natural sounding (uncolored) recordings.

You will almost certainly need a reverberation system of some sort for your studio. The reverb should be selected based on careful listening tests with a wide variety of program material. A simple digital delay line is no substitute for a reverb. However, the latest generation of digital reverberation units provide some excellent sound reverb at prices unheard of just five years ago.

Good luck with your new (low budget) 8-track! ■


VHS
Front Loading System
Hi-Fi
Audio
HD
REW
PLAY
FF
REC
STOP
PAUSE STILL

Mark E. Battersby

On Taxes

Deducting For Lobbying

• It's a sad fact that every small recording operation—and performer—must lobby merely to stay in business. That lobbying can be anything from providing support to local business associations to directly approaching local, state, or federal lawmakers to express your views. In fact, in light of the recently proposed budget and tax cuts, you may wish to lobby Congressional committees or their members.

Although this lobbying process can be an expensive proposition, our tax laws fortunately provide a number of incentives that should encourage you to make your views and opinions known—so long as you do it legally. Of course, as with most areas of our tax law, the regulations governing lobbying expenses have more than their fair share of potential pitfalls.

Most studio owners generally lobby in order to make their views on pending legislation known. It doesn't matter whether this lobbying is *for* or *against* a given proposal, a business or a group of businesses must usually spend money to make their views known. In 1962, Congress provided some relief by allowing a tax deduction for those lobbying expenses—a tax deduction that had not existed prior to 1962.

Although condemned by many political experts as an evil, lobbying is a necessary and a productive part of the American political process. After all, how else will a rural state legislator learn of the water problems of a large metropolitan district? Or, how else can our lawmakers hope to understand the problems these proposed tax and budget cuts will cause recording and sound operations?

Generally the facts and figures are presented to a key legislator, either personally or in committee, by the advocates of the various alternatives. Every time he receives a letter from a constituent supporting or opposing legislation, that legislator is being lobbied. Not only does the lawmaker need this assistance, in most cases he looks forward to receiving it.

The business person will say that he lobbies to stay in business. Most of us must be constantly alert to legislative activities that will affect either our present or contemplated projects. Whether it is a retailer who doesn't want a sales tax or the manufacturer who is interested in union-related legislation, there is one thing that we all have in common—we want to protect our operations as they are today.

Perhaps the most familiar type of lobbying is the appearance before Congressional committees. Fortunately, the tax law's description of lobbying goes much further. It includes, in addition to those personal appearances, the submission of statements and communications to the committees of Congress, as well as to the individual members. The same rules also apply on the state level, that is, for the state legislatures. Lobbying local town or county councils would also presumably qualify because they are subdivisions of the state.

Unfortunately, in giving relief to the business community, Congress did not write a blank check. For instance, there is no tax deduction available for lobbying expenses unless you can show that your business has a direct interest in the legislation. Fortunately, the tax rules take a fairly liberal approach in defining just what a so-called "direct interest" is.

For instance, any legislation (actual or proposed) that will have an impact on your trade or business is of direct interest to you and your recording or sound operation. And, going even further, if you can reasonably expect that at some time in the future, the present legislation will have an impact on your business activities, it will also meet the deductibility test. Naturally, if the legislation's effect is remote, speculative, or personal, it fails.

Some examples of the type of legislation that would generally meet the direct interest test are:

—A bill that would increase or decrease the taxes on your home studio, the operating costs, earnings or administrative burdens.

—A bill that would increase Social Security contributions or liberalize the right to receive them would be of interest to an employer.

—A bill that is favorable or unfavorable to competitors is of direct interest to any business.

—A bill that would improve the local school system is of interest to a community trade group.

—A bill creating a special tax credit or exclusion for shareholders is of direct interest to a corporation.

A bill is of direct interest to any business organization to which you may belong if the subject matter has a direct interest to the organization itself, or to any one of its members. Thus, it would seem that virtually every legislative proposal affecting businesses would be of direct interest to your local Chamber of Commerce or other similar broad-based organizations. This point is important since you can deduct the cost of providing your organization with information if the legislation is of direct interest to it.

The deductible expenses of lobbying are the actual costs of: 1) appearing before the legislative body, 2) preparing the testimony or communications, and 3) dues to organizations.

Your travelling expenses that are directly connected to the legislative appearance are also tax deductible. This means that you must, in order to get the deduction, maintain a travel diary just as you do for regular business travel.

The income tax regulations tie deductibility directly to the stated activities of appearances or communications. This leaves the question of whether you can legitimately claim a tax deduction for entertaining a lawmaker unanswered. By implication that is probably a reflection of the fear of potential wrong-doing being underwritten by a tax deduction and, thus, the costs are not specifically tax deductible.

The directly connected costs of preparing your presentation are deductible. If your employees prepare the material as part of their regular work schedule and no extra costs are incurred, no special tax deduction is needed or available. You can, however, deduct those expenses as a general business expense. You get the benefit of the lobbying deduction when you hire specialists, incur over-time work or other expenses that are specifically traceable to your lobbying activities. Thus, the cost of retaining a professional lobbyist or council to prepare and present your views is deductible.

Lobbying activities directed towards the appointment of an individual, a nomination to office or the like, are not tax deductible as lobbying expenses. According to the rules, a taxpayer cannot have a direct interest in such matters. Likewise, there can be no direct interest in matters concerning the actual operation of the legislature.

Deducting dues as a lobbying expense is complex and often confusing. This complexity results, largely, from the tax law which covers not only lobbying expenses but also political campaign expenses and the cost of influencing the public on legislation.

Simply stated, you can legitimately deduct that portion of any dues that you pay to any organization that are attributable to its lobbying activities regarding legislation or proposed legislation that is directly related to the organization or any one of its members. However, any portion of the dues attributable to other activities covered in the tax regulations (politicking and the like) are not tax deductible.

Any portion of a special assessment by one of those organizations that meets the general dues deductibility test as outlined above, is also deductible. This provision could prove valuable since the cost of the lobbying presentation is rarely known at the time the dues for the year are levied.

As we've already pointed out, purely political type expenses are not deductible. This means that efforts to involve the public at large in supporting or defeating legislation in the state houses or at the polls will not produce a tax deduction. For example, if Able Company were to engage in any activity to make the public aware of any proposed legislation, this would be considered a nondeductible "grassroots" campaign.

The Internal Revenue's view of what constitutes a "grassroots" campaign is extremely broad. For instance, when a trade organization contacts its members about legislation affecting the industry, this is normally a deductible expense. However, if in its message, the association urges its members to engage in "grassroots" activity, then the association itself will be considered to have commenced a "grassroots" campaign.

Conversely, a tax-exempt trade association is not engaging in "grassroots" lobbying when it urges members to write or call their lawmakers to recommend support of legislation of direct interest to the association. But then, should such action be directed at prospective members, it will be considered to be "grassroots" lobbying by a vigilant IRS.

Overall, however, the expenses you incur in expressing your views to our lawmakers on existing legislation or proposed legislation such as the recently proposed budget and tax cuts are tax deductible lobbying expenses. It is also obvious that attempts to educate the public in just what any given legislation will do to your business will have to be financed without the help of a tax deduction.

The lobbying expense deduction rules are complex and narrowly defined as you have seen. But they should never be ignored. Expressing your views and opinions to our lawmakers is never a waste of time while those tax rules can make certain that lobbying is not a waste of money—regardless of the outcome. ■

Fiber Optics

• Very often a new technology is the result of a desire to invent a solution to a problem, but the new technology eventually divorces the orignal problem. Perhaps it is somewhat like teenage children growing up and starting their own family. Those new families may hve different dissues, different values, and different goals in life. The original parents may be either proud or disgusted at what they have produced in the later generations but they are powerless to control it. Unguided missiles cannot be steered once they are launched.

The parents of digital audio were trying to solve a problem with signal to noise ratio and the general issue of degradations of analog signals. Digitization was the family that resulted. Those children are now having their own families; and they are not particularly interested in degradations because there are none in the current digitization process. Instead, they are interested in many other issues.

Interconnection and long distance communications of digital audio becomes a special set of issues because of the extremely high bandwidths required to handle the electrical signal in the digital format. Moreover, the interconnection of studio equipment becomes very complex when the bit rate enters 1 to 50 MHz region. The telephone company was the first to really understand this problem. For a single telephone conversation, one requires a single twisted pair of copper wires. For 100 conversations, one requires 100 pair. As the numbers get larger, the problems become very interesting. A large European broadcast house has a very large number of wires.

Enter the world of optical communications. Signals as we would like to think of them are nothing more than abstract information. That information can be communicated when it is encoded into a physical carrier such as electric voltage. While we classical engineers think of a voltage signal as a single semantic entity in reality it is a compound phrase: voltage + signal. The voltage part of the phrase can be replaced by other physical carriers such as light. Both light and electric voltage are really the same type of energy except that the frequency is different. If an electric signal were to become a high enough frequency, it would be light. However, at these higher frequencies, the energy cannot be conducted in wires. Light can, however, be conducted in special cables called: light-guides, fiber optics, or

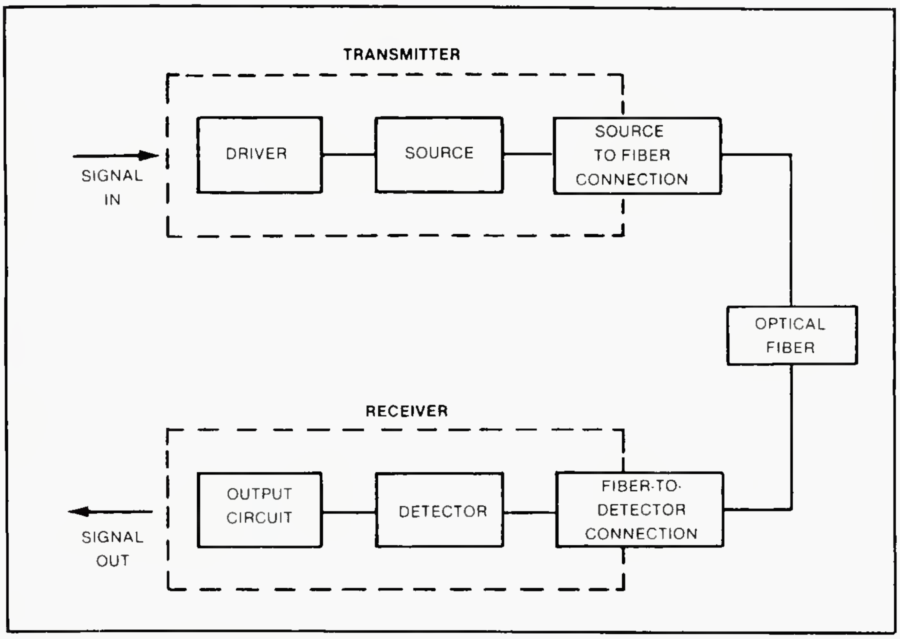


Figure 1. A basic communications system using fiber optics instead of wire for digital data. [page 2 from AMP].

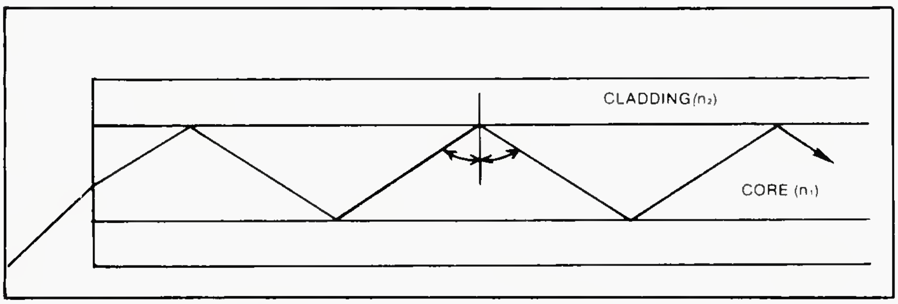
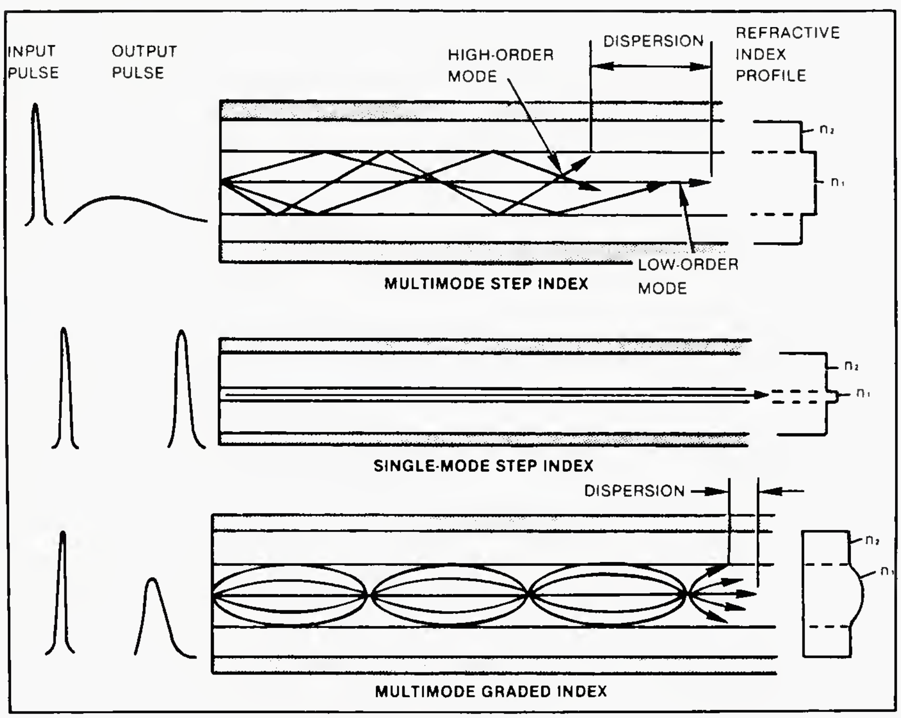


Figure 2. (top) Light entering from left into core with index n_1 reflected at the boundary with the cladding having index n_2. [page 21 from AMP] (bottom) various paths for light ray in different types of fiber optic cables. The core size and refractive index determine the light propagation characteristics of the fiber. [page 23 from AMP].



light pipes. These new conductors of information offer some very interesting advantages over electric signals in wires.

While the basic ideas have been known for a long time, the new technologies make it very practical to use light as an alternative to copper conductors. These developments are being driven by both the telecommunications and computer communications fields; but they will begin to show up in the digital audio context within the next decade. By 1995, audio technicians will no longer be using a soldering iron when wiring a studio; they will be polishing the surface of fiber optic cable for splicing.

AN OPTICAL COMMUNICATIONS SYSTEM

A typical optical communications system in shown in *Figure 1*. An electrical signal such as the bit stream from a A/D converter is fed into an optical modulator. This might be nothing more than an LED (light emitting diode). That diode produces light when the electrical current flows in the diode. Thus, a logical 1 would correspond to light on and a logical 0 would correspond to light off.

The light output of the LED is then coupled to a fiber optic cable which communicates that information to the other end. There we find a light demodulator which might be nothing more than a photo-transistor. It reconverts the light to an electrical signal. Notice that the basic system of communications is very trivial.

The fiber optic cable is a very simple light pipe which has the property that the light is trapped inside and cannot get out. Think of a glass rod with a mirror surface on the outside. The light is reflected back into the center of the rod and bounces (reflected) from the source to the destination. *Figure 2* shows a typical pattern of light reflections inside the cable. The two elements of the cable are called the "core" which is the conductor of light and the "cladding" which is the reflector of light. The process of reflections is perfect; no light can leave the cable except at the ends.

The reflections are created as a result of some very simple physics. The cladding is also a glass type material except that it is a different kind of glass with a different refractive index. This index is nothing more than a measure of the speed of light. In other words, the fiber optic cable is nothing more than a pipe of glass surrounded

How To Hear Yourself

Getting enough vocals on stage. It's a problem facing every performer. In fact, most smaller bands simply can't hear the vocals clearly when they perform.

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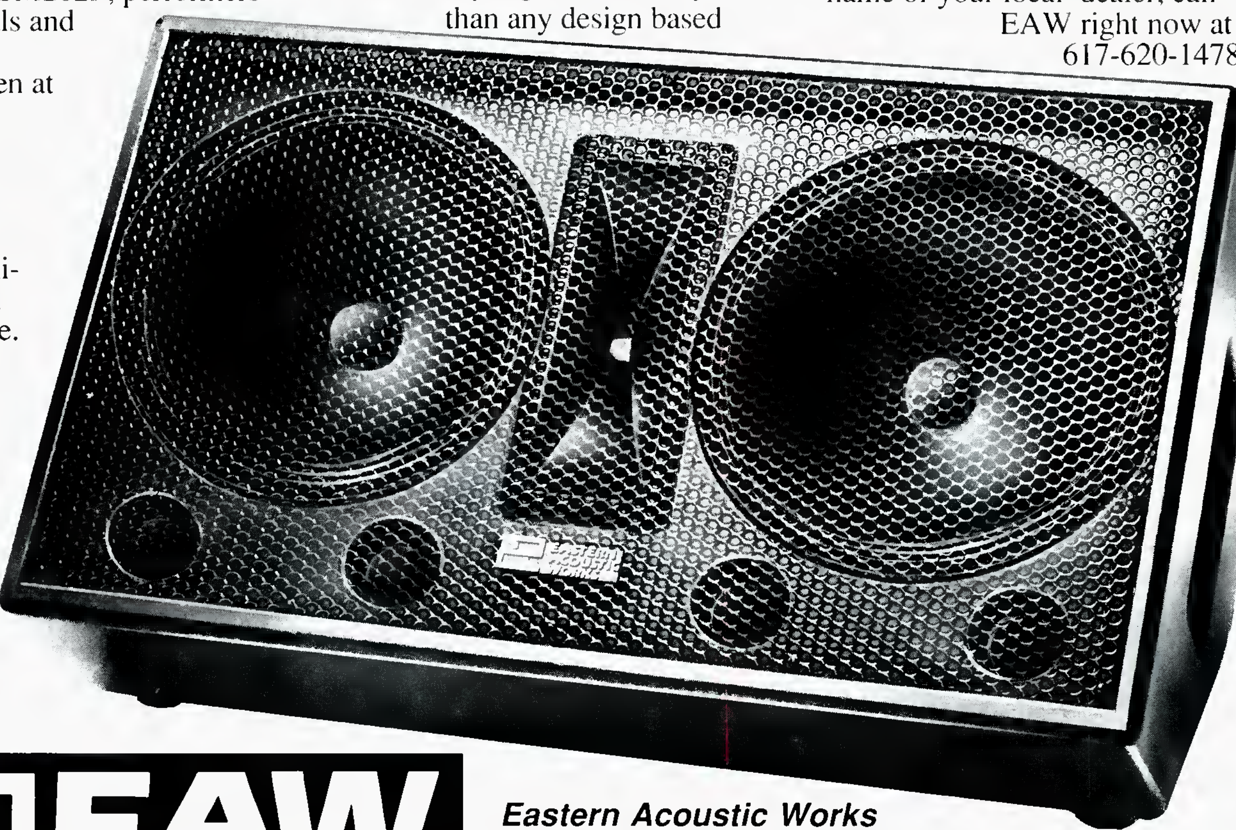
Check Out The SM202P At Your Nearest EAW Dealer.

There's a lot more to the SM202P story than we can talk about in an advertisement. And, it's just one of many EAW solutions to today's sound reproduction problems.

For more information and the name of your local dealer, call EAW right now at 617-620-1478.



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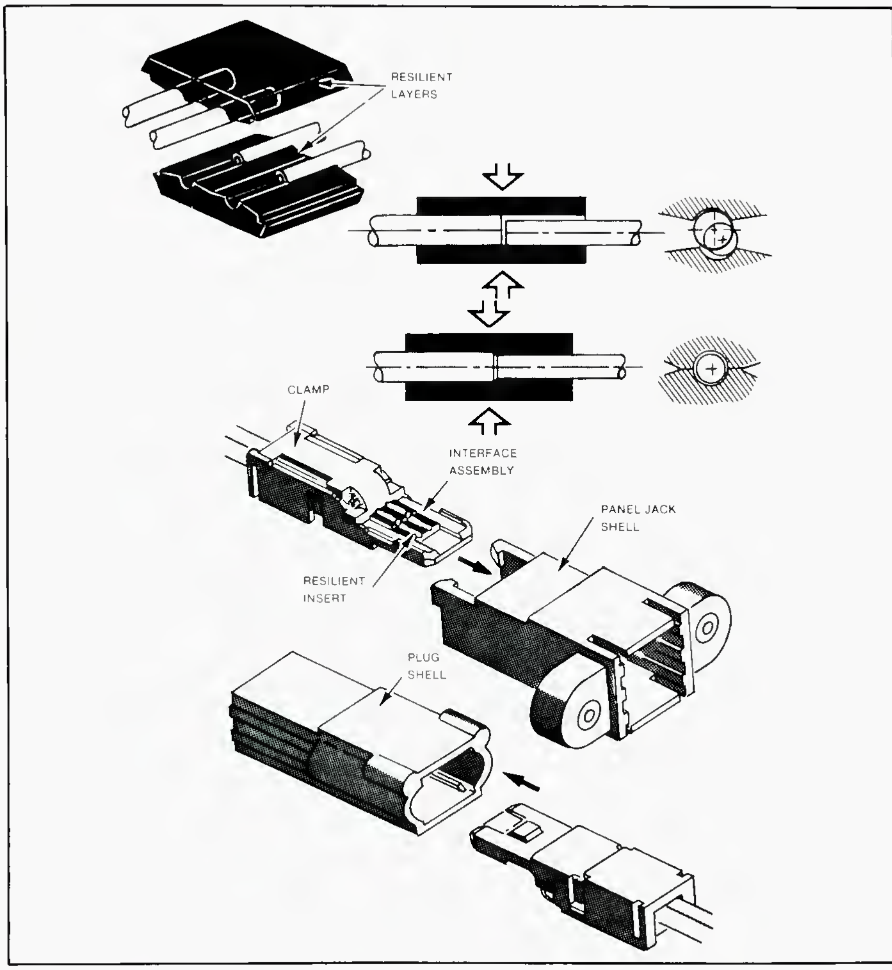


Figure 3. A fiber optic connection method using a commercial technique for users of fibers. [page 83 from AMP].

with another coating of glass. The boundary creates a perfect reflector. We are all somewhat familiar with this idea of reflections as a boundary. We will take the following analogy. Water is a good transmitter of light. One can see underwater when swimming. However, when standing on shore and looking out over a lake, one sees the trees reflected on the water surface and does not see what is under the surface. In this case, the air and the water have a different index of refraction and at certain angles, the water-air boundary looks like a perfect mirror. Unlike the mirrors in a bathroom, the mirror created by the change in index of refraction is perfect.

Already we can see some interest properties of this kind of communications. There is no crosstalk even at very high frequencies; this contrasts to coax cable which does produce some crosstalk even with double and triple shielding. Certainly ordinary twisted pair will crosstalk. Bandwidth is extremely high thus allowing a large number of signals to be multiplexed. Fiber optic cable is extremely difficult to tap; hence there is good control in terms of secrecy and privacy. Glass fiber is inherently inexpensive since it only uses sand as a raw material. After one has listed all of the advantages, one can begin to understand why the telephone companies are converting to this mode of communications. Because of the size of that market, the technology will become very large volume and very low cost. Other industries, such as ours will then begin to use that technology.

Before we continue with the general discussion, let us stay with the puzzle of how the index of refraction change at the boundary creates a perfect reflector. You really do not have to understand this idea to remember that the light stays trapped in the cable. We will do this by analogy. I like teaching by analogies and it is always a challenge to create interesting ones. Consider a tank with independent drive on the left and right sides. Now consider a road constructed with a center region made up of sand and shoulders made up of concrete. The tank treads will obviously run faster on the concrete part of the road than the the sand part. Now let us assume that the tank is at an angle to the road direction such that it is aimed slightly to the shoulder. The

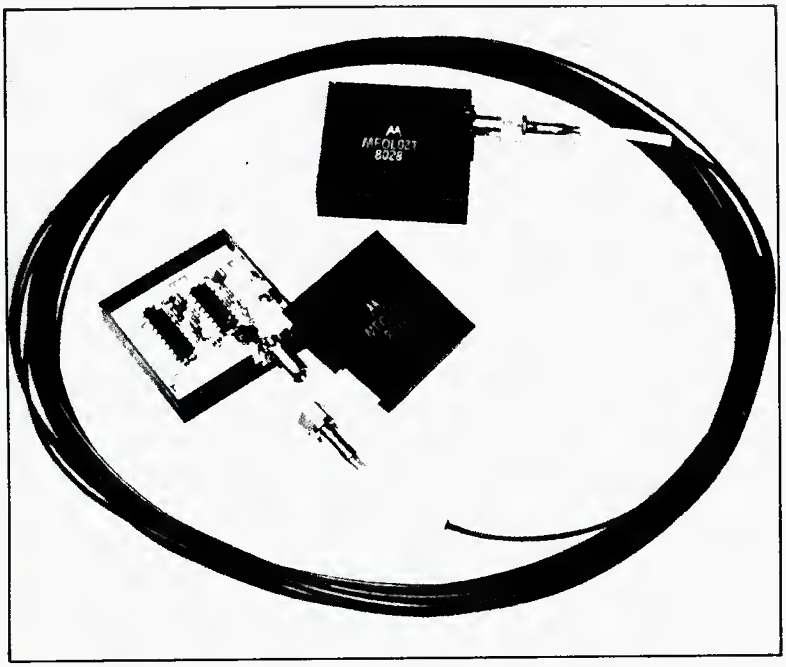


Figure 4. A complete low cost fiber link as might be used in a digital audio system. [page 124 from AMP]

outside treads will enter the region of the road with concrete and hence it will go faster. That tends to turn the tank back toward the center of the road. Now as it heads to the other shoulder the reverse process takes place. Without steering the tank, it can never leave the road no matter what happens. The tank is automatically steered by one side being able to go faster than the other. The path of the tank is shown in *Figure 3*. This kind of "road" is called a "multi-mode graded index" if the transition between sand and concrete is gradual; it is called a "multi-mode step index" if the transition is sharp.

CONNECTIONS

This is the only interesting topic in fiber optics. The problem of connecting the fiber cable to anything else becomes non-trivial. We must couple the light. A splice becomes difficult because the two ends of the cable must be smooth, planar, parallel, and touching. That would not sound like a problem if the core of the cable were 1 inch in diameter. The cores have diameters in the range of 50 to 1000 micrometers. The largest core is 1 mm! Special techniques and equipment are thus required in order to work with such a small surface area. The industry has evolved special technician jigs and tools as well as measuring equipment. For only a few hundred dollars, one can buy such a facility to handle splices.

The same problem takes place at the connection between the cable and the LED and photo-diodes. These are optical connections which require surface preparation. New connector assemblies do minimize this difficulty. The quality of the connection is a more profound issue in the telecommunications area than in the audio industry. A poor connection results in light attenuation. With short runs on the order of 500 feet, there is a large amount of excess capacity and some attenuation is not a problem. With long distance runs on the order of 10 miles, one needs all of the light to travel over the full run and one cannot waste it in the connection process. The studio application is thus a "low technology" in terms of what the fiber optics is able to handle. Several manufacturers offer experimenter kits at very low prices.

Figure 3 shows a typical plastic connector for fiber optics. The main issue is the mechanical holding of the fiber such that the center align and the surfaces are flush and touching when the two ends are connected together. While manufacturers admit that it is not trivial, it only takes a day or two of practice to make good connections. Even a relatively poor connection having 5 dB of loss would be acceptable for short runs. *Figure 4* shows a final assembled link with a cable and the detector and source modules. We can expect this type of technology to begin to appear in the audio industry.

We having incorrectly implied that the transmission bandwidth is virtually unlimited, in fact however, there are degradation modes which limit performance. In the least expensive type of cable, plastic step-index multi-mode, the typical specification is on the order of 200 kbits-km. This means that one could have 200 kbit/second rate for a 1 km cable but 2 Mbit/sec for a 100 meter cable, etc. The limit comes from dispersion of the light. When a light pulse is entered into the cable, some of the light gets to the destination faster than other light. This is clearly illustrated in *Figure 2*. The speed of light is constant but the angle of the zig-zags determines the actual length of the light path. The left hand part of this figure shows an input pulse and the corresponding output pulse. If the spreading is large, then the pulses must be separated in time so that they still exist in a unique place. The degree of spreading is controlled directly by the cable length. Hence, the bandwidth specification is strictly proportional to length.

THE FUTURE

The use of fiber optics in the studio and broadcast house is limited by several factors. Audio engineers have no experience with fiber optics. The technology is relatively new and very rapidly evolving. The costs are just now dropping rapidly as the telephone industry becomes a big user. And finally, certain types of support elements are not yet available. For example, it would be nice to have an inexpensive "concentrator" which could multiplex a large number of digital audio signals onto one cable. Consider a higher grade glass cable that can take 500 Mbits-km. At 500 Mbits/second, a single cable could transmit 500 audio signals. A full studio complex could be wired with a single cable!

ACKNOWLEDGEMENT

All of the figures are reprinted with permission from Designers Guide to Fiber Optics, AMP Incorporated, Hassisburg, PA 1982. ■

Brian Battles

A Venture Into the World of Ads

• This issue marks my first appearance in *db Magazine,* although many loyal readers have followed "Ad Ventures" in *Modern Recording & Music Magazine.* (A brief moment of silence will be held at the end of this piece to commemorate the passing of a great publication.) Therefore, rather than start off from where I left off, I'll present a summary of what this column covers, and the types of topics you can expect to learn more about in the coming months.

"Ad Ventures" is the place to look each month if you're interested in earning money by using your recording facility as a place to produce commercials. Radio and television advertisements often employ a great deal of high quality talents in order to better sell a product or service, and your studio just may be well-equipped enough to provide the essential abilities. Broadcast commercials frequently use music and sound effects in an innovative fashion in order to get attention and build name recognition for an advertiser. It's true that the majority of sponsors on the local level tend to simply leave it to the individual stations to write and produce their spots. Advertising, however, like nearly everything else is becoming increasingly competitive, and it's not too hard too convince advertisers to spend a little extra money to position themselves more prominently before the public. Your expertise in audio production work, coupled with the capabilities of a properly equipped recording studio, can provide a critically important solution to the problem of making a client's advertisements stand out and get more results.

Can you do it? Do you have the necessary equipment? Do you have sufficient connections? Do you know how to make contact with the right people? Do you possess the talent required to make a strong audio statement that compels consumers to take an interest in a particular product? I think so. I spent several years as a production director at some major metropolitan radio stations, and I can assure you that most station management places very little stress on creative production of advertising spots. The main emphasis is on sales of commercial time. The message is secondary to the business of getting the sponsor on the air. That leaves three people in the postion to furnish creative services: 1) the client, who is normally a poor source of inventive advertising ideas; 2) the radio or television station's production staff, who is often overworked and underpaid, and has no particular interest in investing effort in anything fancy; and 3) you.

What's so special about you? To begin with, *you* are in the music and recording field, so your life centers around entertainment. Anything that hits the airwaves is in competition with all of the other stuff cluttering up the ether, so you'd better believe that one of the main goals of advertising is to be entertaining.

Bad commercials inevitably lead to a twist of the tuning knob, and a commercial does no good if no one ever listens to it. Next, you have access to a recording studio that's probably better than eighty-five percent of the broadcast station production rooms in the country. I'm not kidding; if you somehow wangle a tour of a couple of local radio stations you'll be appalled at what passes for audio equipment. I've seen dozens of well-known stations with production rooms that would be an antique collector's paradise. Dusty rooms with painted-over acoustic tile walls, banged up mics, blown speakers, rusting old single track tape machines with crackly pots and worn out motors, mixing control boards that look like props from cheap sci-fi movies, and other odd pieces of electronic junk that probably served some advanced purpose in the days before the Allies captured German tape recorders near the end of World War II. Okay, there are many exceptions, but you'd really be surprised at the number that fall short of 1960's state-of-the-art. That means you have an advantage. Exploit it! There truly are advertisers who, if they understood the difference, would be overjoyed at having your skills at their disposal. What you have to do is seek out these businesses, make a logical and professional presentation, and sell them your services.

If you'll join me next issue, I'll help you to see the various methods you can use to attract business for your own ad ventures, and we'll look at some techniques that can help you make your use of production time more efficient, and perhaps give you some pointers on your sales presentation. I understand that there are quite a few *db* readers who are already involved in advertising work, and I certainly hope we can share your ideas and suggestions, and provide answers to your questions and problems.

Let us now observe that moment of silence I mentioned earlier, and then pick up your crayon and drop me a note. ■

Editorial

THIS ISSUE is new to some readers who may be seeing it for the first time at the NAMM (National Association of Music Merchants) show in Chicago's McCormick Place.

It *is* proper for *db Magazine* to be at a NAMM show since today NAMM is the show for the smaller studio market as much as it remains a show for music dealers that only sell musical instruments from guitars to grand pianos.If evidence is needed for this, simply look at the most recent exhibitor lists. There are two NAMM shows a year, one in January in the West Coast, and one in Chicago, Atlanta, or New Orleans in June. As each show has come and gone over the last several years, we have noticed more and more professional audio manufacturers as exhibitors. The smaller studio market is growing, and NAMM is growing with them.

If you, as an audio professional haven't been in stores such as the New York area's Sam Ash or the West Coast's Guitar Centres, you will be in for a surprise. Typically, in one of these stores you walk into a huge collection of synthesizers, big power amplifiers, reinforcement and performance speakers, and, of course, many microphones.

In these stores you will have to hunt to find where they have hidden the acoustical instruments. For example, at the Sam Ash headquarters store in Hempstead, NY, the instruments are in a side room near the main entrance to the store.

All this is by way of saying that a good segment of the professional audio business is now being sold by "music stores." But are they music stores? They are really equipment stores that also sell musical instruments.

Since we now serve this market especially through our 2 to 8 trk section, *db, The Sound Engineering Magazine* belongs at NAMM.

LZ

2 to 8 trk

WHEN YOU MUST BE 100% MUSICIAN

MCR™ 4 Multi-Track with "Overdubber™ pedal remote"

Historically, musicians doing their own multi-track recording had to be half engineer and half musician. When you do it all, it's difficult to work out musical parts, concentrate on your playing in addition to punching buttons, moving faders and watching levels to make multi-track tape recordings.

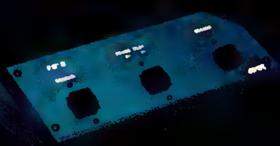
Most musicians are familiar with foot-switches used by guitar players. They allow the performer to change effects or sounds by simply tapping a foot-switch, with no distraction to their playing. This quickly becomes second nature.

Overdubber™ brings the same concept to multi-track recording. It has only three buttons and is simple to use. Now recording tape tracks can be as simple as using a guitar footswitch.

What is the Overdubber™ Pedal Remote?

With the Overdubber™ pedal remote, you can play and rehearse, pause and work out parts, work on the whole tune, work on any portion of the tune, punch in, punch out, rewind and listen over and over and over again. Most important of all, you can keep your mind on the music, your foot will do all the control as second nature. You can now be 100% musician when you need to be. The MCR™ 4 with Overdubber™ pedal remote is only available from AMR. Try it for yourself at your local AMR dealer or call the factory for the name and location of your nearest AMR dealer.

MCR™ 4 Multi-Track Cassette System for Musicians



- Overdubber™ Pedal Remote
- Solenoid-operated Transport Function Controls
- Dolby B and C*
- Zero Stop and Zero Play functions
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- Headphone Jack
- Headphone Track Selector Switch
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- Many Additional Features
- Made in the U.S.A.

*Dolby® is a registered trademark of Dolby Laboratories Corporation

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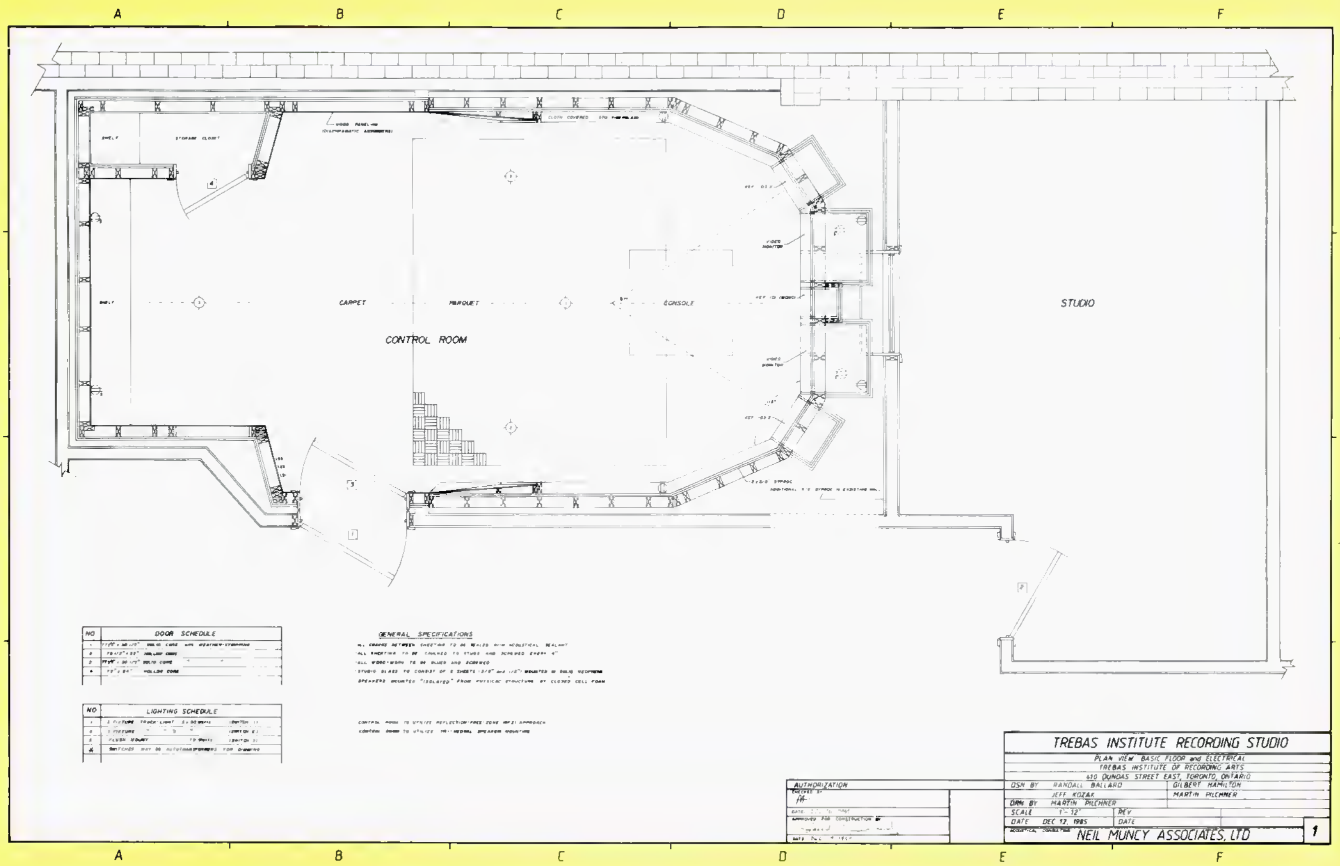
The Trebas Institute of Recording Arts Student Construction Project

David P. Leonard

Students design and construct a professional sixteen-track recording studio.



Diagrams courtesy of Trebas Institute and Neil Muncy Associates.

Basic floor plan for Trebas studio.

A BACKGROUND

THE TREBAS Institute of Recording Arts is a private two year college providing education and training in the recording arts and sciences, and has been in operation since 1979. The Trebas philosophy has always been to provide the best quality and most comprehensive type training for people wishing to enter the high tech music and recording industry.

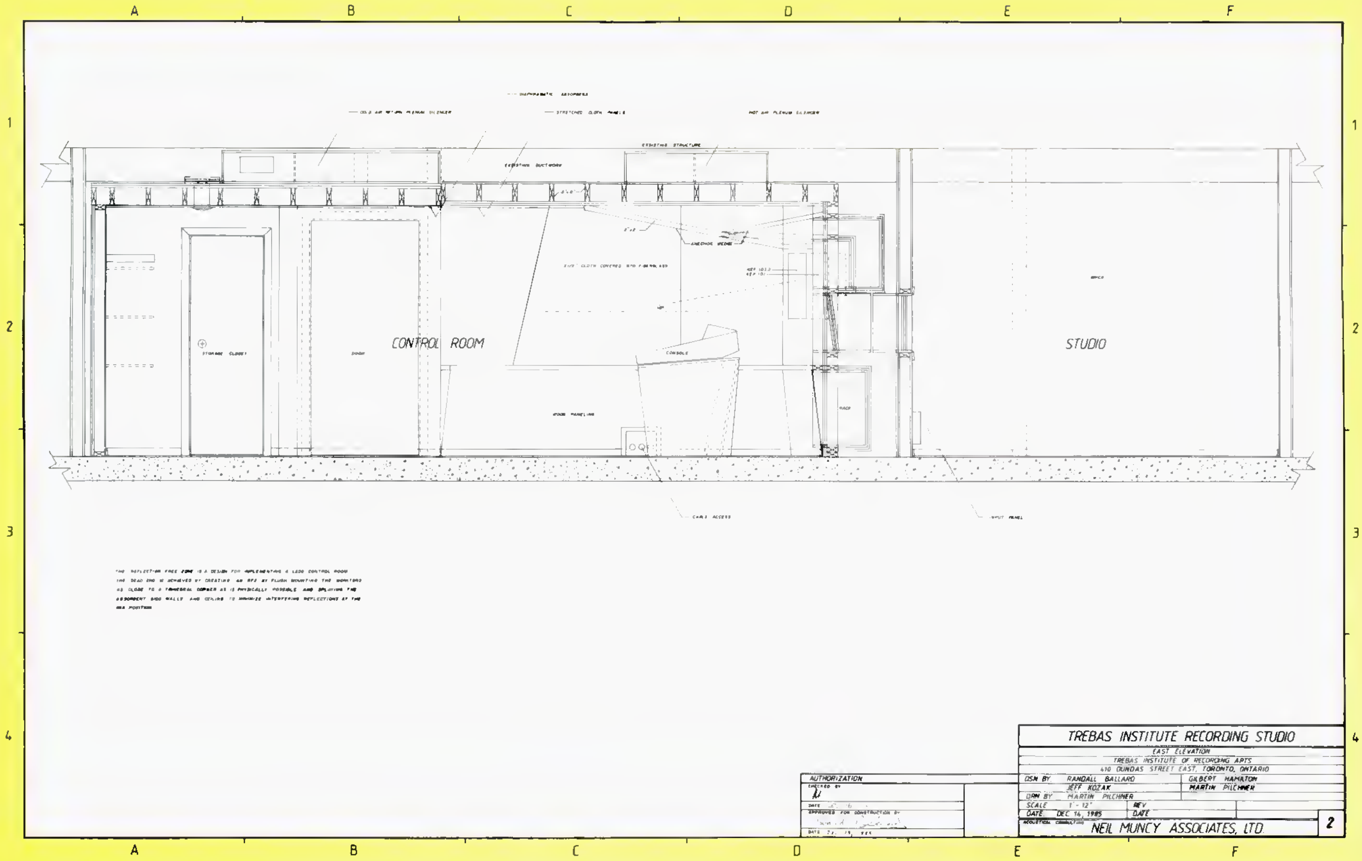
Trebas combines theoretical instruction with practical hands-on training. The range of courses available run in three programs: Record Producing, Audio Engineering Technology, and Music Business Management. They include: record producing, music theory and ear training, song-writing, publishing, copyright law, electronic music, sound recording theory, acoustics, studio design, signal processing, electronics, digital audio, disc mastering, talent development and management, and multi-track recording.

Neil Muncy's article will appear in our July/August issue.

Trebas Institute started with one campus in Montreal, Canada, and has since opened campuses with the same curriculum, in Ottawa, Toronto, Vancouver, and most recently, in Hollywood, California. All curriculum is under the personal supervision of founding President, David P. Leonard, who travels across the continent to meet with his sixty-five instructors, in order to maintain a standard curriculum.

Until a year ago, Trebas rented 24-track professional

David P. Leonard is the president and executive director of the Trebas Insitute of Recording Arts.



East elevation diagram.

recording studios, in some cases with digital recording facilities, to the tune of 4,000 hours per year, as the sole source of recording facilities for their students. However, last year, in addition to the major multi-track facilities provided to the students, Trebas decided to build in-house recording/mixing labs and studios across the country. This provided a unique opportunity for students to become involved, through their acoustics courses, in a very practical project—the design and construction of a real professional multi-track recording studio.

The students' input included, (under the supervision of acoustics instructors): site selection; budget preparation (for materials, equipment, and labor); construction of room; installation of equipment (under supervision of Studio Maintenance course instuctor, Rick Austin, from Tele-Tech Associates, who supplied the equipment); room acoustics testing (under Neil Muncy, acoustics consultant to Trebas Institute and the project); overall evaluation; and finally, the management of the studio itself.

(Editor's Note: The previous names are for the Toronto campus' studio. However, a similar arrangement was made for the other Canadian campuses. The Hollywood campus will be building its studio next year.)

An acoustics course is usually very theoretical. Yet, this project provided a unique opportunity for the students to get hands-on experience in the planning, design, and construction of a professional recording studio.

Comments received to date from musicians, professional sound engineers, record producers, and other industry professionals in Toronto suggest that the Trebas Institute recording room is one of the finest sounding and most accurate control rooms in Toronto today.

All photos courtesy of Trebas Institute.

Student displaying wood formers for walls.

The recording facility was designed, not to replace the 24-track studios that the Trebas Institute uses to give its students professional calibre training (these are still in use), but to offer the students hands-on experience in understanding the complexities of signal path routing and the use of signal processing devices such as compressors, limiters, equalizers, reverb chambers, digital delays, and so on, that are covered in the Sound Recoding Theory courses. The facility is also used for recording student projects in their electronic music lab courses, where they use a combination of digital FM synthesizers (like the Yamaha DX-7) and analog synthesizers (like the Synthi AKS from Electronic Music Systems of England).

Wood formers before the walls are completed to allow placement of equipment.

Students building wood formers for walls.

The control room is connected electronically and visually, with a window and TV camera and monitors, to the electronic music studio. This is used for recording both musical instruments and voice. The two video monitors in the control room allow users to view television pictures and computer data and allow the audio equipment to be synced up to picture. (The sync equipment has just been selected at this time.)

THE STUDIO DESIGN

The studio was designed by Trebas students: Randy Ballard, Gilbert Hamilton, Jeff Kozak, and Martin Pilchner. The facility was constructed by Randy Ballard, Jeff Kozak, Martin Pilchner, Kurt Viebacker, and Andy Yue.

Student working on soundproofing and insulation.

Initial studio design work incorporated a number of different designs, each addressing a different problem in the limited volume available to work with. In consultation with Neil Muncy of Neil Muncy Associates Ltd., the design that was chosen allowed the greatest number of cubic feet so that a proper control room could be constructed. It is important to point out that this decision excluded a "studio proper" of appropriate size and treatment. However, the control room was constructed to serve more of a mixing lab/keyboard studio function.

In finalizing the plans for the studio, several hours were spent consulting with Neil Muncy to streamline the design. Although the physical volume was limited, the "reflection free zone" (RFZ) principles were still applied as outlined by Neil Muncy and as documented by Dr. Peter D'Antonio. Ultimately, the design may incorporate "RFG Diffusers."

A QUESTION OF EQUIPMENT

The criteria for choosing equipment for the studio centered around the idea that the room would be used mainly as a mixing lab for students. Therefore, the equipment had to be somewhat representative of equipment used in professional studios, to facilitate knowledge transfer, and yet had to fit within a realistic budget. A 16-track format was chosen because it represents a happy medium between an 8-track home studio and a full blown 24-track room. The Tascam MS-16 fit the bill as the multi-track because the performance for the dollar was high and the 1-inch tape format decreases tape costs.

The finished studio.

The Soundcraft 600 Series console was a logical compliment to the MS-16 and is a very flexible console offering features of both in-line and L configuration designs.

Following the premise of the room being a mixing lab, the selection of monitors was very important as accuracy was the main consideration. The KEF Loudspeakers were chosen as hands-down winners over other speakers of the same

Why should anyone else listen if you can't hear yourself?

A good monitor mix lets each member of the band hear exactly what he or she needs to hear. And that can be critical in helping any band play its best.

It's with this in mind that Yamaha designed the new MC monitor mixing consoles, the 16-channel MC1608M and the 24-channel MC2408M.

Both offer eight independent monitor mixes via eight outputs. And two auxiliary sends which can be used to patch in signal processors or tape recorders, as well as provide two additional mixes.

Each channel has a phase reverse switch, 20 dB pad, gain control, peak LED, three-band equalization with sweepable mid-frequency, two auxiliary send controls, eight rotary level controls, channel on/off switch, and channel cue switch.

The Input Channel Cue Priority System makes the monitor mix engineer's job a little easier. By pressing the cue switch on one or more input channels or auxiliary inputs, the master outputs being monitored are muted. So he can monitor only the selected input signal through headphones or speakers.

The MASTER section cue function permits the engineer to monitor an individual performer's monitor mix through his own headphones or near-field speakers without affecting the overall monitor mix.

The MCM consoles' communication facilities include talkback assign switches, XLR talkback mic input, and COMM IN jack and level control. So the monitor mix engineer can communicate with the house mix engineer as well as with the individual performers.

All primary inputs and outputs are electronically balanced with XLR connectors. And there are insert patch points on all input channels as well as the master outputs.

Yet with all this flexibility and these features, both the MCM consoles are lightweight and compact. And at $2,895* for the MC1608M and $3,995* for the MC2408M, surprisingly affordable.

So now that you've heard us, it's time to go to your Yamaha Professional Audio dealer to check out an MCM mixing console. And hear yourself.

For a complete brochure, write: Yamaha International Corporation, Professional Products Division, P.O. Box 6600, Buena Park, CA 90622. In Canada, Yamaha Canada Music Ltd., 135 Milner Ave., Scarborough, Ont. M1S 3R1.

*U.S.A. suggested retail price. Canadian suggested prices are $4,295 CDM for the MC1608M, and $5,695 CDM for the MC2408M.

YAMAHA®
PROFESSIONAL AUDIO SYSTEMS

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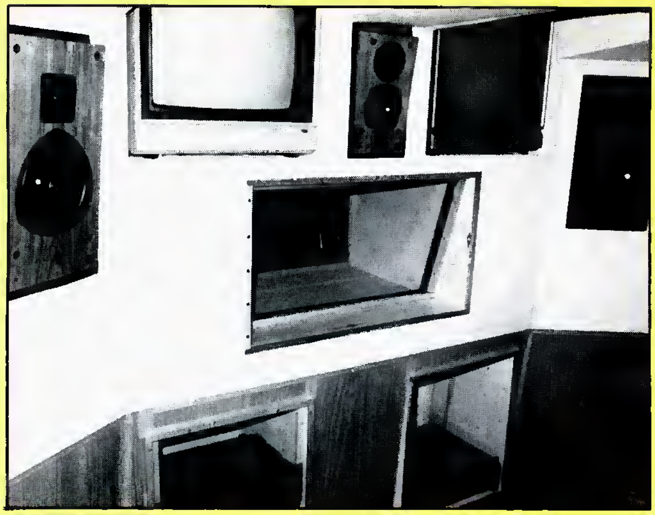


size in a listening test that was also quantified with test equipment. The S-Stop Circuitry of the KEF's was an added bonus since it protects the speakers very well against accidental overload. A center "mono" speaker was added to give the most accurate "monaural" mix possible at all listening positions.

The outboard gear and auxiliary equipment was chosen to complement the basic equipment and represent the kind of equipment used throughout the industry.

THE ACTUAL BUILDING

Construction was on schedule and proceeded very well.

The construction took place over a two week period during Christmas vacation for the school. The room was completed on schedule at the end of the two weeks.

ANTICIPATING PROBLEMS

As in any construction of this nature, problem solving was a daily, if not constant, event that kept the project moving. A few unforeseen obstacles came up that required some rethinking of original plans. The rear ceiling of the room was designed so that its entire surface area could be diaphramatic absorbers of varying sizes. This required the dry wall sheeting to be placed on the outside of the 2 x 6s used to span the ceiling. The original plan was to sheet the rear ceiling before lifting it on top of the wall structures. The ceiling was to be built in two pieces so that the weight would not be too much for us to lift by hand.

As it turned out, on the day the ceiling was ready to go in, we had only three workers instead of five, not enough to lift the weight. So what we did was just one end of the ceiling that would be directly under some ducting and, therefore, have only a couple of inches of clearance. The rest of the ceiling had the dry wall sheeting done by crawling on top in an approximately one foot air space and fastening and sealing the panels while laying on top of the ceiling itself.

AUDIO MONITORS

KEF 103.2 Reference Series Monitors (stereo)
KEF 101 Reference Series Monitor (Mono)
Yamaha P1150 Power Amplifier (Mono)
Yamaha P2100 Power Amplifier CUE Systems)

VIDEO MONITORS

JVC TM-2084U 19-inch color

TAPE MACHINES

TASCAM MS-16 1-inch 16-track with DBX
TASCAM 42 1/4-inch 2-track
AMPEX 300 1/4-inch 2-track
TEAC V5RX Cassette Deck
NAKAMICHI 682 2X Cassette Deck
TASCAM 3440 1/4-inch 4-track

CONSOLES

Soundcraft 600 Series with custom outboard patch bay
TEAC Model 2-A

OUTBOARD GEAR

UREI LA-4 Compressor Limiter
UREI 1176 LN Peak Limiter
DBX 166 Dual Channel Gated Compressor
ROLAND SDE-1000 Digital Delay
YAMAHA REV-7 Digital Reverb

MICROPHONES

NEUMAN U-47 FET
AKG 451 w/CKI & 10 dB PAD
SENNHEISER MC 421V
SHURE SM-57

THE GENERAL COST

The original studio budget of $5,000 proved to be an accurate account of costs required to build this room. Much attention to detail and price comparisons at the time of budgeting not only ensured that cost allocations would be adhered to, but that all materials were obtained at a competetive price.

THE CLIENTELE

The studio is used by the Trebas Institute students as a mixing lab and demo recording facility. Its main objective is to allow students to gain practical skills and knowledge of the recording studio equipment and overall operation. Sixteen-track studio time is provided as part of tuition costs and students book their time in three hour blocks. Multi-track masters can be used at no cost and additional cassettes can be purchased from the Institute.

Some outside acts have recorded in the studio for demonstration purposes only. However, there may be plans for the studio to begin some commercial use in the future. ■

Rita Wolcott

Red Brick Recording Studios From 2 to 8

RED BRICK Recording Studios, a 2-8 track basement facility, was originated back in 1962 in Colesville, MD, by engineer John Miller. Presently, the studio is a joint venture between Miller and musician Billy Gordon (of Billy Gordon and the Blue Rockers) who owns much of the equipment which is housed there. Gordon says, "John is the originator of Red Brick and we've basically been together the whole time. Eventually we worked out a situation where he gave me time in the studio and in return I gave him equipment. It works out really well because he likes to do his engineering thing and now he has the opportunity to work on better equipment. Actually, it began as a hobby and as our own venture and then other people decided they wanted to do tapes so we started working with them for a minimal fee."

Red Brick started out with a Wollensack 6250 3m reel-to-reel. Then in 1975, a 4-track TEAC, a 4-track TEAC mixer, 4-track Tapco equalization, and a 4-track TEAC Dolby AN 180 were added. The studio went 8-track in 1983 when an 8-track Tascam reel to reel, DBX, and another 4-track mixer (which was combined with the original) were added.

On expanding to 8-track Gordon says, "We didn't have to buy an 8-track board. We just bought another TEAC 4-track which we matched right into line with the other one. Then when we bought the 8-track machine we were able to use the four track in conjunction with that for 1/2 track stereo mixdown.

As is the case with all studios, Red Brick encountered some problems. Gordon remembers, "Some problems

Photo courtesy of Red Brick Recording Studios.

Control room at Red Brick showing the two TEAC 4-track mixers and 8-track Tascam recorder.

included calibrating 2- and 4-track boards to one 8-track machine. We solved the problem by purchasing a tone oscillator and calibrated to -10 dB at 1,000 cycles."

In regard to himself and the Blue Rockers, Gordon expands on the recording of their latest LP, *Psychotic Revolution.* "We recorded our first album in 1979 when Red Brick was a 4-track studio. You can tell that that's about all it was recorded on. Yet, the fact was that we had made an album with the minimum amount of expenses through the studio and through the record factory. It was more or less a learning experience, but on this album we had 8-tracks, and we took about two or three years to put it together in our studio part-time."

Red Brick's list of clientele includes "part-time professional musicians from all over the country who can record for a minimal amount based on the amount of time used." Gordon explains his philosophy on servicing his clientele, "I think it's important that your customers are involved in a way that they're interested in not only themselves, but in the studio. For us, it's important that we treat them just as we would engineers. A lot of studios rush clients in and out as they get more and more work."

Gordon also points out that their specific target market is one of indpendent producers. "We basically rely on word of mouth to reach our target market. In any business the bottom line is word of mouth. If you give quality service and take care to be organized and have the proper equipment, then you'll usually automatically increase your business."

Photo courtesy of Red Brick Recording Studios.

Isolated vocal area.

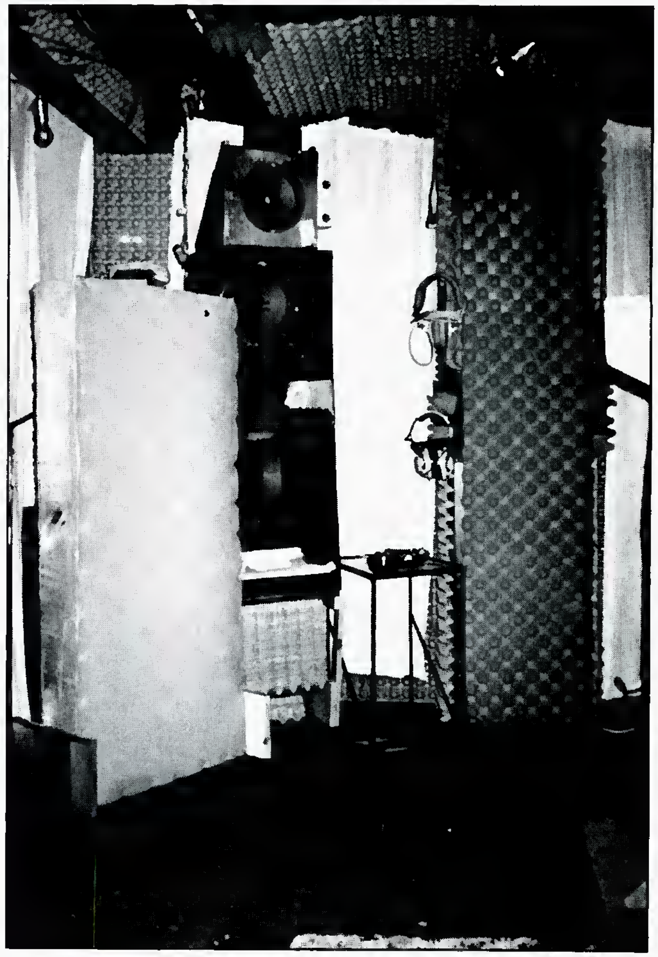


Photo courtesy of Red Brick Recording Studios.

Speaker system in studio.

Gordon also has a few words of wisdom for small studio owners and operators. "We reinvest everything we make right back into it, so as our equipment has increased we've gone up in price. Eventually, when you have a basic idea of how much you bring in within a year and you establish a certain amount of equity in your equipment, you can go to a small business administration or even a bank and they'll give you a loan. You can show them that you brought in a certain amount of business in the last year and what you can expect for the next year."

As for future improvements to be made for the studio, Gordon has one specific alteration in mind (besides going to 16-tracks). "One of the improvements we would like is to have a complete studio information manual available for newcomers. I like the musician to create as much as possible, but a lot of times the musicians are not really aware of exactly what they have to work with and how it can be used. They're not really sure about things like when you have equalization in noise reduction, what effects are available, how they can be used, and how they can cancel each other out."

"I just think that a lot of times when the musician gets into the studio they leave a lot of it up to the engineer and I think the project should be controlled more by the artist. The more they're informed as to what they have and how they can work it, the better off they're going to be as far as creating their work of art the way they want it to be." ■

db TEST
Audio-Technica A-T-RMX64 4-Track Cassette Recorder/6-Input Mixer

Len Feldman

GENERAL INFORMATION

● At first glance, Audio-Technica's AT-RMX64 resembles the familiar "Porta-studio" units offered by another firm. On closer examination, however, you quickly realize that this A-T unit is far more than that. It is, in fact, more like a full-featured 6 x 4 mixing console with an attached 4-track cassette 2-speed tape recorder. There are three basic sections to the unit: The INPUT CHANNELS section, in which inputs are numbered 1 through 6, the "SUBS" outputs which are numbered 1 through 4 and the "TRACKS" section, or recording channels which are also numbered 1 through 4.

You can solo monitor each channel, return bus, or sub output of the unit through a SOLO OUT jack at the rear of the console, when the system is used with an external amplification system. You can also monitor through headphones. The three basic solo monitoring functions available in this mode are: INDIVIDUAL CHANNEL monitoring, SEND/RETURN bus monitoring, and SUB BUS monitoring.

As for the recording capabilities and flexibility of the system, any channel can be recorded on any track by assigning the signal through the sub buses. Monitoring of previously recorded tracks while overdubbing is accomplished by routing the signals from those recorded tracks back through the corresponding numbered channels. You can also "bounce" signals recorded on several tracks to yet another track. This is sometimes referred to as "ping-ponging" tracks. Finally, you can, of course, mix down from a four-track and, by connecting a standard 2-channel mastering deck to the AT-RMX64 you can create an equalized, remixed stereo recording on the external 2-track deck.

There are many more things you can do with this versatile unit; limited only by your own creative abilities and your understanding of the signal path employed in the system. That signal path, is best appreciated by referring to *Figure 1*, which is a block diagram of the the mixer section. Although the tape deck section itself is not shown in this diagram, you can see at what point signals are sent to the tape deck inputs and at what point they are returned to the mixing console from the tape deck.

CONTROL LAYOUT

Over at the left of the console top surface are six banks of channel input controls. Each bank includes controls for two channels of parametric equalization. Not only can the center frequency of the low and high frequency EQ controls be shifted over a wide range, but the shape of the EQ curve can be changed from the usual boost/cut range to a shelving mode (for the high frequency EQ) and a high-pass filter mode (for the low-frequency EQ). Gain control of each input amplifier is provided over a range of 40 dB, with a signal overload indicator nearby to insure against gain settings that might cause distortion. A Line/Mic-Tape switch on the first four channels selects the signal source for that channel, including input from the respective tape track. On channels 5 and 6, AUX is substituted for the TAPE position, and for those channels you can input a signal such as any additional external 2-channel source.

Four SUB ASSIGN buttons on each input channel module are used to assign the outputs of that channel to one or more of the four busses connected to both the sub-group inputs and the tape deck inputs. A "PAN" control allows panning between bus 1 and 2 for stereo mixes, while a channel fader control established the overall output level of the channel. A SOLO button allows selective listening to an individual channel or channels without affecting the channel output or having to have the channel fader up. This is especially useful in sound reinforcement applications for making EQ changes on a particular input channel without disturbing the entire mix. the Channel Solo buttons have precedence over all other SOLO buttons on the mixer, as you can see by referring once more to *Figure 1.*

Controls associated with the four output channels are located to the right of the input channel controls and include SUB OUT faders, additional SOLO buttons, RETURN GAIN controls, SUB ASSIGN controls, and PAN controls. A HEADPHONE OUT pushbutton selects either the SOLO bus or SUB 1, 2, OUT and routes the signal to a switch selector found on the front apron of the unit, where the headphone level control and the headphone jack itself are

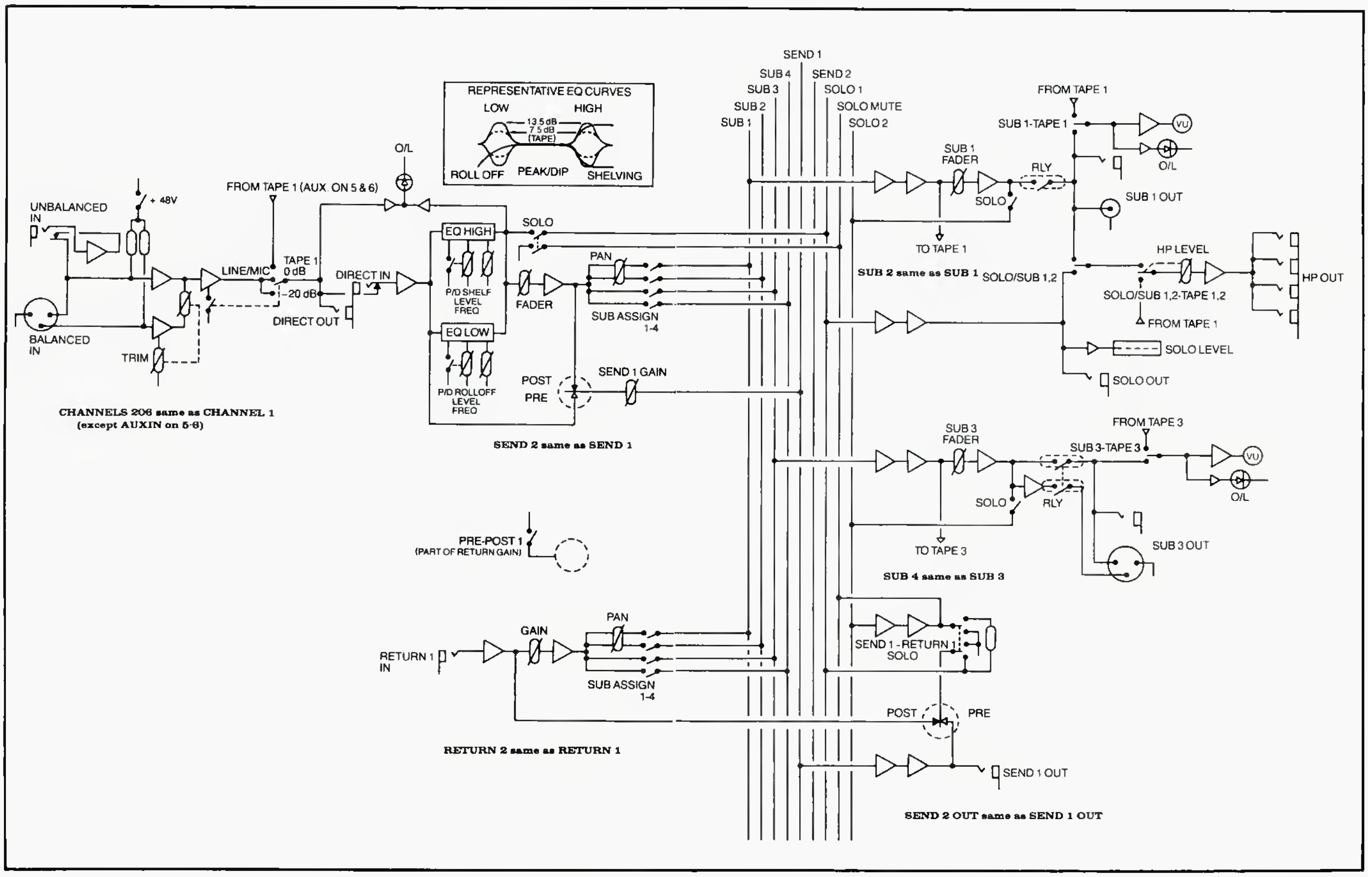


Figure 1. Block diagram of the mixer section.

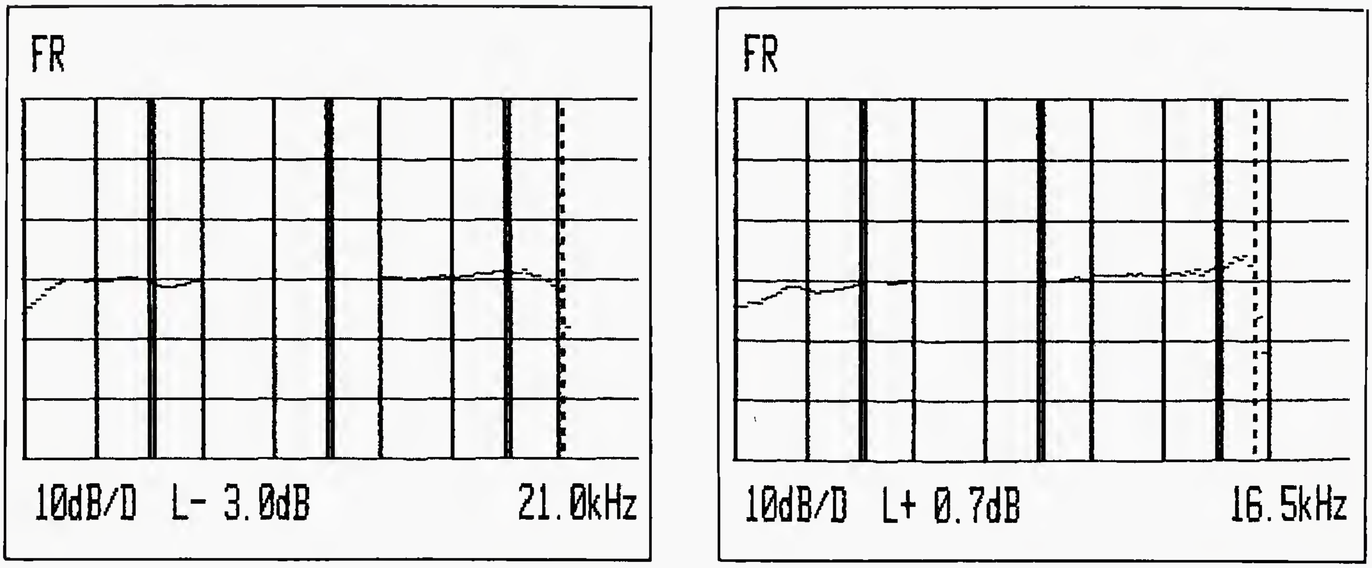


Figure 2. RECORD/PLAY frequency response, without Dolby, at 3 3/4 ips (A) and at 1 7/8 ips (B).

also found. Four VU meters are mounted at the rear of the top surface, sloped for easy viewing. On that same sloped section are switches which assign each meter either to the SUB output or to the TAPE bus, while all the way to the right of the four meters is a bank of LEDs that serves as a level indicator for the SOLO output bus.

All of the controls associated with the tape deck section of the unit, as well as the deck transport itself are located at the right of the top surface of the system. Here record-ready indicators, record level controls and record selector push-buttons for each of the four available tape tracks are found. Dolby On/Off and Dolby B/C switches are to the left of the cassette compartment, as is a combination pitch control/ tape speed selector. This tape deck operates at the standard 1 7/8 ips as well as at a faster 3 3/4 ips speed. At the slower speed, only standard stereo (2-track) recordings can be made, but at the 3 3/4 speed the deck can be used as a true four-track, 4-channel recorder. A three digit tape counter with a reset button and a "memory" rewind button are found just in front of the cassette compartment door, while still further towards the front of the console are the usual tape transport touch buttons with appropriate indicator lights to show when the deck is in the PAUSE mode or in the RECORD or PLAY modes. There's even an indicator light to show you when you have activated the "Memory" return-to-zero counter feature.

The power on/off switch for this unit has been wisely placed on the rear apron, to prevent accidental power turn off during an otherwise frenetic recording or sound-reinforcement session. The rear apron is where you will also find balanced XLR mic/Line inputs as well as unbalanced 1/4-inch phone-jack type inputs, unbalanced AUX inputs for channels 5 and 6, DIRECT IN and DIRECT OUT jacks, unbalanced and balanced SUB output jacks SEND OUT and RETURN IN jacks, a SOLO OUT jack and a couple of additional stereo headphone output jacks.

To aid in the familiarization process, Audio-Technica has augmented their somewhat abbreviated owner's manual with a well-written and concise four-page addendum entitled "AT-RMX64 'How To' Instructions." In view of the diverse applications to which this unit can be applied, the addendum sheet will prove invaluable to you (it did to us), and you should read it and follow the step-by-step instructions which provide examples of just about all of the things

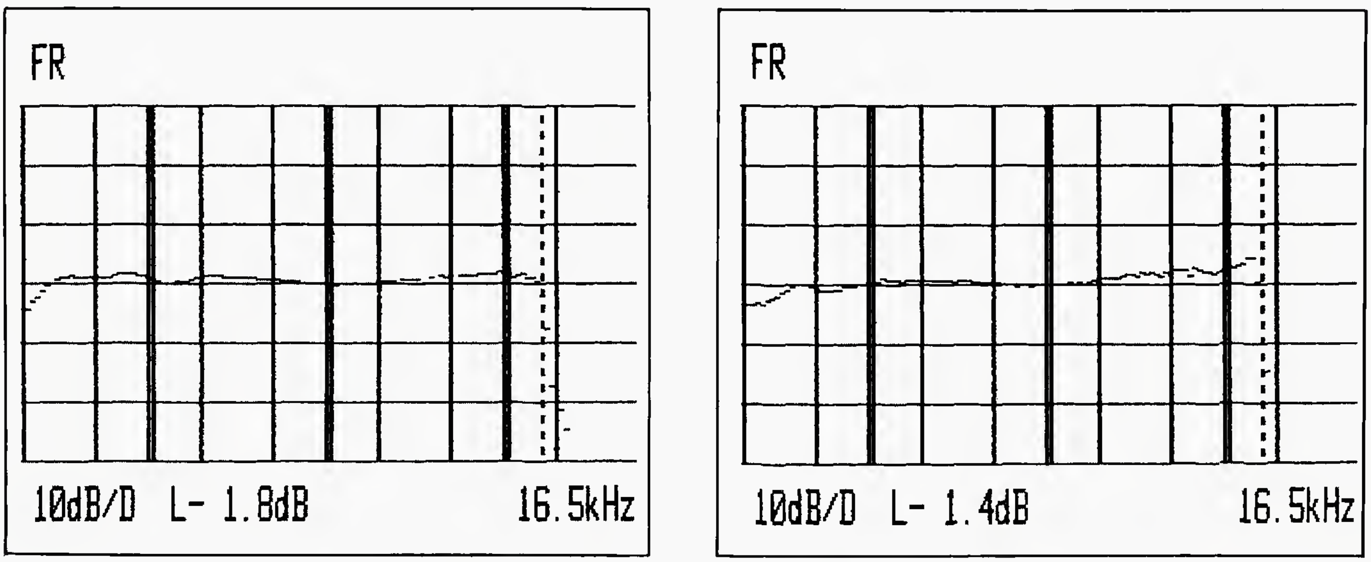


Figure 3. RECORD/PLAY frequency response with Dolby B N/R, at 3 3/4 ips (A) and at 1 7/8 ips (B).

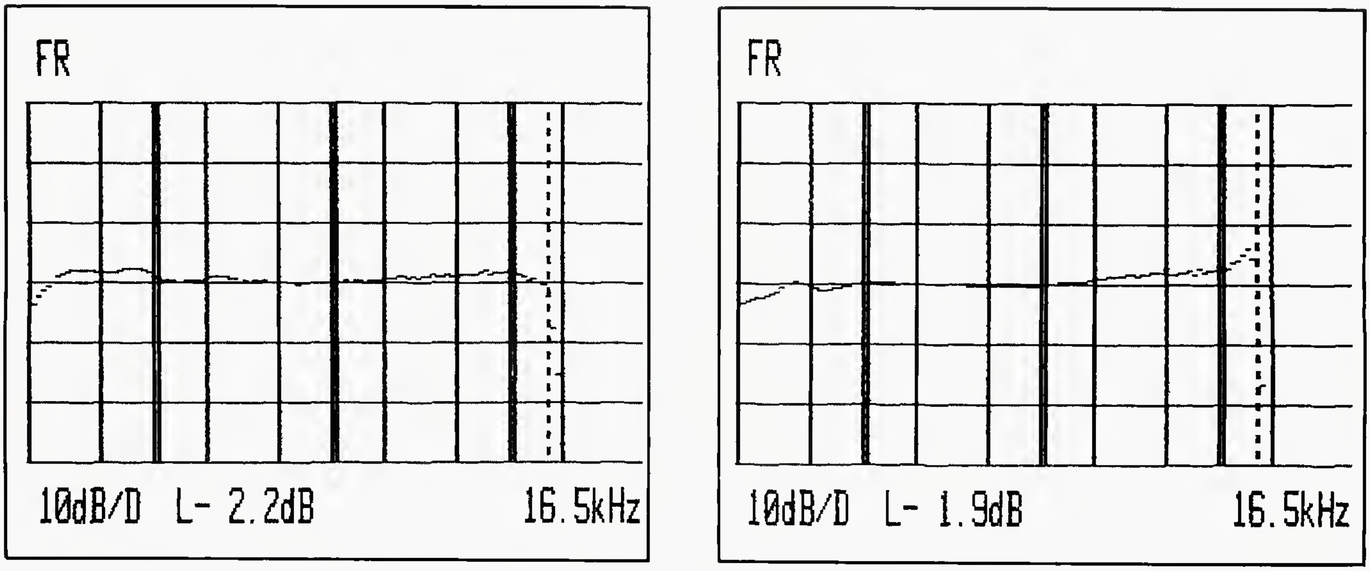


Figure 4. RECORD/PLAY frequency response with Dolby C N/R, at 3 3/4 ips (A) and at 1 7/8 ips (B).

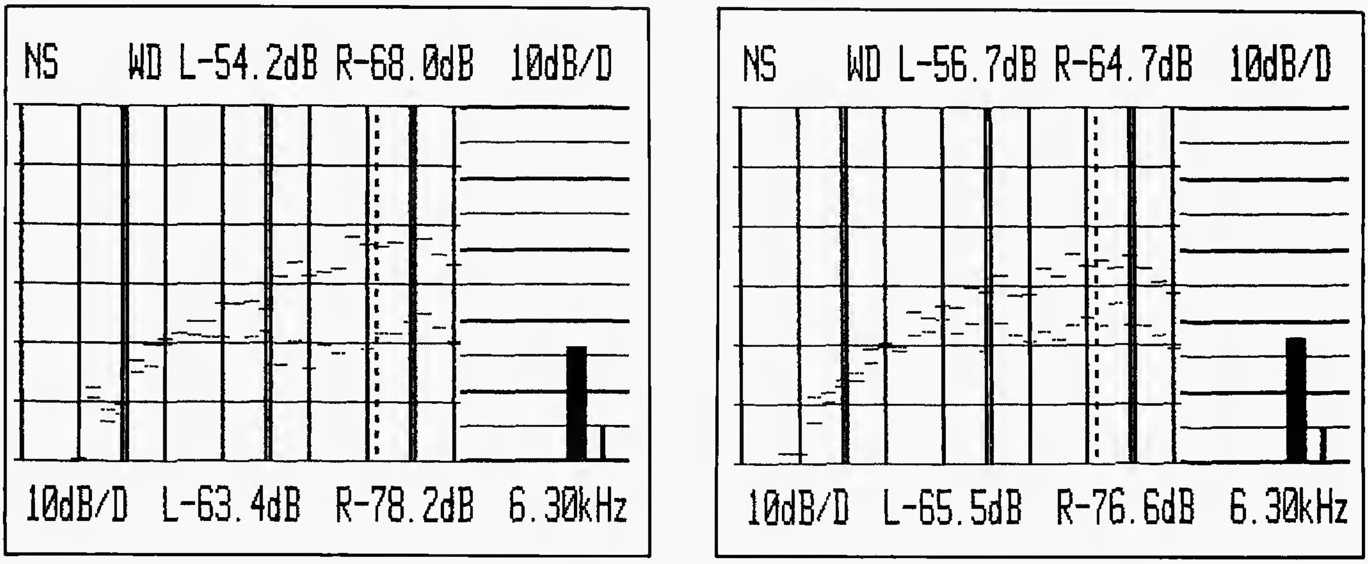


Figure 5. S/N analysis, without and with Dolby B, at 3 3/4 ips (A) and at 1 7/8 ips (B).

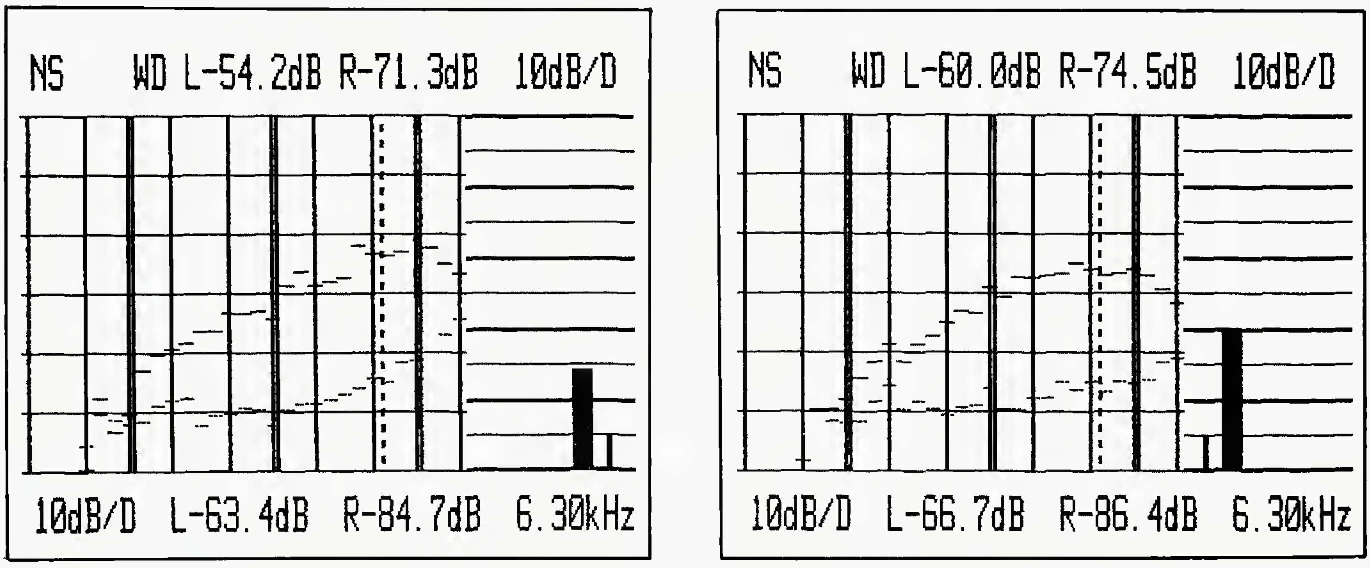
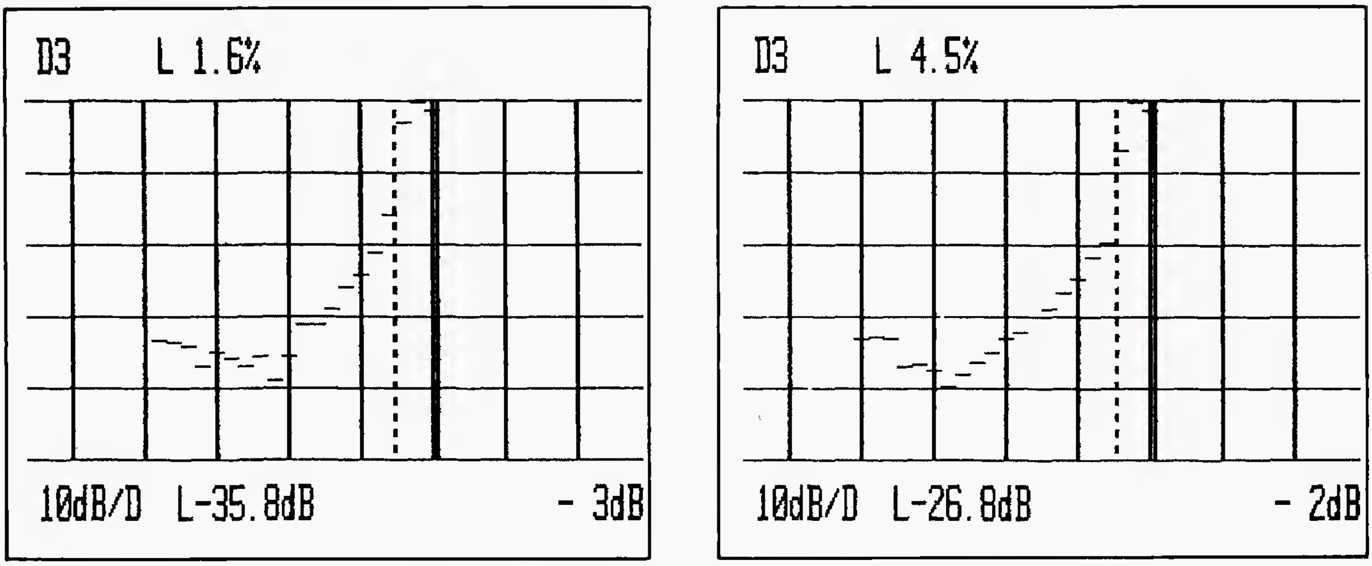


Figure 6. S/N analysis, without and with Dolby C, at 3 3/4 ips (A) and at 1 7/8 ips (B).



Distortion vs. record level (re: +4 dBv) at 3 3/4 ips (A) and at 1 7/8 ips (B).

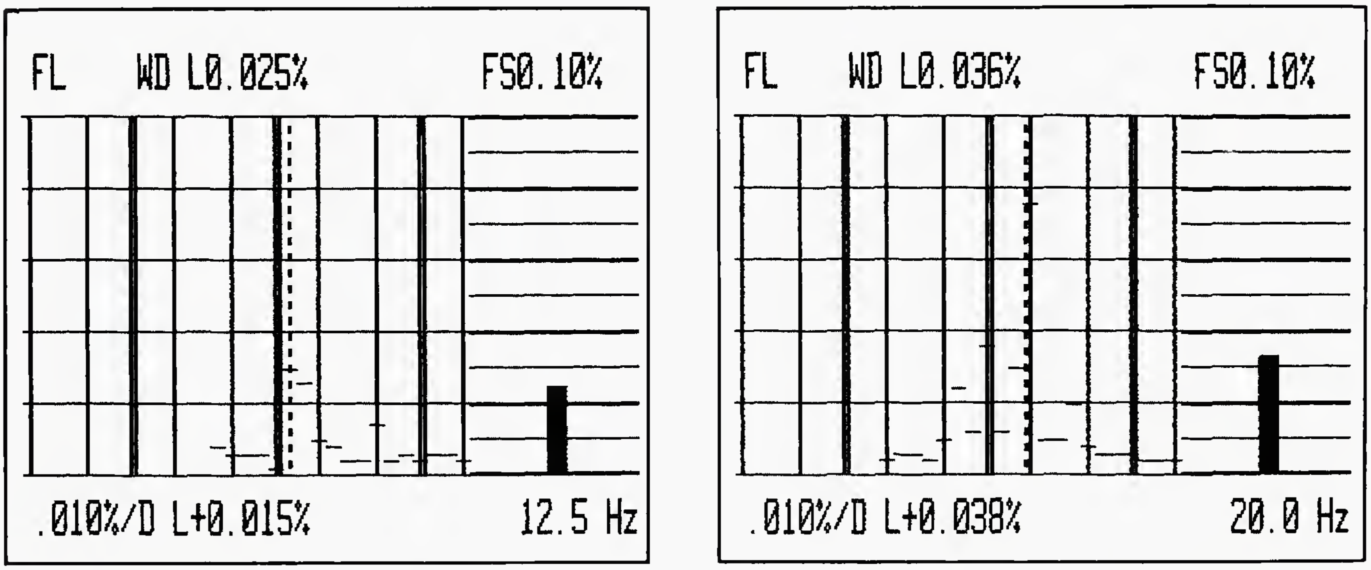


Figure 8. Wow-and-flutter at 3 3/4 ips (A) and at 1 7/8 ips (B).

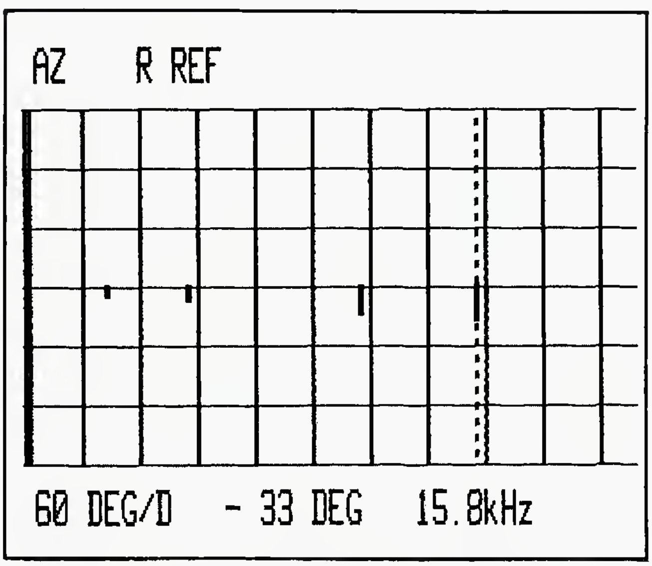


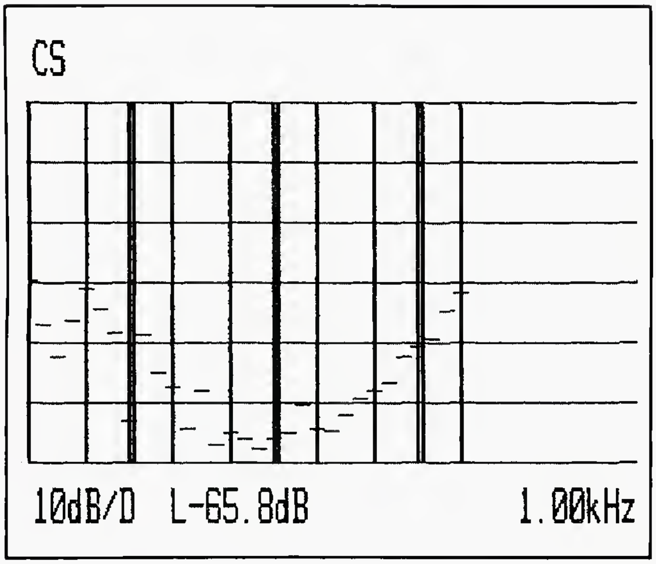
Figure 9. Azimuth alignment accuracy during playback of calibrated azimuth alignment tape at 1 7/8 ips.

you will want to try initially if you purchase this remarkably compact mixer/recorder system.

LAB MEASUREMENTS

A complete table of VITAL STATISTICS, covering manufacturer's claimed performance specifications and our own laboratory test results will be found at the conclusion of this report. The unit was actually tested as two separate products. There were few tests needed to confirm the claimed specs for the mixer section. It performed exactly as claimed, and delivered clean, noise free signals at even lower distortion and lower noise levels than claimed. Hands on use of the mixer section will tell you more about its performance and the smoothness of its controls and switches than any lab tests we could have made.

By far the greatest amount of lab testing had to do with the performance of the tape deck section. Where we thought we might observe a difference between operation at 1 7/8 ips and 3 3/4 ips, results at both speeds were measured. Most of the measurements involved the use of our Sound Technology Model 1500A Tape Tester, from which *Figures 2* through



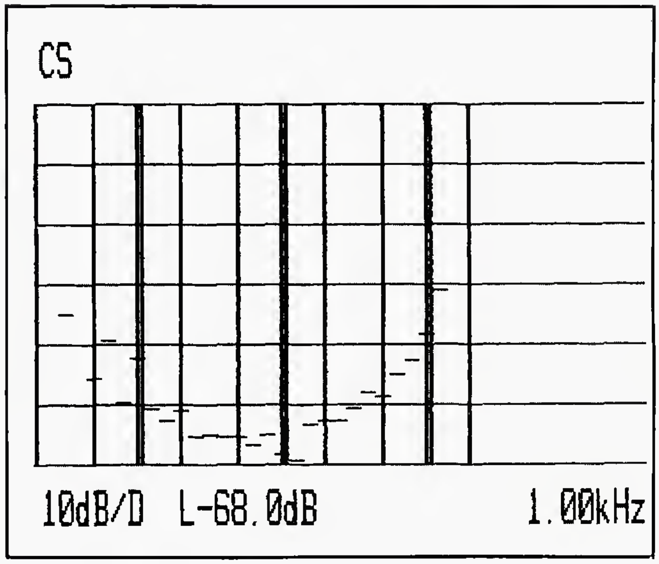


Figure 10. Channel separation vs. frequency at 3 3/4 ips (A) and at 1 7/8 ips (B).

13 were taken directly, using a video printer to make hard-copy prints of the graphs and curves created on the CRT of the tester. Since the Audio-Technica AT-RMX64 is designed to be used only with Type II (Chrome or chrome equivalent) tape, we used Maxell XLIIS tape for all our tape deck measurements.

Figure 2 compares frequency response at the two speeds. The -3 dB point at the faster speed occurred at 21 kHz, while at the slower speed, the -3 dB roll-off point occurred at approximately 17.0 kHz. The cursor in *Figure 2B* is set to 16.5 kHz, where response was still +0.7 dB, since the next cursor setting of the test instrument is at 18.5 kHz, by which time the response was down about -6 dB.

In *Figure 3*, the frequency response measurements are repeated at both speeds, this time with Dolby B turned on during record and playback. Dolby tracking was excellent over most of the audio range, although at the fast speed (*Figure 3A*) you can see that response rolls off slightly earlier than was the case without Dolby. Still, this is about the best Dolby tracking we have seen for any cassette deck, professional or consumer.

In *Figure 4*, frequency response was plotted once more, this time with Dolby C turned on. Again, Dolby tracking was excellent, with a slight rise in RECORD/PLAY response observed at the 1 7/8 speed above 10 kHz. This rise is attributable to minor mis-tracking of the Dolby circuitry at those high frequencies, but it is certainly minimal and nothing to worry about, especially in light of the magnificent noise reduction afforded by Dolby C.

The benefits of Dolby B and C are clearly visible in the graphs of *Figures 5* and *6*. In *Figures 5A* and *5B*, signal-to-noise ratios with and without Dolby B, at the standard and 3 3/4 ips speeds are compared. In *Figures 6A* and *6B* the same sorts of measurements are made, this time comparing Dolby C with "No Dolby." In each case, the dB figures at the top of each graph next to the "L" designation represent results obtained without Dolby, while the numbers next to the "R" notation represent S/N readings with Dolby.

The sole disappointment we encountered during test measurements had to do with third order distortion. For some unexplained reason, our sample showed distortion levels of 3 percent at record levels somewhat below 0 VU on the unit's record meters. This was true regardless of which tape speed we used, as can be seen by referring to *Figures 7A* and *7B*. Possibly a slight misadjustment of record bias is responsible for this. If this is true of all production units our recommendation would be to never exceed the 0 VU meter readings when using the recorder function of the instrument. That level, after all, does correspond to a +4 dBv. Even if you back off a couple of dB during recording, the use of Dolby B or C will still provide you with first generation recordings whose signal-to-noise ratios are as good or better than what you can get from several open-reel decks we've measured over the years that aren't equipped with any sort of noise reduction circuitry.

As you can see by examining *Figures 8A* and *8B*, wow-and-flutter of this tape transport was superb, exceeding even the excellent published specification of 0.04 perecent at either tape speed. Azimuth alignment of the RECORD/PLAY head was excellent too, as evidenced by the results obtained using our standard laboratory azimuth checking tape and plotting the azimuth error of four test frequencies as shown in *Figure 9*. The cursor has been set to 15.8 kHz and the notation of a 33 degree azimuth error does not mean a tape head angular error of 33 degrees. It does mean a phase error of a 15 kHz sinewave of 33 degrees. Translate that to an actual tilt of the tape head (you can go through the

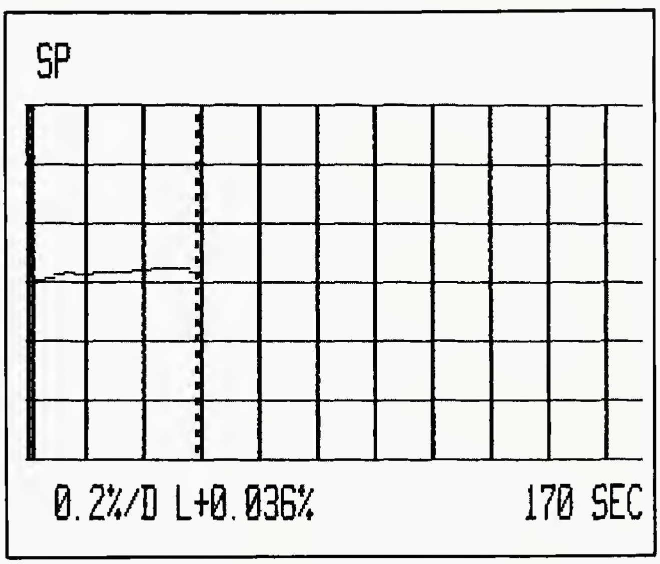


Figure 11. Tape speed accuracy, at 1 7/8 ips.

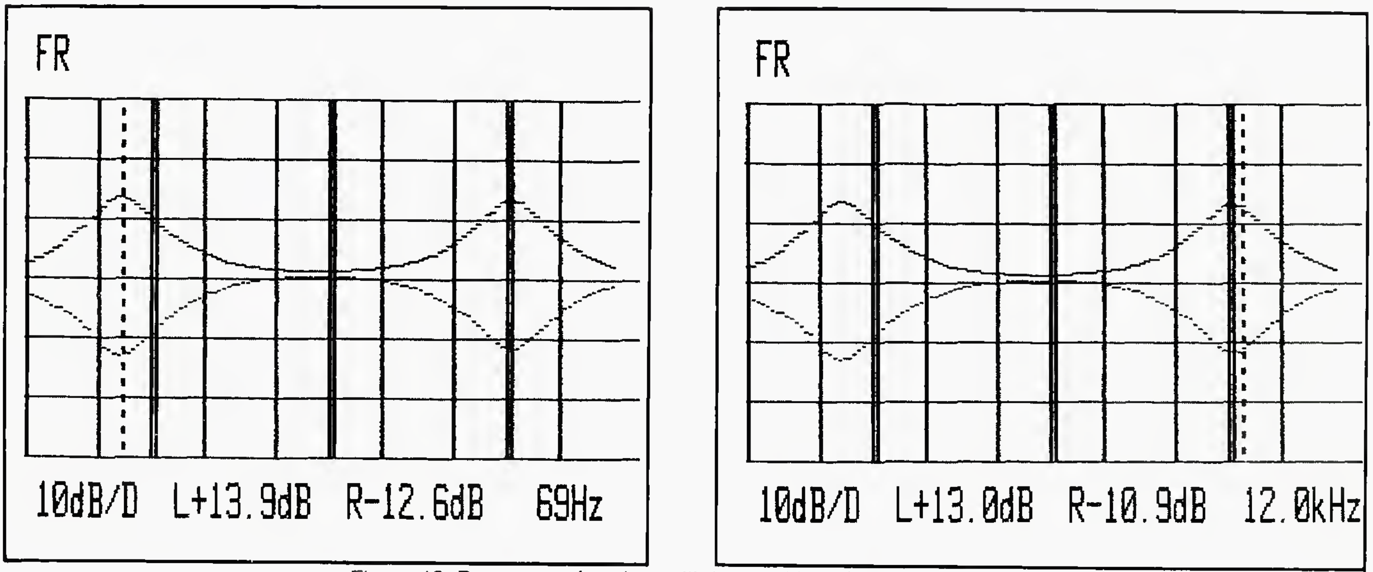


Figure 12. Response of system with low and high frequency EQ controls set to one extreme and EQ level controls next to maximum and minimum positions.

mathematics yourself if you care to) and you'll find that the error is truly negligible. With many cassette tape decks, the 15.8 kHz error we ordinarily read is usually in excess of 120 degrees or even worse. Of course, in terms of a full RECORD/PLAY cycle there will be virtually no azimuth error, since the same head is used for both recording and playback, and the tape traverses the same physical path during record and playback.

Channel separation, plotted as a function of frequency in *Figures 10A* and *10B* was superb, measuring better than 65 dB at the 3 3/4 ips speed and an even 68 dB at the slower tape speed. With the pitch control turned OFF, speed accuracy at the standard 1 7/8 ips speed was excellent, deviating by no more than 0.036 percent over a continuous run time of three minutes, as plotted in *Figure 11*. Results were almost as good at the higher tape speed, which we calculated by recording an accurate 6000 Hz signal at that speed and then measuring the frequency during playback.

We used our response plotting function of the Sound Technology Tester to check out the range and flexibility of the EQ controls. In *Figures 12A* and *12B* the low-frequency EQ control was set to its most counter-clockwise setting while the high-frequency EQ control was set to maximum clockwise. Then two plots were taken; the first with EQ level controls set to maximum (maximum "boost"), the second with controls set for minimum (maximum "cut"). The plots in *Figures 12A* and *12B* are identical. Only the electronic "cursor" has been moved to show, in one case, the maximum cut and boost at the treble end of the spectrum and, in the other case, the maximum cut and boost for the low-frequency EQ controls.

Figure 13 is similar to *Figure 12*, except that in this case the EQ frequency controls have been moved to their opposite extremes to show the highest center-frequency of the "Low EQ" control and the lowest center frequency of the "High EQ" control. Finally, in *Figure 14* we see what happens to the response at the bass end of the spectrum when the "Low EQ" controls are used in their alternate, "high pass filter" mode.

COMMENTS

The Audio-Technica A-T-RMX64 is, without a doubt, the

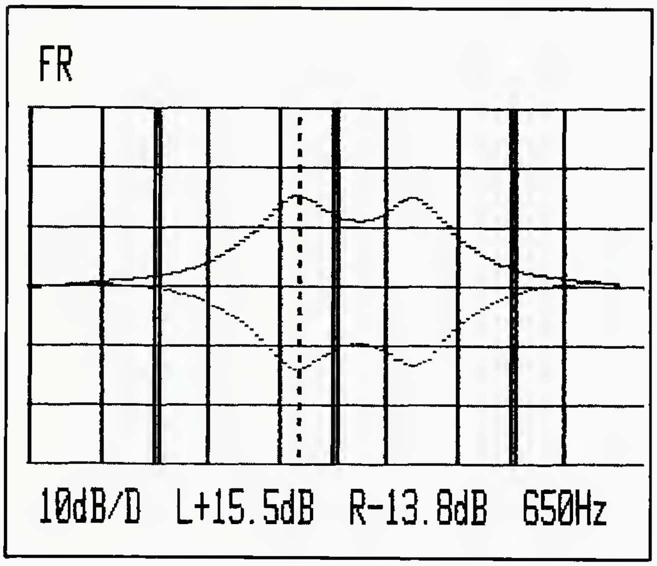


Figure 13. Response of system with low and high frequency EQ controls set to opposite extremes from that shown in Figure 11 and EQ level controls set for maximum and minimum positions.

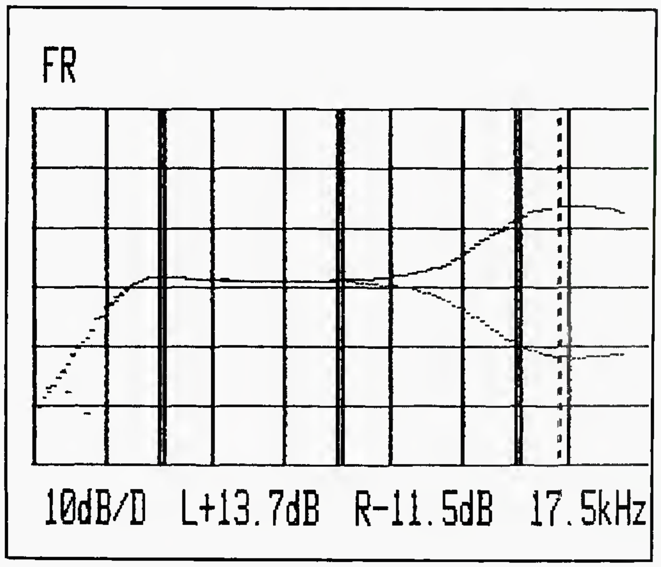


Figure 14. Action of the "Low EQ" control when it is used to provide high-pass filter action.

most flexible and compact combination mixer/recorder we have seen. It is so versatile that as you use it, you are likely to discover practical uses for it that neither Audio-Technica nor we can envision. Besides the obvious uses as a personal studio and sound reinforcement console, you will find it equally useful in the audio-visual production room, as an auditioning deck, and in countless other situations where a combination of portability and professional features is required. As we said at the outset, applications will be limited only by the knowledge, experience, and imagination of the end user. We tried our hand at using the instrument for recording as well as sound reinforcement applications. In the former application, punch-in/punch-out recording was flawlessly accomplished, without so much as a click or pop during playback. The combination of channel and sub-group controls was a welcome addition to this console, allowing us to adjust several input/output levels simultaneously using a minimum of controls. All in all, the designers of this mixer/recorder have obviously spent some time in the field and in recording studios. They have somehow managed to come up with all of the features that many recording engineers and technicians have hoped for in a portable/personal unit of this type. Once this unit gets out into the field in greater numbers, we suspect that Audio-Technica will have a tough time keeping their dealers supplied with the A-T-RMX64.

VITAL STATISTICS

4-TRACK CASSETTE RECORDER/6-INPUT MIXER

MAKE & MODEL: Audio-Technica A-T-RMX64

MIXER SECTION

SPECIFICATION	MFR'S CLAIM	db MEASURED
Line/Mic Amplifier Gain	58 dB	59 dB
Total Available Gain	72 dB	73 dB
Aux/Return input Gain	0 dB	Confirmed
Send/Sub output gain	0 dB	Confirmed
Channel & Sub. Overload Indicator	+17 dBv	+17 dBv
Max. Output before Clipping	+17 dBv	+17 dBv
(Balanced Mode)	+21 dBv	N/A
Max. Phone Output	1.2 W/8 ohms	1.2 W/8 ohms
Hum and Noise	-122 dB Equiv.	Confirmed
THD	0.05%	0.02%
Frequency Response	20 to 20 kHz,+/-1.5 dB	20-20 kHz,+/-1.0 dB
EQ Range	60-1.5 kHz (Low)	69-650 Hz (Low)
	600-10 kHz (High)	500-11 kHz (High)
EQ Boost/Cut Range	+/-13.5 dB	See Figs.11&12.

RECORDER SECTION

SPECIFICATION	MFR'S CLAIM	db MEASURED
Tape Output Level	+4 dBv 0 VU	Confirmed
Frequency Response (Rec/Play)	40-15 kHz, +/-3 dB	See Figs. 1-3
Bias Frequency	85 kHz	Confirmed
S/N Ratio (3 3/4 / 1 7/8 ips)		
Dolby Off	55 dB/N.A.	54.2/56.7 dB
Dolby B On	64 dB/N.A.	68.0/64.7 dB
Dolby C On	68 dB/N.A.	71.3/74.5 dB
THD at 0 VU	1.5%	See text
Channel Separation	60 dB at 1 kHz	65.8 to 68.0 dB
Tape Overload Indicator	+8 dBv	Confirmed
Tape Type	High Bias (Type II)	Confirmed
Tape Speed Accuracy	N/A	+0.036% 1-7/8
Pitch Control	+/-15%	Confirmed
Wow-and-Flutter	0.04% WRMS	0.025% 3-3/4 0.036% 1-7/8
Fast Forward/Rewind Time (C60)	80 sec.	68 sec.7Y
Number of Motors	3	Confirmed
Number of heads	N/A	2

GENERAL SPECIFICATIONS

SPECIFICATION	MFR'S CLAIM	db MEASURED
Dimensions(W x D x H in.)	23.2 x 20.4 x 5.4	Confirmed
Net Weight	48.5 lbs.	Confirmed
Power Consumption	65 watts	58 watts
Price	$1695.00	

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WABC77

Ralph W. Hodges

Something Old, Something New

Tony Bennett delves into digital recording.

When tony bennett and his people decided to record *The Art of Excellence,* (bringing their album count on CBS alone to something over eighty), they searched with vigor and inspiration, and they found Ray Charles, who sat in on—no, let's make that "took over"—one of the final cuts. Another thing was a museum-worthy collection of ancient Telefunken microphones, vacuum-tube preamps and all. A third was digital, in its most advanced form. In fact, digital was at the core of the project from the start, and in effect, releasing initially on CD rather than LP became something of a priority.

Photo by Annie Leibovitz.

Says Danny Bennett, Tony's son, who shares production credits with Ettore Stratta, "There was no reason to put anything in this album that wasn't absolutely the best—no reason to rush to market or make hasty decisions, because Tony will sell as well next year as he does now. Digital seemed like (it could be) either the best or the worst, depending on whom you talked to, so we had to give the choice some very careful consideration. In the end, we were swayed by the way average consumers are accepting digital. They love it, and it doesn't seem to matter whether they're passionate and sophisticated about music or not. The response is instantaneous, and is almost always very positive. When you have something like that going for you, the carping of some professionals and even some audiophiles ceases to mean very much."

Fine. But how do you hedge that bet? In this, the Bennetts were admirably resourceful. "It's most severe critics call digital cool, clinical, calculated, and even harsh and monodimensional. Our experience, now that we've gained some, has been otherwise, but our thinking was to counter the risk with an exceptionally warm room and exceptionally warm microphones. That's how we wound up in the big studio at Olympic (London), even though we knew we wanted to mix at CTS (also London—well, Wembley, which is close enough) because of the Neve DSP-1 console. Olympic had the right room, however, and it also had the right mics. Lord, did it have mics."

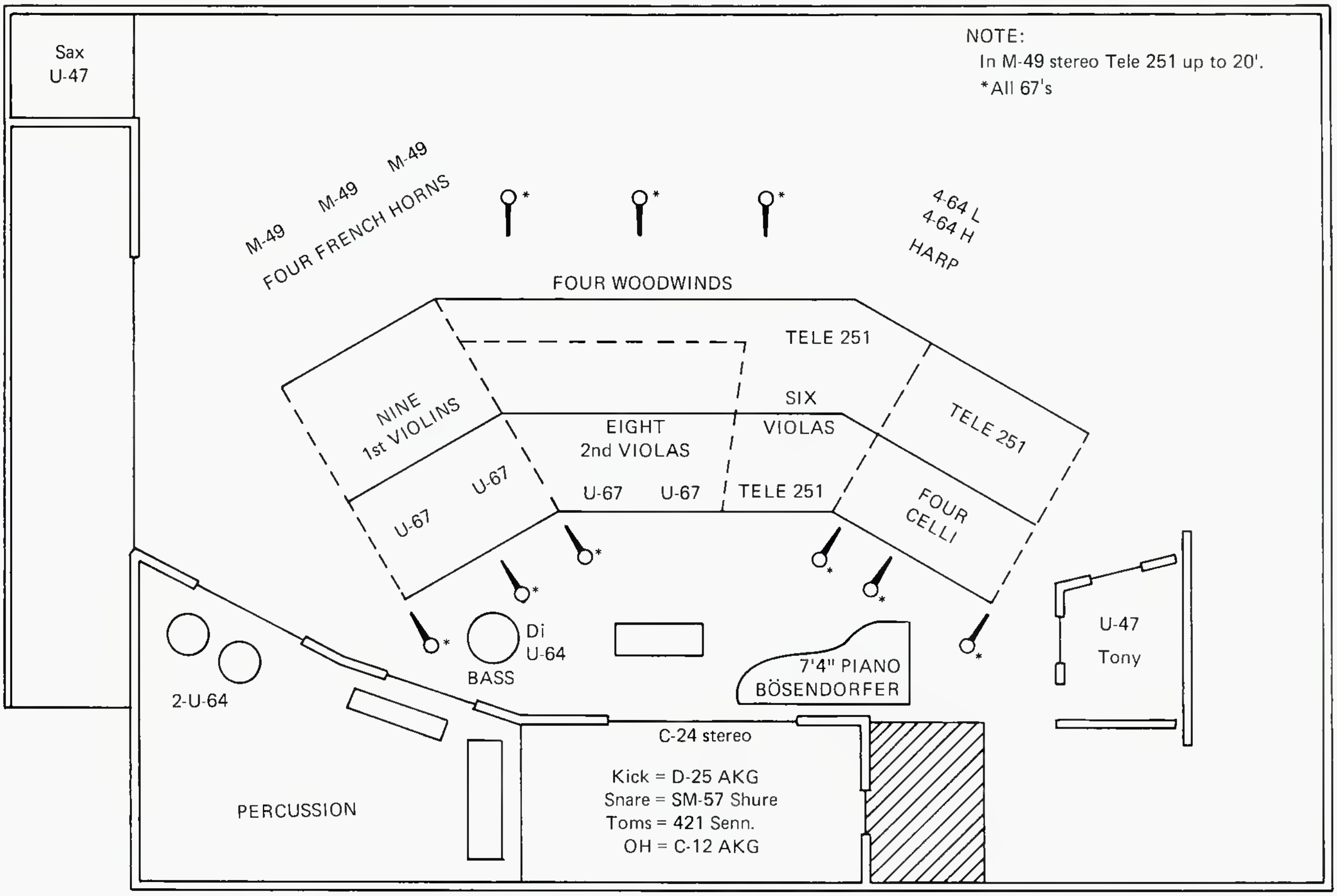
Although the plan (see *Figure 1*) doesn't immediately suggest it, chief engineer Paul Mufson calls the pick-up pretty much basic stereo, captured by feeds from a C-24 behind the conductor and an MS configuration of an M-49 and Telefunken 251 in the back of the room, twenty feet or so up. The track sheet, designed around a Sony PCM-3324 recorder, tells most of the story. The sax got a touch of reverb (PCM-6) now and then, the harp was panned low and high, and a track was reserved for a direct-feed bass. Beyond that, everything was pristinely straight-forward, with a minimum of EQ and no compression whatever. The room and the microphones, which were essentially aranged in two vertical tiers to exploit the exceptionally high ceiling, did it all, with a result that is called "a real community of sound."

TIGHT SHIP

Two days sufficed to do the basic recording, which is certainly a credit to the efficient way the Bennett operation has learned to work. Bennett maintains his own New Jersey studio and a trio of musicians, and all possible preparations, including arrangements (done here by Jorge Calandrelli), are taken care of with those resources. At the last moment the band is added—in this case the forty-five piece UK Orchestra Ltd.—for a carefully plotted series of sessions that are run live in most cases.

Or so you hope, anyway. With *The Art of Excellence,* the new technology raised a few matters that no one was quite sure how to handle at the time. "Digital is really astonishingly revealing," says Dan Bennett. "People were going nuts trying to figure out where all these subtle sounds were coming from—real sounds, performance sounds, but sounds you're just not used to having wind up on the final tape. We had clacks from Tony's tongue, fingernail strikes on wood instrument bodies, and of course, we could hear the mic preamps above the noise floor of the recording system. We had to resort to some unconventional repositionings of the orchestra to keep the incidental noise build-up from getting out of hand, and then, during the mix, we had to decide what

Ralph W. Hodges is a freelance writer in the audio industry.



Mic placement used during most of the recording.

to leave in and what to try to take out. In the end we left most of the clicks and creaks in, really because they did sound so incredibly real, vivid, and intimate. We'll have to see whether that was the right thing to do."

BOARD WITH IT ALL

The mix at CTS occupied four days of time with the new Neve DSP-1 digital console, mixing to the Sony PCM-1610, with editing (of which there was only a bare minimum) implemented by the DAE-1100. Mufson found the Neve board very user-friendly, but points out that it requires a user to think in terms of a vertical rather than a horizontal layout, which some people have trouble with. "With the narrow-band EQ available, we were able to do punch-out/punch-in glitch removal that defied anybody's detection, and the speed with which we were able to get back to elements of the previous day's work for touch-ups and afterthoughts had us all delighted. The exciting thing is that, with the Necam interface, all this will be completely automated soon, and since every function on the console is digital, there is no limit to where this board can go. You'll be able to get back to exactly where you were hours or even days ago within ten *sounds*."

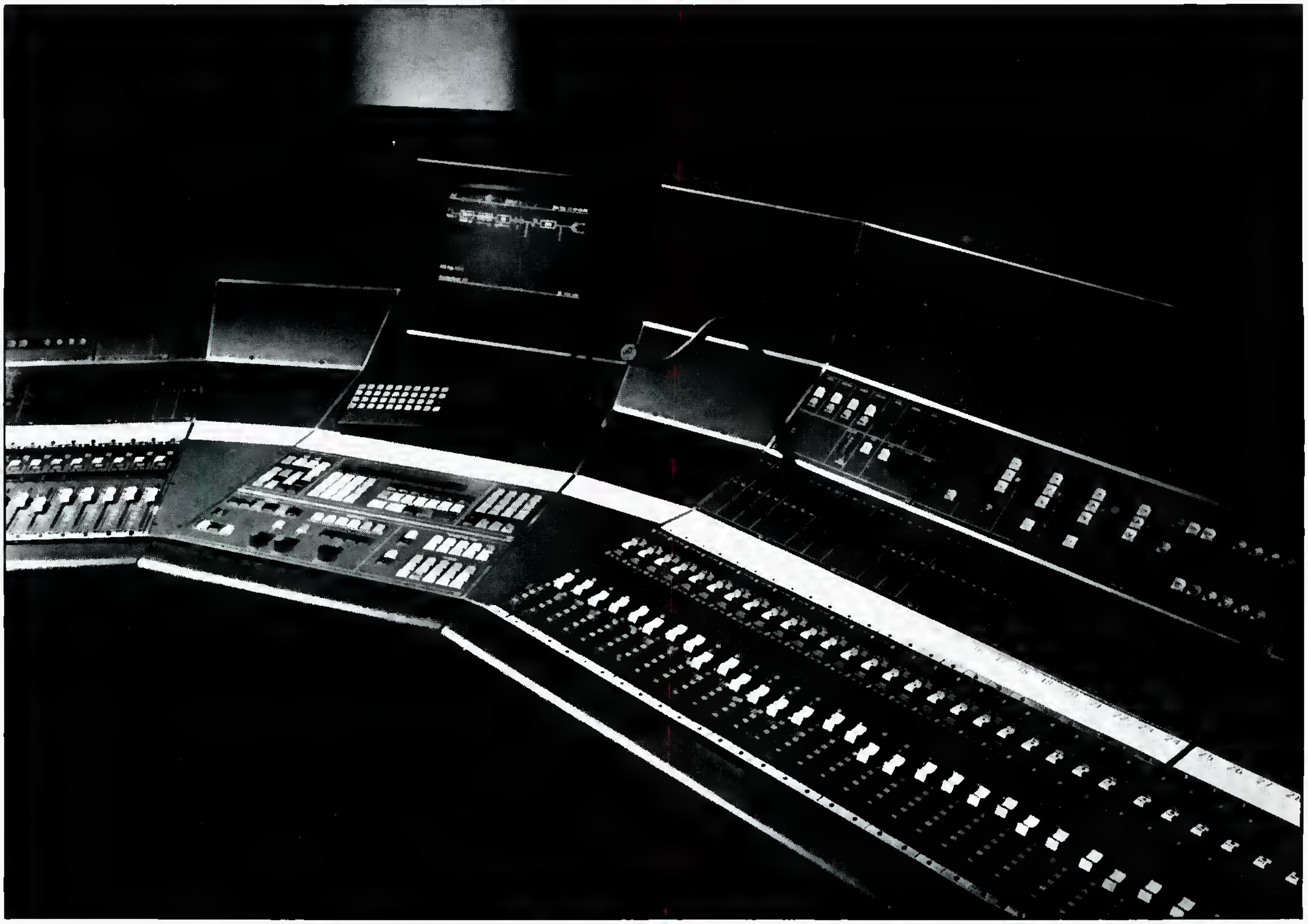
Danny Bennett, who had to approve the rental bills, is no less enthusiastic. "Well, yes, very expensive on the face of it. But when you think of what you save in time and energy, and the amount of creative concentration you're able to keep fixed on the music rather than on the mechanical chores of manipulating it, you begin to feel very comfortable about the expense. We'll go back again, no doubt about it. We can't imagine returning to a conventional analog console for *any* big project now that we've learned what we've learned. Even for the relatively straightforward work we did on this album, it doesn't seem to make any sense to go back."

AS FOR THE PRODUCT...

The Art of Excellence was mastered for CD by Mike Ellis at DADC in Terre Haute, and again with very little fuss. "We did a few things in Terre Haute to fine-tune the master, but I doubt they're anything anyone will ever hear.

1 PIANO	2	3 BASS	4 BASS DRUM	5 KIT	6	7 SNARE	8 VOCAL
9 1	10 2	11 3	12 4	13 HORNS	14 HARP	15 WOOD WIND	16
17 49 C24 FRONT	18 49	TO BE DECIDED 8 C24 BACK M	8 S	21 PERC.	22	23 SOLOS	24 BASS D.I.

Basic track allocation used during the recording of most of the tracks.



Neve DSP digital recording console.

We were just being picky." (*Figures on inital production run quantities were not available when this was written.*)

The LP version, deliberately scheduled for release a month after the CD was handled by Bob Ludwig at New York's Masterdisk, and it was done DMM. According to Bennett, there was no special processing whatsoever of the digital master for LP transfer. "Ludwig decided he could cut the tape as is, which was not something we were exactly expecting, since we did lay on the levels rather lavishly, and there was no compression in the chain. He didn't even mono the bass. So score one for DMM." Well, let's hope so. Test pressings had just arrived when this interview was conducted, and no one seemed about to play them with strange reporters lurking around. However, a visual inspection turned up no problem.

And so, it would seem that another segment of the musical mainstream has been irretrievably seduced by digital madness. It's a pity that Mabel Mercer, to whom the album is dedicated, never got her chance. That would have been especially interesting.

Bob Ludwig

Mastering Tony Bennett's Newest Record

Find out what went into the mastering of The Art of Excellence.

I HAVE HAD the good fortune of meeting and working closely with many artists in the pop, jazz, folk and classical fields. Yet, when Tony Bennett came to the studio to supervise the mastering of his album I admit that I was very excited. I actually phoned my parents that evening and said, "Guess who I worked with today?" I hadn't been that enthusiastic in some time.

When I first heard that Bennett's new album was being recorded at CTS in London on a totally digital console I was very anxious to hear it. As you may know, these multitrack digital consoles don't yet exist anywhere in America.

Bob Ludwig is the mastering engineer at Masterdisk Recording Studios.

Masterdisk is going to be among the very first studios in the US to receive a totally-digital mastering console specifically designed for compact-disc mastering, so naturally I was very interested in how this new technology would sound. I'm happy to say that it sounds spectacular! The large string section sounds very natural and open. Bennett's voice was beautifully recorded without the use of compressors anywhere in the chain. There are no negative "digital" artifacts in the sound. The only other project from this console I had worked on until now was a Duran Duran hit, and it was difficult to tell with all the sampled sounds and effects just how good this desk was!

As we began the mastering session I could tell that the tape sounded wonderful and it wouldn't need much done to it artistically. One of our "EQ" decisions was to decide on which digital processor to reproduce the tape. Yes, folks, all digital recorders are not created equal! The master was recorded on a Sony PCM-1610 system. Masterdisk has three kinds of Digital-to-Analog converters and electronics to choose from: the standard Sony PCM 1610, the Sony 1630 with oversampling and new filters and a digital converter made by Studer, the DAD-16. We chose the one that sounded the most musically appropriate to this material and proceeded from there. Parenthetically, I should note that with the compact- disc version of this record a similar difference in sound is created by each purchaser's own CD player. I have read, with amusement, articles that claim there isn't much difference between CD players. I can assure you that there are indeeed such differences. Every time someone tells me he was comparing a CD I had mastered against the same record I had done and tells me the record sounded much better, I ask if they own a particular kind of CD player. They have so far, always said yes, and I request they try a different one! One of the most difficult things about mastering this record was the dynamic range. Bennett has a voice with frankly surprising dynamics. Years and years of hearing his voice on other recordings or on TV with both Bennett and the TV sound engineer "working" the mic had lead me to believe that he belonged in the crooner category. I was quite mistaken. To do a proper job I had to treat the recording as if it were a classical record and do a test cut of the entire record. The ultimate transfer level was determined by doing a test of the loudest parts (and they were very loud) and playing them back on a phonograph pickup that would be found in an average stereo. I tried different levels until it sounded clean. Basically, all the levels then fell naturally from this point.

With so much care and musicianship lavished on this project it seemed natural to continue the search for technical excellence by using the new direct-to-metal cutting process. This process, invented and licensed by Teldec in West Germany, cuts into raw copper instead of plastic. Normally I cut into an acetate lacquer. Lacquer material is a constant source of headaches as the quality changes not only from manufacturer to manufacturer and not only from batch to batch, but even within the same stack of lacquer blanks! Often a lacquer will be rejected for a "streak" or piece of foreign material that will be picked up momentarily which causes an increase in hiss on one channel or another. Often the lacquer is rejected for excessive ticks and pops in the formulation.

Once a good lacquer is cut it must be raced to the electroplating plant to be plated. Each hour that passes between the cutting and plating introduces more groove "echo" into the lacquer. Groove echo is where the music can be heard before or after it is actually supposed to occur. The

material on Bennett's record, which is so transparent and exposed, can cause disastrous problems in this area. The electroplating plant then cleans the lacquer and sprays it with a fine layer of silver. The slightest mistake here will cause ticks and pops in the final record. A negative of the record, (called a Father), is made. This has ridges instead of grooves and cannot be played. Only one Father can usually be made from the extremely fragile lacquer master. Then, a Mother, (or positive) is plated from the Father and many Stampers (negatives) are made from this. It is also from these that records are pressed (similar to a waffle iron with Side A and Side B stampers heating and forming the vinyl "batter").

The Direct Metal Mastering lathe cuts directly into a copper foil, creating an instant Mother. This completly by-passes all the myriad problems with the lacquer material. No problems getting it into the electroplating bath quickly or with the critical silvering and pre-plate steps and the making of the Father and Mother.

There are additional advantages besides the manufacturing steps. The background noise is reduced about 10 dB. Because copper is so much harder than acetate, there is no lacquer "springback" which yields usually 15 dB less groove echo while improving transient response. Sibilance that can cause tracking problems with normal cutting usually cuts and plays back with no distortion on copper. The inner diameters of all records have a natural roll-off of the highest frequencies which is greatly lessened with the DMM record. Whereas a conventional recording might have 16 kHz rolled-off 4 dB the DMM will still be uniform (no roll-off) at this point. This is one of direct-to-metal's most easily heard advantages. In addition, depending on the program material, one to two more dB level can be added to the copper transfer compared to the plastic. This extra dynamic range can result in either very long forty minute sides (six minutes more information per disk than a CD can carry!) or hotter levels for a given timing. Early DMM disks from Europe seemed to have been lacking in low frequencies, but recent modifications to the system now make the bass superior to what has been available on any cutting system. Indeed, there is still 20 dB of seperation at 10 (ten) cycles! The sound of the DMM records is much closer to that of a good Compact Disc. There is that "extended bass" quality that is so satisfying on CD.

Bennett's record was both a challenge and a rewarding musical experience. The test pressings came out beautifull and I'm sure there will be many others in his audience wh will savor his search for technical excellence.

DMM is a registered trademark of TELDEC Schallplat Gmbh, Hamburg, West Germany.

Bruce Bartlett

Recording Techniques

Confusing Mixer Functions

• In this month's installment, we'll try to explain some confusing mixer functions. You may want to review the articles on mixing-console theory and operation in the October '82 and November '82 issues of *Modern Recording & Music.*

A REVIEW OF THE BASICS

Before getting into details, let's briefly explain what mixers do. A *mixer* combines or mixes several different signals into one or more composite signals. You plug several microphones or other sources into the mixer *inputs.* Next you adjust the volume of each microphone with a volume control (a rotary *pot* or sliding *fader*). The mixer then blends these signals into one or more output channels to feed a tape recorder or sound-reinforcement system.

Figure 1 shows a block diagram of a simple mixer. Note how the level of each input signal is adjusted independently by a fader. Then the signals are combined by a *combining amplifier*—also called *Summing Network* or *Active Combining Network (ACN).* Finally, the level of the composite signal is adjusted by a *MASTER* volume control.

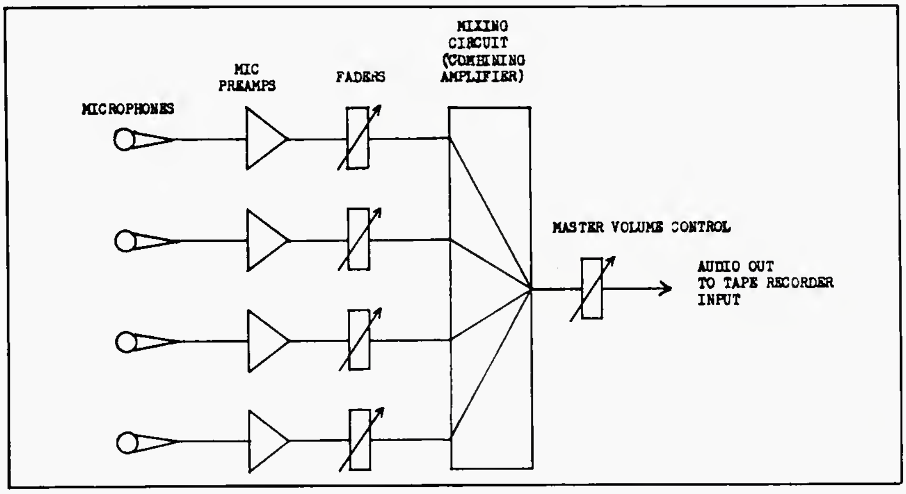


Figure 1. Signal paths in a typical mono mixer.

Mixers are specified by the number of inputs and outputs they have. A 6 x 1 mixer has six inputs and one output. A 16 x 4 x 2 mixer has sixteen inputs, four submixes (explained later) and two main outputs. There may also be connectors for external equipment such as reverb units and monitor amplifiers.

Figure 2 is a block diagram of a stereo mixer. A *pan pot* on each input adjusts how much mic signal goes to Channel 1 (left) and how much goes to Channel 2 (right). This panning action places the sonic image of each instrument wherever desired between a stereo pair of loudspeakers. With pan pots, you can locate an instrument at the left speaker, right speaker, or anywhere on a plane in between.

Some mixers include CHANNEL ASSIGN buttons to route each mic to Channel 1, Channel 2, or both.

More elaborate mixers provide extra control over the sound quality. They might include such features as the following:

Equalization (EQ) or tone control for each microphone. With EQ you can make an instrument sound more or less bassy, more or less trebly, by boosting or cutting certain frequencies.

Input Attenuation (pad): This reduces the voltage of input signals to prevent input-overload distortion. An LED flashes when preamp clipping occurs.

Echo bus (aux bus, effects bus): A channel for sending and receiving signals to and from an external signal processor such as a reverb unit. By doing so, you can add a sense of room acoustics or spaciousness to an otherwise "dry" track. Some mixers have more than one aux bus, allowing you to add a variety of special effects.

Figure 3 is a block diagram of a 2-channel mixer with the features just described.

The set of controls that affect a single input is called an *input module.* Each input module in the mixer has an identical set of these controls. *Figure 4* shows a simple input module with the features just mentioned. *Figure 5* shows several modules in a row; they comprise a simple mixer. Input modules in a mixer are numbered consecutively from left to right.

THE MIXING CONSOLE: AN ASSEMBLAGE OF MIXERS

Mixing consoles (also called mixing boards, recording consoles, or mixing desks) are large, complex devices that include several mixers within a single chassis. Each of these smaller mixers operates independently; each has its own function. They are the program mixer, monitor mixer, effects mixer, and cue mixer (all explained shortly). The output of each small mixer is called a *bus* sometimes spelled *buss,* or *mixing position*). A bus is a channel in the console.

The *program mixer* creates the mixes that are recorded onto tape tracks. Its inputs are the input modules; its outputs are the program buses. Using the channel-assign buttons and faders on the input modules, you create various *submixes* to record on different tape tracks. A submix or *group* is a small preset mix within a larger mix, such as a drum mix, keyboard mix, vocal mix, etc.

For example, you might want to assign several drum microphones to a single channel (program bus). All those mics are mixed to one channel. You control the overall level of all those mics simultaneously with the *submaster* fader (*bus master, group master*) for that channel. In other words, you create a *submix* of the drum kit within the overall mix.

The *monitor mixer* creates the mix heard over the monitor speakers. This mix is independent of any mixes being recorded on tape.

The *effects mixer* includes the effects send (echo send) circuits. Its output is the effects bus (echo bus). This bus feeds an effects device such as a reverb unit or digital delay. The output of the effects device returns to the "echo-return" or "effects-return" connector in a program-bus module, where the echo signal blends with the program-bus signal.

The *cue mixer* blends pre-recorded tape tracks and live microphone signals into a mix that is sent to the musicians' headphones in the studio. This mixer's input levels are controlled by CUE pots; the output is called the "cue bus."

Some consoles include a *stereo mixdown bus*: the final 2-channel mix that is recorded onto a 2-track tape. Other consoles just use program buses 1 and 2 for the stereo mix.

Figure 6 shows the signal paths in a typical multi-channel console during recording. Note the various submixers just described.

MIXING-CONSOLE SECTIONS

Mixing consoles can be divided into three main sections: INPUT, OUTPUT, and MONITOR.

The INPUT section is a series of input modules. Each module includes an input connector, microphone preamplifier, input attenuator, fader, EQ, echo send, channel assign, and other functions.

The OUTPUT section controls the output of the console. It includes a series of *bus master modules* (or *submaster modules*), each with a *submaster* fader. The *master* faders control the overall level of all the buses. Also in the output section are the echo-receive circuits. Output-level indicators such as VU meters are used to set recording levels.

The MONITOR section controls what you're monitoring or listening to. It typically includes a monitor mixer, a cue mixer (for headphones in the studio), a monitor-level control, a cue-level control, switches to choose control-room or studio speakers for playback, mono/stereo select, solo buttons, solo level, etc.

Figure 7 shows a hypothetical mixing console arranged in clearly defined sections.

CONFUSING MIXER LAYOUTS

Now it starts to get confusing. Although there are three main sections each with its own set of controls, some of these controls are scattered around the mixing console, rather than being grouped together.

For example, if you look for a group of pots labeled "cue mix," you might not find it. Often the cue-mixer pots are distributed among the input modules. That is, each input module might include a cue-mixer pot that adjusts the level of that input in the cue mix. The same goes for the monitor mixer, whose pots might be located in the bus master modules.

Actually, this arrangement is convenient. Suppose you need to turn up the cue level of the bass guitar. You just look for the bass-guitar input module, then find its cue pot within that module.

Not all consoles are laid out as just decribed; some do have separate monitor and cue mixers. Others have multipurpose submixers to be used for whatever you need: monitor mix, cue mix, or effects mix.

Similarly, there might be no separate submaster section. Some consoles use an Input/Output (I/O) type of construction (also called "in-line"). In this type of console, all the modules (other than the monitor section) contain one input channel and one output channel.

INPUT-MODULE FUNCTIONS

Let's examine each mixer function in detail, starting with those in the input module. Each input module may include some or all of the following functions or connectors:

- Phone-jack input: This unbalanced input might be for a low-impedance microphone, a high-impedance source (such as an electric guitar), or a line-level source.
- DIRECT OUTPUTS: These are output connectors following each microphone preamp to feed individual tracks of a tape machine. These connectors bypass the summing amps in the program-bus (output) section, thus providing a cleaner signal for recording one mic per track.

Another use of direct-out jacks: Suppose you have an 8-track recorder which you want to use with a 4-channel mixer. If the mixer has 8 inputs with a DIRECT OUT on each input, you can record eight tracks simultaneously from the direct-out jacks.

- CUE jack: Some inexpensive mixers have a cue jack on the back of each input module. These jacks can be used to feed an external cue mixer.
- AUX IN jack: On some consoles, this jack accepts the output of a 2-track tape machine.
- ACCESS jacks: These jacks allow access to various points in the signal path, like a "patch point." Usually each input module has a pair of access jacks—send and receive—which you

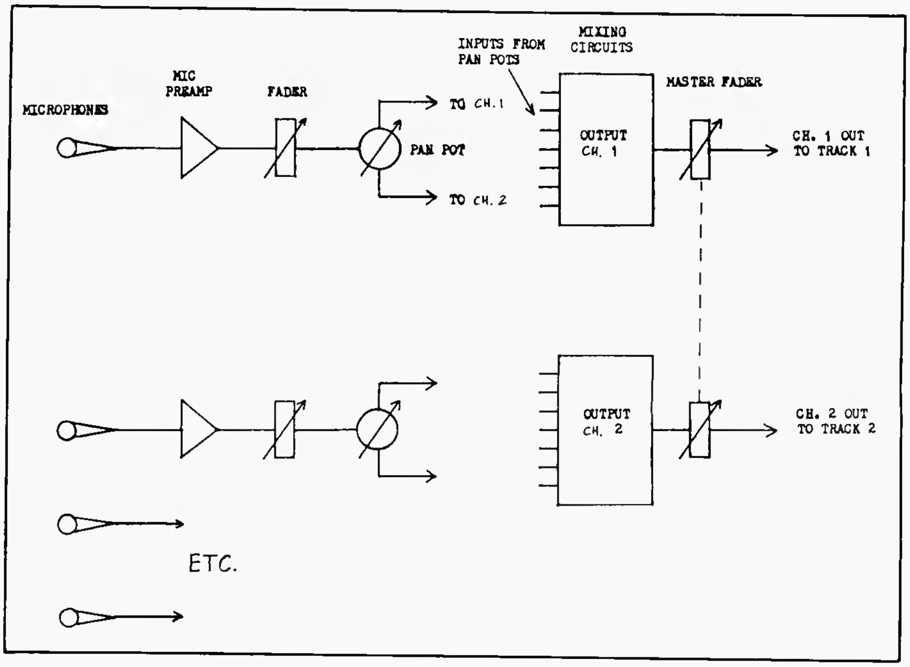


Figure 2. Signal paths in a typical 2-channel mixers.

can connect to the input and output of a compressor or limiter. Many mixers also include access jacks in each program-bus module.

Here's a major use for the access jacks: Suppose you want to apply compression to several vocals at once. Assign all the vocals to one or two program buses, then patch the compressor (mono or stereo) between the ACCESS jacks for those buses.

• TRIM (GAIN): On an input module, this pot provides continuously variable gain adjustment of the input preamplifier. High gain is needed for low-level signals; low gain is needed for high-level signals to prevent input-overload distortion (also see INPUT ATTENUATOR).

In many consoles, TRIM affects the input level of mic, line, and tape signals. During mixdown, TRIM is sometimes used to fine tune the levels coming from the multitrack tape for optimum console gain staging.

• INPUT ATTENUATOR (PAD): This is a resistive network ahead of the mic input transformer (if any). The input attenuator prevents overload of the transformer, as well as the mic preamp, by inserting fixed amounts of loss. The TRIM control (just described) prevents input overload by adjusting the gain of the mic preamp.

Since the pad decreases signal-to-noise ratio, use it only when the trim pot can't stop distortion. For example: Suppose a mic is in Input 1, and the fader is set in its optimum position for noise and headroom (about 3/4 up). If the overload LED is flashing, first turn down the TRIM until the flashing stops. If the LED still flashes, switch in the PAD. Now, if the VU meter reads too low, turn up the TRIM until you read 0 VU average.

• EQUALIZATION (EQ): The simplest equalizer is a bass and treble control. *Multiple-frequency equalizers* allow more control because you can select certain frequency ranges to boost or cut. *Sweepable equalizers* let you "tune in" the exact frequency range to work on. Most exotic is the *parametric equalizer*, which permits complete control of frequency, amount of boost or cut, and resonance or "Q."

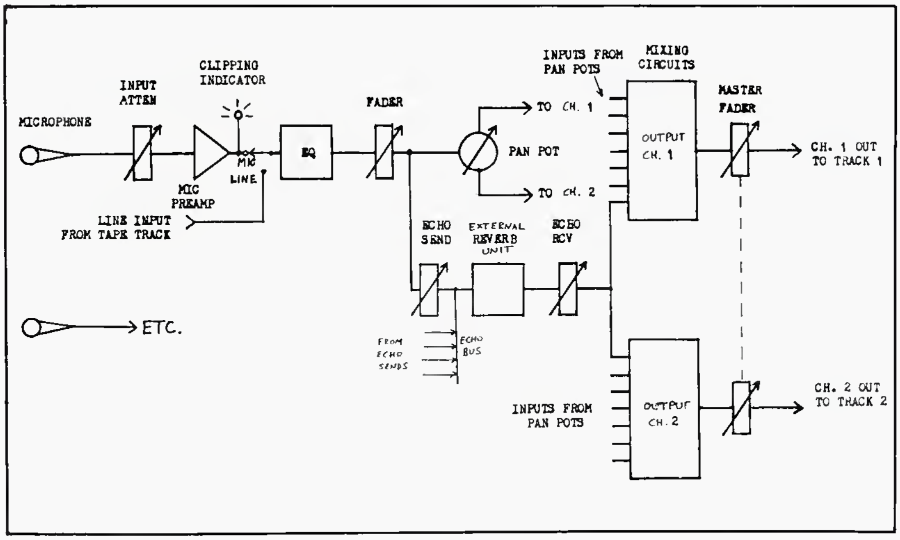


Figure 3. Signal paths in a typical 2-channel console (or in a typical multichannel console during mixdown).

• ECHO SEND (AUX SEND, EFFECTS SEND): The term "echo send" is a misnomer that stuck. The ECHO SEND control usually is used to adjust the amount of reverberation on an instrument, rather than the amount of echo. *Reverberation* is a continuous decay of sound, such as heard in an empty gymnasium or large cathedral. *Echo* is a discrete repetition of a sound, such as produced by a digital delay.

The ECHO SEND knob might be called AUX SEND or EFFECTS SEND. There may be several such channels in your console, called AUX 1, AUX 2, etc.

Whatever this function is called, it can be used to create any submix you wish. Suppose you need to run four vocal mics through a compressor, and you don't have a submaster section. You can turn up the echo sends for the four vocal-mic channels, then feed the ECHO SEND output to the compressor input. Connect the compressor output to a line input on the console. That input module can be used as a "submaster" module that controls all the compressed vocals.

In the arrangement above, the vocal-mic input channels are not assigned to program buses. Only the compressed vocals are fed to tape.

So, even though a knob is labeled "ECHO SEND," you can use it for whatever submix you need to set up. Similarly, the cue-mix bus and monitor-mix bus can be used for submixes other than what their name describes.

One popular console uses the following designations: AUX 1, AUX 2, AUX 3, AUX 4, AUX A, AUX B, and even AUX OUT. When you use this console, you have to memorize what each "AUX" means, or stick labels all over the control panel. No wonder some consoles are confusing!

You can use those aux buses for about anything, but the manufacturer suggests the following:

AUX 1: Pre-fader cue on each input module.

AUX 2: Post-fader echo send on each input module.

AUX 3: Tape-machine-A tape cue on each output bus module.

AUX 4: Tape-machine-B tape cue on each output bus module.

AUX A: Cue master.

AUX B: Echo-send master.

AUX OUT: A set of jacks in parallel with the LINE OUT jacks.

• PRE-FADER/POST-FADER switch: When used with an echo-send pot, the PRE/POST switch selects

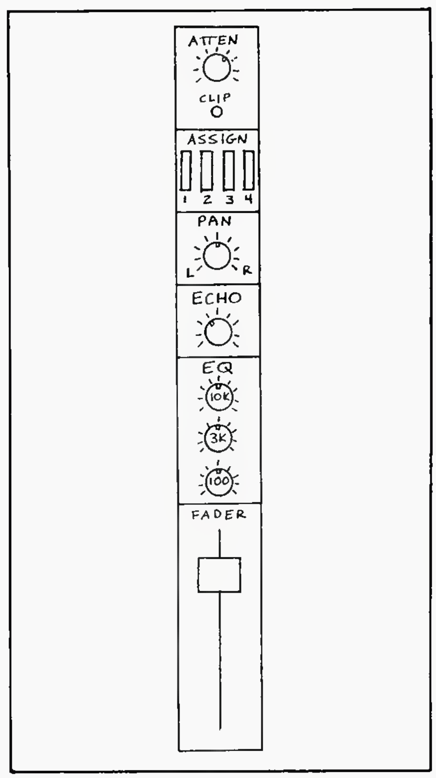


Figure 4. An input module.

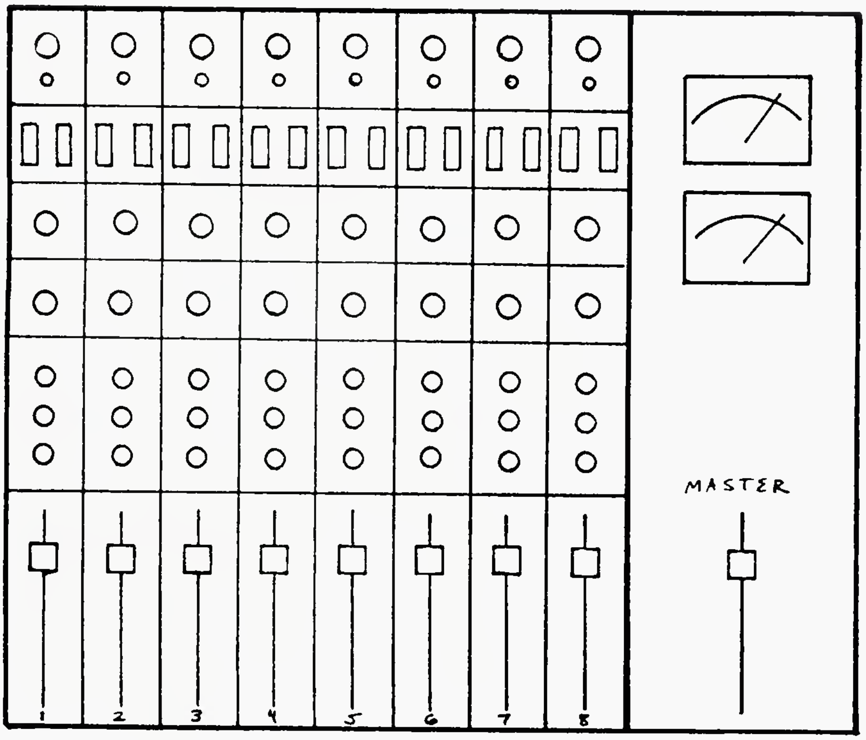


Figure 5. A mixer comprised of several input modules and one two channel output module.

whether the echo send is derived pre-fader (before the fader) or post-fader (following the fader). A *pre-fader* echo send is not affected by the input's fader: If you turn down the input fader, the echo remains. A *post-fader* echo send follows the fader action: If you turn down the fader, the echo send level goes down too.

Use POST when you want the recorded instrument to sound close (high direct-to-reverb ratio). Use PRE when you want the instrument to sound far away (low direct-to-reverb ratio). That is, adjust the relative levels of the pre-fader echo send and the input-module fader for the desired sense of distance. Remember, POST equals close, PRE equalsawee (away, or distant).

Other functions such as AUX send may be pre- or post-fader.

- CUE (also FOLDBACK or FB): In each input module, the CUE pot controls the level of that input in the cue mix. The cue mix is the mix heard over headphones in the studio.

When overdubbing, musicians need to hear their own instruments over headphones, as well as previously recorded tracks to play along with. The headphone levels of the live instruments being recorded are controlled by MIC CUE or CUE pots; the levels of recorded tracks are controlled by TAPE CUE pots.

Some consoles have two CUE buses. They can provide two independent cue mixes, or a single stereo cue mix.

- CHANNEL ASSIGN (BUS ASSIGN): A set of switches to assign input signals to various output channels. For example, suppose inputs 1-6 are drum microphones, and you want to feed them all to Track 2 of a four-track recorder. On input modules 1-6, you'd punch in the Channel-2 assignment button. Then all those drum mics would be routed to Track 2 of your recorder (assuming you connected Console Channel 2 to Tape Track 2).

If you assign a single input to two channels, the input feeds both channels. The PAN control then varies the relative level being fed to each of those two channels. Two-channel assigns typically are used for stereo piano, stereo background harmonies, and stereo drum mixes. In a stereo drum mix, you pan each drum to its desired location between the monitor speakers.

- SOLO (PFL): The SOLO button in an input module lets you monitor only that input. More than one input can be soloed at one time. On British consoles, the SOLO function is called "PFL," which stands for PRE FADER LISTEN: You listen to or monitor the signal before the fader.

In consoles that have both PFL and SOLO, PFL is pre-fader and is used mainly to listen for distortion during tracking; SOLO is post-fader and is used for soloing during mixdown.

- MUTE turns off the input by disconnecting the input-module output from CHANNEL ASSIGN and DIRECT OUT. During mixdown, you can reduce noise by muting inputs that are temporarily unused. In some mixers MUTE is called "CHANNEL ON/OFF."
- PHASE (POLARITY INVERT): This switch inverts the polarity of the input signal. That is, it switches pins 2 and 3 to flip the phase 180 degrees. You might use it to correct for a mis-wired microphone cable that is out-of-phase with other such cables.
- +48: This button turns on 48-volts phantom power at the mic input connector for powering certain condenser microphones.

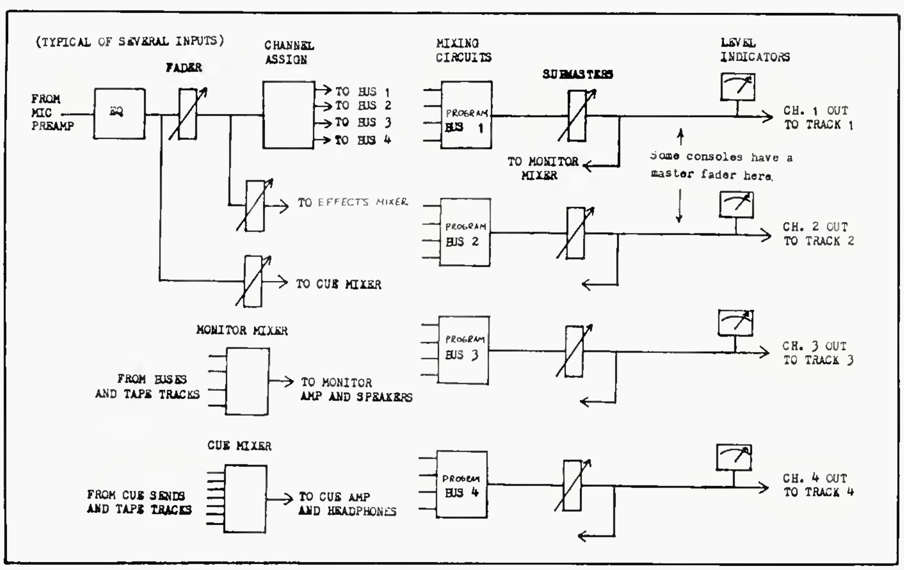


Figure 6. Signal paths in a typical multichannel console (during recording).

• Automated mixing controls: These controls (Read, Write, Update, etc.) are beyond the scope of this article. Basically they control the functions of a computer-assisted mixing system. The computer memory remembers and updates console settings so that a mix can be performed and refined in several stages.

mix, keyboard mix, vocal mix, etc. The level of each mix or group is controlled by a SUBMASTER fader.

A 16 x 4 x 2 console has five output modules: Four submaster modules and a stereo master module.

• SUBMIXER: Some small mixing boards include a multi-purpose submixer to be used for whatever you need. Instead of providing a set of pots labeled "CUE," the console may provide a submixer that can be used as a cue mixer, a monitor mixer, or whatever.

By using a single group of pots for many functions, mixers can be made less costly. Often you need only one function at a time. During tracking and overdubbing you need only a cue mix; during mixdown you need only an echo mix. You also need a monitor mix, which can be handled with an outboard monitor mixer. More-expensive consoles allow simultaneous cue, echo, and monitor mixes for maximum flexibility.

• BUS TRIM: This rotary pot provides variable gain reduction on the bus, used in addition with the bus master (or submaster) fader.

• BUS/MONITOR/CUE switch for ECHO RETURN: A switch that feeds the echo-return signal to your choice of three destinations: program bus (for mixdown), monitor mix, or cue mix.

• METER SWITCHES: In many consoles, the VU meters can measure signal levels other than console output levels. Switches near the meters can be set so that the meters indicate bus level, effects-send level, effects-return level, monitor-mix level, etc.

Those readings help you set optimum levels for the outboard devices receiving those signals. Too-low a level results in noise; too-high a level causes distortion in the outboard unit. For example, if the echo-return signal sounds garbled or distorted, the cause may be an excessive echo-send level. That condition can be verified by checking the VU meters switched to read the ECHO or EFFECTS bus.

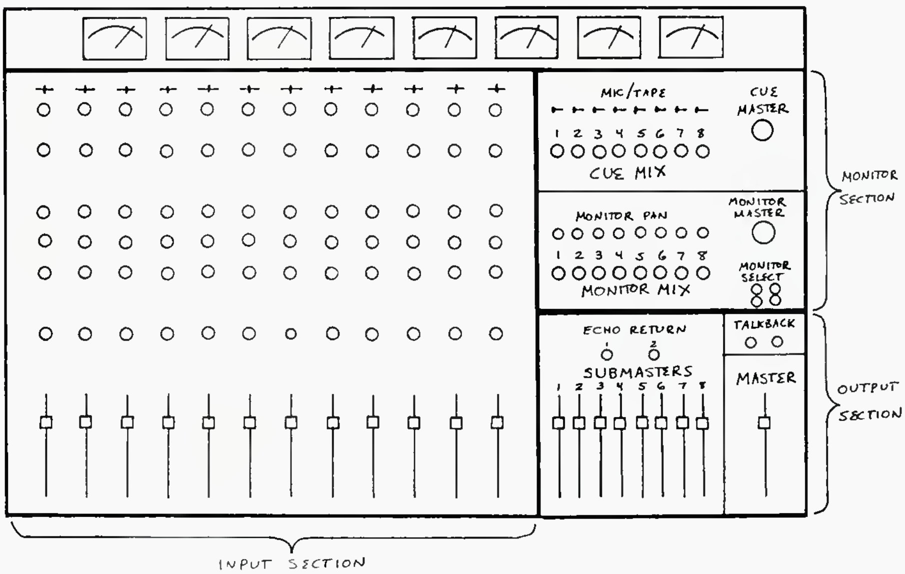


Figure 7. A hypothetical mixing console arranged in clearly defined sections.

OUTPUT-MODULE FUNCTIONS

• BUS IN jack: A line-level input to a program bus, used as an effects return.

• FOLDBACK connectors: These connectors are in parallel with the TAPE IN jacks. The output of each tape track is connected to a TAPE IN jack, so the FOLDBACK connector mults or parallels the tape-machine output. The FOLDBACK jacks can be used to send the tape outputs to an outboard device in addition to the console, such as an external cue mixer.

• OUTPUT MODULES (also GROUP, SUBMIX, SUBMASTER, or BUS MASTER modules): Each output module includes a combining network (fed by the input modules) and a SUBMASTER fader that controls the overall level of the submix feeding each program bus. The SUBMASTER fader might be called a *GROUP MASTER* or *BUS MASTER* fader.

As decribed earlier, small submixes can be set up on the console: a drum

MONITOR-SECTION FUNCTIONS

• MONITOR MIXER: A submixer that controls the balance of the instruments heard over the monitor speakers. The monitor mixer lets you hear an approximation of the final product without affecting the recording levels. Monitor-mixer pots are typically distributed among the submaster (bus master) modules. The monitor mixer might include pan and echo.

• BUS/TAPE SWITCH: This lets you monitor either the program buses or the tape machine. When the source/tape switch on a tape machine is set to "tape," then the BUS/TAPE switch lets you monitor off-tape to check for tape distortion.

• MONITOR SELECT SWITCHES: Similar to the bus/tape switch, this group of switches allows monitoring of various signals such as LINE (for 2-track recorders), AUX (such as cue or echo), or MON (the monitor mix of the buses).

• DIM: A switch that reduces the monitor level by a preset amount (as in "Dim the lights").

ECHO RETURN TO CUE is an echo-return level control that affects the amount of echo or reverb heard in the studio headphone mix. This echo is independent of any echo being recorded on tape.

• ECHO RETURN TO MONITOR: This echo-return control affects the amount of echo or reverb heard in the monitor mix. This echo is independent of any echo being recorded on tape.

CONCLUSION

We've covered some of the more confusing mixer functions. The next time you encounter an unfamiliar mixing console, you may have a better idea of what those strange labels mean. And if all else fails, read the instruction manual! ■

Overview Of Synthesizer Technology

Jesse Klapholz

Whether we like it or not, synthesizers are here to stay!

SYN-THE-SIS, n. [L (Gr. (syn-, together + tithenai, to place], the putting together of parts or elements so as to form a whole. Synthesizers are gadgets, of a strange brew, that many of us audio engineers frown upon—members of the musical community also cringe at the mention of this blasphemous word. Synthesizers, however, are here to stay—they will not fade away as some fad associated with the Carter/Reagan era. These contraptions are deeply rooted in applications of speech and musical instrument analysis, voice and communication research, speech recognition and synthesis, and music-composition/production/presentation just to name a few broad and diverse areas. Because of the key role the audio engineer plays in all sectors of communications, along with the understanding of digital and computer technology, a good working knowledge of synthesis is becoming more important.

BACKGROUND

The history of musical instruments spans over many centuries. Over the years they have evolved around several families including string, woodwind, brass, and percussion. The way these instruments produce sound, and the way in which they are played, vary greatly. One thing that they do have in common is that energy is transformed from the player (either by the movement of air or a mechanical disturbance) through the instrument and the air to the listener who perceives the end product as music. In short, the player provides the energy-input for *acoustical* musical instruments.

In contrast to the acoustical instruments, electronic instruments' energy source is electricity. The earliest musical instrument using this new energy source was the Telharmonium. In 1895, when Thaddeus Cahill invented the Telharmonium it marked the epoch of electronic instruments. His invention clearly had short term, as well as long term, effects on the music and entertainment business. People talk about *phoning in* their synth-parts, but as early as 1905 Cahill was generating sound with over 100 alternators, controlling them from a keyboard, and transmitting musical concerts to subscribers over leased telephone lines.

About the same time, the vacuum tube was invented, setting the stage for electronic modification of signals and electronic tone production. While most of the work in early electronics focused its efforts on broadcasting and cinema related applications, over 100 patents for electronic instruments were issued by 1930.

In 1920, Leon Theremin appeared on the scene with a radical musical instrument, not simply because it was electronic, but more importantly because it employed new playing techniques. The instrument, called the Theremin, consists of two radio-frequency oscillators which beat against each other to produce an audio-frequency tone, in a manner similar to a beat-frequency oscillator. The capacitance of the player's hand-proximity to the instrument changes the pitch and volume of the Theremin. Therefore, the Theremin was the first *modulation* type synthesizer using analog/real-time control, in contrast to the recent embodiment using digital circuitry to control the frequency output and level over time.

By the 1930s, the Hammond organ had been developed along with a host of other electronic instruments including the electric violin, piano, guitar, music box, and tympani. Efforts to use electricity in the production of music actually pre-dates the 1900s starting with Diekmann's 1887 patent on electromagnetically exciting tuned strings. At least seven patents were issued on electrical instruments prior to 1900. As is evidenced by the many patents issued through the 1930's, much work was advancing tone sources for musical instruments and tone modification.

ANALYSIS

Of the many researchers to contribute to the science of sound during the 1800s, among the most notable were Fourier, Helmholtz, and Rayleigh. Helholtz used resonators to analyze musical instruments in terms of fundamental frequencies and their harmonics. Fourier's contribution was his mathematical description of a complex waveform—where any arbitrary waveform can be described by a trigonometric series of sine and cosine multiples of the fundamental and this (or the whole equation itself) is often referred to as Fourier's Series. Rayleigh analytically described virtually any vibration found in nature. It is often amusing that many so called modern inventions find their roots, or even descriptions, in his two-volume book, *The Theory of Sound.*

The Scott-Koenig phonautograph in 1859, the Koenig Manometric Flame in 1862, the Blondel-Dudell oscillograph in 1893, the photographic reproduction of waveforms from talking-machine records by Hermann, Bevier, and Scripture around 1900, and Miller's Phondeik in 1909, were the early means by which sound waves could be recorded for the purpose of analysis. Resonators and tuning forks were the predominant analysis instruments available in the 1800s, clearly, more advanced apparatuses were needed.

Having developed several different methods of taking pictures of sound, the process of analyzing the picture or *curve* was the next step. Fourier showed how any waveform could be analyzed by means of mathamatical computation; but, as this tedious process required several days' work for a single curve, various mechanical wave analyzers were constructed to simplify the task. Henrici's Harmonic Analyzer in 1894; Miller's Amplitude and Phase Calculator in

Jesse Klapholz is a contributing editor.

1916; Michelson's Harmonic Analyzer and Synthesizer in 1898; and planimeter type of harmonic analyzers by Rowe in 1905, Mader in 1905, and Chubb in 1914, all reduced the calculations of the Fourier Series to a matter of minutes.

The inverse of harmonically analyzing a waveform is to combine several simple curves to find their composite curve; this is *harmonic synthesis.* This can be performed by manual computation and graphic techniques; however, because of the impracticality of these methods, *harmonic synthesizers* were developed. One of the first synthesizers was developed by Rayleigh in 1876, to be used as a tide-predicting machine. Later, in 1914, Miller built a 32-element harmonic synthesizer. It took twelve minutes to synthesize a 32 component waveform!

Subsequently, analyzers and synthesizers became an integral part of the research work in speech and musical acoustics. In 1906, Cahill, built an eight-harmonic synthesizer using harmonically-geared shafts driving AC dynamos. Organs that were then introduced used superposition of partials and formant filtering techniques. In the 1930s, work at Bell Labs and UCLA led to two independent reports by Fletcher and Knudsen which both stated the necessity of an electronic synthesizer capable of combining a large number of harmonic partials. In 1940, Bell Labs built a 100-element harmonic tone synthesizer. Its sound source was 100 separate harmonically-tuned sine waves recorded on a rotating magnetic drum.

All the synthesizers developed until this point were hybrids of mechanical and electrical designs. In 1942, an all-electronic musical instrument was patented and built by E.L. Kent capable of additive harmonic tone synthesis. Its principle of operation is based upon a fixed ultrasonic frequency complex waveform that is sampled at a variable ultrasonic rate.

American composers began to experiment in earnest with *electronic music* in the 1950s. At first, the equipment was simple, often just tape recorders to record and transform the improvised sound of flute, piano, or voice. Leopold Stokowski's interest (conductor of the Philadelphia Orchestra) in electronic music, starting with the Telharmonium in 1906, led him to schedule the first American public performance of music for tape recorder by composers Vladimir Ussachevsky and Otto Luening in 1952. Stokowski had a life-long dream of a "temple of music," and constantly featured the latest developments in sound production techniques in his concerts.

In the early 1950s, a synthesizer program was under development at RCA under the direction of Olson and Belar which resulted in the RCA Electronic Music Synthesizer. The RCA synthesizer was programmable in its control parameters over a tone's frequency, intensity, growth, duration, decay, portamento, timbre, and vibrato. A punched paper roll contained all the information of a performance and could change any parameter. The RCA synthesizer was initially used as a musical instrument and musical composition analysis tool. Later, it was applied to speech analysis and voice synthesis.

While work was going on at RCA using analog techniques, Bell Labs were getting into digital synthesis. In 1957, Max Mathews first demonstrated his work on digital speech encoding. The IBM 704 computer, used in his work enabled him to study encodings in several months, a process that would have otherwise taken years to develop the appropriate electronics.

Computer generated music at Bell Labs was a spinoff of their digital speech communications technology. When they realized that music could be readily represented in the digital domain, they applied the technology to the synthesis of musical instruments. A program was then written called Music I which used a triangle waveform and three variable parameters: pitch, duration, and amplitude. Even though the Bell Labs synthesizer was digital and the RCA was analog, the digital-analog issue is actually not germane to the way in which synthesizers produce sound. In fact, they were both *subtractive* synthesizers. There are four well-known methods for synthesizing musical tones. They are: 1) *additive synthesis,* 2) *subtractive synthesis,* 3) *waveform synthesis,* and 4) *modulation synthesis.* Below is a brief discussion describing these four basic *families* of methods used in all synthesizers as we know them today.

ADDITIVE SYNTHESIS

Additive synthesis was used in the first mechanical synthesizers and some early organs where a number of harmonics were added to a fundamental frequency. The technique of additive synthesis is rather straightforward. It is the simple combination (addition) of two or more sound waves. Using additive synthesis any arbitrary finite harmonic spectrum can be produced by controlling the amplitudes and phases of the harmonics over time in relation to a fundamental frequency. This technique is a direct application of the Fourier Series whereby, the addition of two or more waves point by point obtains a complex waveform. The waves may be any combination of pure and/or non-pure waves. If two waves are combined that are not harmonically related, their sum will be continuously changing in shape because of the relative phase between the component waves.

Additive synthesis is extremely attractive to those involved in the exploration and analysis of the characteristics of musical iinstruments. It is a direct application of the classical conception of musical instrument physics. Much of what we know today of musical instruments' timbres was learned through additive synthesis/analysis techniques. While we have talked about harmonically related components in additive synthesis techniques, inharmonically related components may be created as well. Controlling the paramaters over time and introducing other sources, e.g., random noise, many realistic percussive sounds including piano instruments have been implemented within the realm of additive synthesis.

Additive synthesis, as we have shown, can be implemented mechanically in its most rudimentary form. Today these mechanical embodiments are only of historic or educational/demonstrative utility. Electrical implementations may either use analog, digital, or a hybrid of both technologies. With the use of a computer the parameters may be set using complex math, analytically with no prior knowledge of the instruments characteristics, or graphically *ad hoc.*

SUBTRACTIVE SYNTHESIS

It does not take a genius to state that subtractive synthesis is the reciprocal of additive synthesis. In the subtractive synthesis technique one starts with a waveform such as sine, sawtooth, triangle, square-waves, or any combination of these, which is very rich in harmonic content. This harmonically-rich waveform is then filterd to attain the desired spectral and timbral characteristics. The filters used vary greatly in type from fixed to ones that can be changed as a function of performance input.

Subtractive synthesis works in much the same way as the human voice does; i.e., in the human voice there is a noise source provided by air flow from the lungs, which is then modulated by the vocal chords, and then *filtered* by the vocal tract, mouth, and nasal cavities. These *filters* are called *formants* and there are usually about a half-dozen or so which can be moved up and down slightly to reinforce various pitches. Musical instruments work in much the same way; i.e., there is a waveform input from a guitar string, for example, that is coupled to the guitar body, which has its own *formants* that alter the spectral and timbral characteristics of the original waveform. Therefore, we can say that subtractive synthesis is an emulation of the manner in which acoustic musical instruments operate.

The main limitation to the subtractive synthesis process is the complexity of the *formant* modelling and the complexity of the original waveform. At Bell Labs Mathews and Kohut simulated the body sound of an acoustic violin using between seventeen and thrity-seven resonant filters. They used a magnetic pickup near the bridge of a violin to isolate the *input* waveform of the strings, which closely approximates a triangle wave, and input this signal to a series of filters whose resonant damping and frequency centers were adjustable. They were able to emulate the sound of the violin body with twenty or thirty resonances in the 200-5000 Hz range. The main attraction to subtractive synthesis is the ease in which a harmonically rich waveform can be produced and easily filtered, all in simple analog circuitry.

WAVEFORM SYNTHESIS

Waveform synthesis is a process whereby a periodic signal is generated by specifying the instantaneous values of the waveform. As in all synthesis techniques the envelope of the waveform over time may be controlled with either analog or digital circuitry. The wave form may also be produced with analog or digital circuitry. However, waveform synthesis has one disadvantage: for accurate control over the spectrum and timbral characteristics complex computations are necessary.

Some sampling synthesizers are part of this family. Sampling synthesizers are, in principle, digital recorders, with variable speed playback—the key word here is *variable speed.* Once we have changed the playback speed so has the timbral characteristics. Also, due to memory (record length time) the so-called steady-state portion of sounds is usually *looped* and its envelope modified and filtered. Therefore, while a sampling synthesizer starts with a *true* recording of a musical instrument, in reality, we are capturing a true waveform, eliminating the complexity of analysis and computation that would otherwise be needed in arbitrarily specified waveforms. Some sampling synthesizers also have limitations imposed upon them due to recording techniques, the amount of recording *channels*, and timbral control. However, because a sound is recorded rather than analyzed or calculated *ad hoc*, strikingly realistic sounds can be duplicated easily with sampling synthesizers.

MODULATION SYNTHESIS

Modulation synthesis may be implemented in many ways. The main sub-categories under this heading include *amplitude modulation* (AM), *frequency modulation* (FM), and *pulse width modulation* (PWM). The output signal in modulation synthesis is produced by combining a *carrier* frequency with a *modulator* frequency. In AM, the modulator signal changes the amplitude of the carrier. In FM, the modulator causes the carrier frequency to vary however, it does not change the level of the carrier—the amplitude of the modulator controls the frequency deviation (modulation) of the carrier.

The FM process yields a spectrum with peaks at the carrier and in sidebands at integral multiples of the modulation frequency with respect to the carrier. In FM, both harmonic and inharmonic spectra may be produced by control of the relationship between the carrier and modulator. Furthermore, dynamic spectra may be produced. This may be implemented by the use of *algorithms*, i.e., a step-by-step set of mathematical procedures. While this type of synthesis does not lend itself to be compared easily with the classic understanding of musical acoustics, its virtue is the complex time-variant sounds that may be produced with a minimal amount of computation. Many sub-sets and similar math/modulation techniques have been developed since John Chowning first introduced FM synthesis techniques in 1973.

APPLICATIONS

At first electronic synthesizers were controlled by paper-punched rolls; later, computer programs and languages were written to perform the task with greater performance. The turning point was when Bob Moog used VCOs and VCAs to connect a piano keyboard to subtractive synthesis modules in 1964. W. Carlos' "Switched on Bach" was the record that really popularized the new synthesizer sound and got things going.

The synthesizers first available to musicians were monophonic, hard wire patch, analog controlled, and not touch-sensitive. One by one these features were incorporated into commercial designs by the early 1980s. With the advent of practical and affordable digital control and serial communications of synthesizer control, via the MIDI standard, synthesists are able to produce and control a myriad of sounds. Real-time synthesizer control is now being extended to guitar players, and most recently to any mono-voice instrument, or even the human voice. Sequencers can drive synths from either real-time input or *step-programming* vis-a-vis word processing type techniques.

Synthesis has its roots in the analysis of musical and speech acoustics. The devices used have been in evolution since the turn of the century. What technology is available to the musician has evolved around electronics becoming practical and affordable enough to implement synthesis techniques. Computer, digital, LSI, and VLSI electronics, are among some of the developments over the past decade that have enabled manufacturers like Casio to sell synths for under $500 that do more than any of the first generation.

FUTURE DIRECTIONS

Predicting the future in the short term is easy since synthesizer technology draws upon the technology from the computer and communications sectors. Array processing, systolic processing, artificial intelligence, partitioned-micros (equivalent to mainframes), and transversal filters are just a few examples of current technology that will be incorporated into the next synthesizer generation.

In 1962, in "Music, Acoustics & Architecture," Leo Beranek wrote, "If purely electronic music becomes the vogue, even the musician would be eliminated; the concert hall would pass from the scene, along with the symphony orchestra, the concert-goer, the usher, and the ticket-taker. The musical innovator and the electronic technician would then rule supreme—unless the electronic computer manages to displace them." ■

db Buyer's Guide Synthesizers (drum and keyboard)

AKAI

The AX80 is an 8-voice polyphonic velocity sensitive synthesizer with 16 DCOs and 8 sub oscillators, 3 LFOs, 2 envelope generators, fluorescent graphic display, cassette interface, pitch and modulation wheels, and 61-note keyboard. It has MIDI IN/OUT/THRU.
Price: $1,195.00.

The AX60 is a 6-voice MIDI polyphonic synthesizer with 6 VCOs, stereo chorus, arpeggiator, 2 envelope generators, sampler editing system, 64 user programmable memories, 4 split modes, 8 preset splits, pitch bend and modulation wheels, 61-note keyboard and edit compare switch.
Price: $999.95.

The S612 is a 6-voice MIDI polyphonic, digital sampler with 8-second sample time, 16 kHz bandwidth, 32 kHz sampling frequency, overdub, MIDI channel assign, automatic looping, manual splice, alternating mode, low pass filter, decay, LFO, tuning control, external trigger switch, mic and line input and input monitoring, rear interface for disc drive, and for multipin sampler voice out jack.
Price: $799.50.

The ME 20A is a MIDI digital arpeggiator/sequencer with 1096 notes of arpeggiation, sequencer, step write, chord or monophonic operation, gate control, and up, down, or sequence patterns.
Price: $149.95.

CASIO

The CZ-3000 is a 61-key synthesizer featuring 16/8 note polyphonic keyboard with 32 presets and 32 internal memory locations, key split, tone mix, polyphonic portamanto, and ring modulation. It has MIDI IN/OUT/THRU with 8-channel multi-timbral playback.
Price: $999.00.

The CZ-5000 has all the same features as the CZ-3000 above, and has an 8-track sequencer with both real and step time entry. Also includes the ability to dump sounds and sequence data to cassette tape.
Price: $1,199.00.

The CZ-101 is a 49-key digital programmable synthesizer that is 8-note polyphonic when using a single line of sound controllers and 4-note polyphonic when using both lines for a richer, fuller sound. The MIDI interface will read four MIDI channels at once, allowing multi-timbral performance.
Price: $499.00.

The CZ-230S is a 49-key digital preset version of the CZ-101, above, with 100 on-board sounds and a built-in speaker. It has an on-board drum machine with 20 preset patterns and 30 bars of user-programmable patterns. It has MIDI IN/OUT/THRU with 4-channel multi-timbral playback and level control over each channel.
Price: $449.00.

The CZ-1 is a 61-key digital synthesizer with both velocity and after-touch sensitive keyboard. The velocity control can control volume, timbre, and pitch envelopes independently for each line. It is 16/8 note polyphonic with 64 user programmable memories, key split, tone mix, polyphonic portamento, chorusing, pitch bend, and modulation wheels. It also has MIDI IN/OUT/THRU with 8 channel multi-timbral playback with polyphonic capabilities.
Price: $1,199.00.

The RZ-1 is a PCM digital drum machine with user sampling. It includes 12 sound sources, 100 patterns, 20 songs and real and step entry recording. It has individual line outputs and volume controls, user sampling at 20 kHz with 4 different 0.2 second pads which can be grouped together for longer sounds. It has MIDI IN/OUT and cassette dump for patterns and sample data.
Price: $599.00.

DURALINE INDUSTRIES INC. (SYNDRUM)

The Model CM is a self-contained drum machine package with basic tom, clave, and kick drum sound. On-board controls include sustain, sweep, tune, and volume.
Price: $79.00.

The Model 178 is a percussion synthesizer capable of producing a variety of sounds and effects from which to choose. The single Syndrum and control console provide complete control over the sonic characteristics including pitch control, modulation, sweep, wave form, and noise. Velocity sensitivity provides full expression while playing.
Price: $179.00.

The Model 278 is the same as the 178 above, except the features include dual sound source, mic level out, headphone out, and individual and mixed outputs with master volume.
Price: $349.00.

The Model 478 is the same as the 178 above, except additional features which include quad sound source.
Price: $699.50.

ENSONIQ

The Mirage is a 61-key digital sampling keyboard with polyphonic velocity sensitvity. It features 8-voice polyphonic and poly timbral operation, availability of up to 16 different sampled sounds at one time, four instantly accessible programs per sound bank, and variable sampling rate (from 2 to 8 seconds). Sound storage is by means of a 3 1/2-in. micro floppy diskette with each capable of storing 3 upper band and 3 lower band sounds.
Price: $1,695.00.

The Mirage digital multi-sampler has all the same features as the Mirage keyboard, above, but without the keyboard.
Price: $1,395.00.

The Sampled Digital Piano is a 76-key digital piano with polyphonic velocity sensitivity. It features octave switch control, assignable split point, 10-voices, 12 different available sounds, and MIDI IN/OUT/THRU.
Price: $1,395.00.

The ESQ-1 Digital Wave Synthesizer is a 61-key complex waveform synthesizer with multitrack MIDI sequencer that is 8-voice polyphonic with 3 oscillators per voice. It features programmable split point, sound layering, programmable panning, and MIDI IN/OUT.
Price: $1,395.00.

EUROPA TECHNOLOGIES

The Lync Systems LN-1 MIDI Controller is a strap-on remote MIDI controller with four-octave keyboard, direct master program selection, 64 master programs, assignable switch for octave transposition and sustain, hold function, reversable pitch wheel, chord function and program advance.

MUSIC INDUSTRIES CORP.

The Bit 99 synthesizer is a 6-voice, programmable, touch sensitive keyboard which features program chaining, digital and analog voices, programmable keyboard splits, MIDI control, and "parking your sound" which enables the user to store a preset and edited version of a sound for comparison before it is written to memory.
Price: $1,495.00.

The Bit 01 Expander is a 6-voice programmable, touch sensitive synthesizer designed to be controlled by any MIDI keyboard or controller. It has 75 preset sounds and 24 doubling or split combination sounds for a total of 99. It also provides the same features as the Bit 99, above.Price: $1,095.00.

The Bit MMK is a 72-key MIDI master keyboard with velocity and after-touch response, programmable controllers, 64 MIDI configurations which can be memorized and recalled. It has a sequencer with 4000 note capability, 2-octave transpositional range, tape interface, and two splits.
Price: $1,095.00.

THE MUSIC PEOPLE (DRUMFIRE)

The DF-500 Electronic Drums feature five electronic sensors which are attached to the acoustic drums. Each of the five channels is independently mixed for sensitivity, oscillator decay, sweep, volume, balance, pitch, and left/right pan. Also includes footswitch and headphone jacks.
Price: $440.00.

The DF-2000 is a rack mounted dual channel adjustable preset percussion synthesizer with two sensors which are attached to the acoustic drums to produce four different sounds. Each channel has 10 control parameters for sensitivty, oscillator decay (two), noise decay, noise filter, balance, volume, sweeep, pitch, and wave shape.
Price: $195.00.

NEW ENGLAND DIGITAL

The Synclavier Digital Music System's features include 76-note velocity/pressure sensitive keyboard, 16-track memory recorder with super floppy disc drives. Options include Sample to Disc, Music Printing, Polyphonic Sampling, Multi-channel Outputs, and SMPTE.
Price: $35,000.00-200,000.00.

OBERHEIM/ECC

The Matrix 12 is a MIDI programmable synthesizer based on the Oberheim Xpander. It features a 5-octave velocity, after-touch, and release velocity sensitive keyboard, controlling 12 independently programmable voices.
Price: $4,995.00.

The Matrix 6 features a 5-octave, velocity, pressure, and release velocity responsive keyboard. The six voices can be split or doubled by means of two independent keyboard zones. Memory capacity is 100 single patches and 50 multi-patches.
Price: $1,595.00.

The Matrix 6R is the same as the Matrix 6, above, but is a rack mounted unit without the keyboard.
Price: $995.00.

The Xpander is a 6-voice, individually programmable synthesizer. Each voice contains two oscillators, a filter, two output amplifiers, five envelope generators, five LFOs, four ramp generators, three tracking generators, and a lag processor.
Price : $2,995.00.

The XK is a 5-octave, velocity, release velocity and pressure sensitive keyboard that can be split into three zones, overlapping if desired, each with its own MIDI channel.
Price: $995.00.

The DMX is a programmable digital drum machine capable of storing over 5000 events of 8 notes each in memory, 200 sequences, and 100 songs. It features real and step entry and comes standard with seven voices.
Price: $2,195.00.

The DX is a digital programmable drum machine containing a crash/ride cymbal voice, and a full compliment of MIDI functions. Other features include punch in and out, record countdown, cue tempo, and selective loading of sequences and songs.
Price: $1,395.00.

The DX Stretch is an addition to the DX drum machines. It provides up to four rows of three voice buttons each for user-changable voice chips. The features are the same as the DX above, and storage is achieved via improved cassette interface.
Price: $495.00.

OCTAVE-PLATEAU

The Voyetra Eight is a fully programmable 8-voice synthesizer containing a powerful microprocessor controlling eight versatile synthesizer circuits. It responds to all MIDI controllers with adjustable sensitivity. Optional velocity and pressure sensing keyboard with programmable joystick (VPK-5) available for $995.00.
Price: $4,595.00.

ROLAND

The MKS-80 Super Jupiter is an 8-voice MIDI sound module that responds to single or dual MIDI channel assignments, velocity, pressure sensitivity, hold pedal, system exclusive, pitch bend, volume, and patch change information.
Price: $2,495.00.

The MKS-30 Planet S is a 6-voice MIDI sound module featuring 64 patches plus RAM cartridge storage, and can be programmed with the PG-200 Programmer. It responds to MIDI velocity, pitch bend, patch change, and pedal hold.
Price: $995.00.

The MKS-10 Planet P is a 16-voice, velocity sensitive MIDI sound module containing 8 preset keyboard instrument sounds (pianos, clarinet, harpsichord, and electronic piano) and features MIDI IN/THRU and built-in stereo chorus and flanger.
Price: $995.00.

The JX-8P is a fully MIDI compatible synthesizer/controller featuring 32 user programs, 64 presets, and a RAM cartridge storage. It sends and receives velocity and after-touch. Programming is possible by parameter/step editing.
Price: $1,695.00.

The Juno 106 is a 6-voice MIDI synthesizer with 128 user programs, a 61-key keyboard, and MIDI IN/OUT/THRU. It uses system exclusive for patch storage/editing to personal computers.
Price: $1,095.00.

The MKS-7 Super Quartet is a multi-timbral MIDI sound module that features four sections including melody, chord, bass, and 11 PCM drum sounds. Each section can be assigned to a separate MIDI channel for use with controllers/sequencers and MIDI software.
Price: $1,195.00.

The MKB-1000/300 are MIDI keyboard controllers designed to send specific MIDI information to MIDI compatible devices.
Price: MKB-1000-$2,195.00; MKB-300-$1,295.00.

The TR-707 Rhythm Composer uses a digital sampling process to create 13 different sounds. Flam and shuffle functions allow you to produce rich and expressive patterns. Create up to 64 individual patterns and use these patterns to write complete rhythm tracks with a total memory capacity of 999 measures.
Price: $650.00.

The DDR-30 is a digital drum system consisting of the SSR-30 Module and two kinds of drum pads. Features include realistic digital sound sources, flexible sound modificaation, and MIDI capability. The system has 6 drum voices and four different PCM digital sounds are provided for each.
Price: $1,395.00.

SEQUENTIAL

The Prophet 2000 is an 8-voice sampling instrument featuring a 61-key velocity sensitive keyboard. It includes pitch and modulation wheels, arpeggiator, 3 1/2-in. diskette storage, programmable split keyboard, full sample editing and looping capabilities, and MIDI/IN/OUT/THRU.
Price: $2,499.00.

The Prophet 2002 is a rack mounted version of the 2000, above, but it does not include keyboard.
Price: $2,299.00.

The Model TOM is a fully programmable drum machine featuring 8 digitally recorded instrument sounds. It lets you program volume, tuning, and stereo pan individually for each of its sounds. Rhythm patterns can be recorded in real time or in single step mode. Total memory capacity is over 3000 notes.
Price: $799.00.

SIMMONS

The SDS1000 is a programmable 5-piece electronic drum set capable of producing both digital and analog sounds via the striking of the Simmon's floating head pads. There are a total of 10 different drum kits (5 factory, 5 user programmable) available at the touch of a button or footswitch.
Price: $1,000.00.

The SDS9 is a programmable, 5-piece drum set with 20 factory and 20 user memory patches. It has digitally sampled snare with rimshot, microprocessor controlled dynamic expansion, auto-trigger mode for programming ease, programmable sequence of memory patch changes, and fully assignable MIDI interface.

The SDS7 is an expandable, modular electronic drum system with analog and digitally sampled sounds, plus it has user sampling, full programmability and 99 drum kit memory patches.

YAMAHA

The DX-7 is a 16-voice FM digital synthesizer with 61 keys. It has 32 presets, 6 operators, and 32 algorithms to the FM sound source. It features velocity and after-touch keyboard with full MIDI compatibility, and RAM and ROM memory cartridges.
Price: $2,045.00.

The DX-5 is an FM digital synthesizer with 2 FM tone generating systems, velocity and after-touch 76-note keyboard, 64 memories, 64 additional performance memories, and full MIDI compatibility.
Price: $3,645.00.

The DX-21 is an FM digital synthesizer with 61 keys, 128-voice memory, 2 FM tone generators, 4 operators, 8 algorithms, 8-note polyphonic operation, and built-in chorus.
Price: $845.00.

The KX-5 is a keyboard controller for use with any DX synthesizer, TX-7, or TX 816. It has 37-note velocity sensitive keyboard and transmits on 2 MIDI channels.
Price: $495.00.

The KX-88 is an 88-note master MIDI keyboard for use with any MIDI compatible keyboards. It can also control the QX-1 sequencer.
Price: $1,795.00.

The KX-76 is a 76-key velocity and after-touch sensitive master MIDI keyboard controller. It controls real time performance effects, voice selection, and programming. It has 16 code memory, and three mode function capability.
Price: $1,095.00.

The DX-100 is an FM digital synthesizer with 61 keys, 4 operators, 8 algorithms, 192 preset ROM voices, 24 RAM voices, sophisticated effects, and cassette storage.
Price: $445.00.

The DX-27 is an FM digital synthesizer with 61 keys, 4 operators, 8 algorithms, 192 preset ROM voices, 24 RAM, and controls for voice selection, pitch bend, and modulation wheel with LFO for range in tremolo, vibrato, etc.
Price: $645.00.

The RX-15 Digital Rhythm Programmer is a fully MIDI-compatible drum machine using LSI technolgy and PCM digital sampling. The drum machine can create simple or intricate rhythm patterns and they can be programmed into complete songs and played back with the digitally recorded drum sounds.
Price: $495.00.

The RX-11 is a digital drum machine with 29 percussion sounds with individual outputs for each. The sounds include 3 bass drums, 8 snare drums, 2 rimshots, 2 open and 2 closed high hats, a high hat pedal, four toms, ride and crash cymbals, two hand claps, two cowbells and a shaker.
Price: $895.00.

The RX-21 is a digital rhythm programmer with 9 percussive sounds, stereo outputs, 56 programmable patterns, 44 permanent internal patterns, 3 song memory, MIDI capability, and cassette storage.
Price: $295.00.

The RX-21L is a digital rhythm programmer with 16 Latin percussion voices, stereo and headphone outputs, MIDI IN/OUT, 50 patterns, 4 song memory, and cassette storage.
Price: $295.00.

SYNTHESIZERS

AKAI INTERNATIONAL MUSIC CO.
1316 E. Lancaster
Fort Worth, TX 76102

CASIO, INC.
15 Gardner Rd.
Fairfield, NJ 07006

DURALINE INDUSTRIES
11300 Rush St.
El Monte, CA 91733

ENSONIQ CORP.
263 Great Valley Parkway
Malvern, PA 19355

EUROPA TECHNOLOGY, INC.
1538 W. Washington Blvd.
Venice, CA 90291

MUSIC INDUSTRIES CORP.
105 Fifth Ave.
Garden City Park, NY 11040

THE MUSIC PEOPLE
PO Box 648
W. Hartford, CT 06107

OBERHEIM
11650 W. Olympic Blvd.
Los Angeles, CA 90064

OCTAVE-PLATEAU
51 Main St.
Yonkers, NY 10701

ROLANDCORP US
7200 Dominion Circle
Los Angeles, CA 90040

SEQUENTIAL
3051 North First St.
San Jose, CA 95134

SIMMONS GROUP CENTRE, INC.
23917 Craftsman Rd.
Calabasas, CA 91302

Computer Software

DR. T's MUSIC SOFTWARE

The Keyboard Controlled Sequencer is a music composition and sequencing program for Commodore 64, 128, Apple IIe, and II+ computers. It features real, step, and type-in entry of note information, full editing of all note/event parameters, copy, merge, delete, move, append, transpose, invert, auto-correct, and time reverse commands for complete or partial sequences.
Price: C64-$100.00; C128-$225.00; Apple-$175.00.

The DX/TX Patch Editor-Librarian is for the Commodore 64/128, Apple IIe/II+, and Atari 520/1024 ST computers. It provides full screen edit of all operator parameters, 2 banks of patches in memory simultaneously, bulk dumps, built-in sequencer, and special fast edit mode.
Price: C64/128-$95.00; Apple-$100.00; Atari-$125.00.

The CZ 101/1000/3000/5000 Patch Editor-Librarian is for the Commodore 64/128, Apple IIe/II+, and Atari 520/1040 ST computers. It provides full on-screen editing of all CZ parameters. The Commodore versions have built-in sequencer. Comes with multiple sets of patches on disk.
Price: C64/128-$95.00; Apple-$100.00; Atari-$100.00.

The Echo Plus is for the Commodore 64/128 computers. It is a computer effects generator that features keyboard splitting, doubling, harmonizing, echoing, one finger chords or scales, arpeggiated chords, infinite loops, and pre-programmed patch changes. Eight user-defined presets may be saved to disk.

MARK OF THE UNICORN

(Runs on the Apple Macintosh 512K and MacPlus.)
The Professional Composer is an interactive music notation program that allows the user to create a score of up to 40 staves long. You can connect the staves with braces and brackets, enter and delete all types of musical symbols and lyrics, group notes and phrases, transpose, and change duration.
The Performer is a MIDI sequencer, editor, and performance tool with 50,000+ note capacity, and it allows you to work with 200+ tracks at a time. Beat resolution is 480 parts per quarter note. You can record in real time as well as step record. The software will sync to drum machines and tape interfaces.

MUSICDATA

The MIDI Sequencer II is for the Commodore 64. Features include: single screen real time sequencer, song composer, MIDI event editor, 1/192 resolution, 16 tracks, 63 patterns, 17 delays, two kinds of quantization, MIDI merge function, MIDI position pointer, track transpose, loop, and single step recording.
Price: $175.00.

The DX Editor is for DX and TX synthesizers and runs on the Commodore 64. It is a totally menu driven program containing parameter choice, enlargement of value, edit page, LFO and modulation page, operator page, graphic page, choice of editor memory, choice of color, help menu, bend curve, and choice of two edit memories.
Price: $75.00.

OCTAVE-PLATEAU

The Patch Master is a MIDI software package for the IBM-PC and compatibles. It is a system organizer for MIDI studios, a universal librarian for uploading and downloading patches, performance data, drum patterns, and a bank arranger for creating and arranging banks of sounds. It eliminates cassette interfaces and cartridges.
Price: $149.00.

The Voyetra Voice Editor runs on either an IBM-PC or compatible. It provides complete and easy access to all parameters of Voyetra steps and programs. Full screen graphics editor shows all features and signal paths of the Voyetra, making it easy to understand and use its sound capabilities. All changes to the programs are heard instantly and sounds may be named, stored and loaded on disk.
Price: $79.00.

The Sequencer Plus is a 64-track MIDI sequencer and editing software. Each tracks has its own looping, transposition, quantizing, program and MIDI channel. It offers a View Screen that lets you see up to 72 measures of 22 tracks at the same time, adjustable playback start, automatic display scrolling, complete MIDI event editing, control windows, programmable tempo track, play range, external MIDI sync, and customized displays.
Price: $495.00.

The OP-4001 is an IBM-PC based intelligent MIDI interface. It handles all timing and buffering of MIDI information. It receives or transmits standard 5V clock pulses allowing master and slave sync to most drum machines. It also has an audio click metronome.
Price: $295.00.

PASSPORT DESIGNS

(All Passport products require an Apple IIe, IIc, Commodore 64 or IBM computer, and a Passport MIDI Interface.)
The MIDI/4 Plus allows you to compose, orchestrate and arrange complete multitrack recordings using a synthesizer, drum machine, personal computer and tape recorder. Four separate channels control individual keyboards simulataneously, or the same channel controls several keyboards. It combines unlimited overdubs, real time edting, and tempo control with powerful editing features.
Price: $129.00.

The MIDI/8 Plus has all of the features of the MIDI/4 Plus, above, plus it has sequence chaining, linking, and the ability to merge tracks together on four additional tracks.
Price: $169.00.

The MIDI Player is a computerized music presentation system for MIDI/4 Plus and MIDI/8 Plus recordings. It allows you to arrange and digitally store a whole set of music on computer disk, and play it back through the MIDI system in any order. It syncs music playback with real time computer graphics.
Price: $99.95.

The Leadsheeter is a printing program that prints out sheet music in Treble Clef Piano Score format. The program transcribes directly from any MIDI keyboard in real time with auto correction. Once the music has been transcribed, lyrics, chord symbols, markings and titles can be entered and edited on the screen.
Price: $149.95.

The Polywriter Utilities integrates and links Passport's music printing and recording software. It allows you to take the music you have recorded using MIDI/4 Plus and MIDI/8 Plus and transcribe it into music notation, edit the arrangement on screen graphically, add lyrics, and print out scores. It also lets you playback Polywriter and Leadsheeter files through the recording packages.
Price: $99.95.

ROLAND

The MUSE (MIDI User Sequencer Editor) is a sequencing/editing program designed to run with the Roland MPU-401 interface and any MIDI equipped instrument. It works with the Apple II, IIe, II+ (64K minimum), and Commodore 64 computers. It is a comprehensive system which offers a wide range of functions in a fast and easy format. It can be operated by the diamond pattern formed by the I, J, K, and L keys on the keyboard, or by a joystick or game paddle.
Price: $150.00.

The MPS (Music Processing System) is for the IBM-PC and compatibles. It transforms your computer into a sophisticated music composition, transcription, and printing system that can be used with any MIDI device. It has three basic modes, song, score, and print. Individual measures may be cut and pasted into any order and printed on an IBM compatible dot matrix printer.
Price: $495.00.

YAMAHA

The FM Music Composer Software enters key, time signature, notes, rests, tempo, dynamics commands, voices and other music parameters directly onto score. Plays back music through internal FM synthesizer, MIDI equipped synthesizers drum machines.
Price: $50.00.

The FM Voicing Program Software allows user to edit the CX5M's standard 46 voices or create new ones using the built-in FM digital synthesizer.
Price: $50.00.

The DX-7 Programming Software enables the Yamaha CX5M user to create and edit voices for the DX-7 FM synthesizer. By displaying the DX-7 voice parameters through the CX5M, changes in programming are easily made. It also stores up to 46 voices.
Price: $50.00.

The MIDI Sequence Recorder Software is 4-track MIDI recorder software featuring real time and step recording. It has editing capability that permits modification of recorded tracks and sequential playback of banks with tempo, transpose, track and MIDI channel changes.
Price: $55.00.

The DX-21 Voicing Software enables the Yamaha CX5M user to create and edit voices for the Yamaha DX-21, DX-27 and DX-100 FM synthesizers. By displaying DX voice parameters, changes in programming are easily made.
Price: $55.00.

COMPUTER SOFTWARE

DR. T'S MUSIC SOFTWARE
24 Lexington St.
Watertown, MA 02172

MARK OF THE UNICORN
222 Third St.
Cambridge, MA 02142

MUSICDATA, INC.
8444 Wilshire Blvd.
Beverly Hills, CA 90211

OCTAVE-PLATEAU
51 Main St.
Yonkers, NY 10701

PASSPORT DESIGNS
625 Miramontes St., Suite 103
Half Moon Bay, CA 94019

ROLANDCORP, US
7200 Dominion Circle
Los Angeles, CA 90040

YAMAHA COMBO DIVISION
PO Box 6600
Buena Park, CA 90622

Editor's Correction: The following is a reprinted correction from our January/February Buyer's Guide. We apologize to all concerned for the unfortunate error.

JBL

4691B uses 15-in. cone woofer and has a black finish with a black fabric grill. Impedance is 8 ohms and the frequency response is 40 Hz-20 kHz, -10 dB. Dimensions are 30.2 x 20.1 x 18.8; weight is 109 lbs.
Price: $795.00.

4695B uses 18-in. cone woofer and has a black finish with a black fabric grill cloth. Impedance is 8 ohms and the frequency response is 30 Hz-20 kHz, -10 dB. Dimensions are 40.2 x 29.6 x 18.2; weight is 142 lbs.
Price: $795.00.

4602B uses a 12-in. cone woofer and has a black finish with a black fabric grill cloth. Impedance is 8 ohms and the frequency response is 50 Hz-15 kHz, -10 dB. Dimensions are 20 x 16 x 14.7; weight is 57.25 lbs.
Price: $528.00.

4612B uses two 8-in. cone drivers and has a black finish with a black fabric grill cloth. Impedance is 8 ohms and the frequency response is 60 Hz-22 kHz, -10 dB. Dimensions are 18.5 x 21.5 x 10.25; weight is 45 lbs.
Price: $498.00.

4625B uses a 15-in. cone driver and has a black finish with a black fabric grill cloth. Impedance is 8 ohms and the frequency response is 40 Hz-3.5 kHz, -10 dB. Dimensions are 30.2 x 20.1 x 18.8; weight is 89.5 lbs.
Price: $498.00.

4628B uses a 15-in. cone woofer and a 8-in. cone mid-driver with a crossover at 3 kHz. Finish is black with black fabric grill cloth. Impedance is 8 ohms and the frequency response is 35 Hz-22 kHz, -10 dB. Dimensions are 30.2 x 20.1 x 18.8; weight is 108.5 lbs.
Price: $795.00.

MEYER SOUND

UM-1A is capable of 125 dB SPL and is a biamplified, wedge-type floor monitor using 12-in. cone woofer and 1.4-in. horn tweeter with a crossover at 1.6 kHz. Finish is black with a gray metal grill. Impedance is 8 ohms and the frequency response is 80 Hz-16 kHz, +/-4 dB. Dimensions are 14.5 x 14 x 22.5; weight is 66 lbs.
Price: $2,185.00.

UPA-1A is the same as the UM-1A above, but it is a trapezoid section cabinet. Dimensions are 22.5 x 14.5 x 13.
Price: $2,185.00.

MSL-3 is similar to the UPA-1A above, in it has a trapezoid section cabinet enabling configurations in large scale arrays for high SPL applications. It uses a 12-in. cone woofer and 2-in. horn tweeter with a crossover at 800 Hz. Impedance is 4/8 ohms and the frequency response is 75 Hz-20 kHz, +/-4 dB. Dimensions are 56.75 x 21.25 x 30; weight is 265 lbs.
Price: $4,200.00.

500 Series Loudspeaker System has two matched cabinets and complimentary professional power amplifiers. It uses 15-in. cone woofer and 1.4-in. horn tweeter with a crossover at 700 Hz. Finish is black textured with a removable fiberglass grill. Frequency response is 40 Hz-15 kHz, +/-4 dB. Dimensions are 32 x 20 x 14; weight is 100 lbs.
Price: N/A.

MODULAR SOUND

TA-12C is a two-way time aligned system using a 2-in. cone woofer and 5 x 6 horn tweeter with a crossover at 3.5 kHz. Finish is textured paint with a metal/foam grill. Impedance is 8 ohms and the frequency response is 70 Hz-19 kHz, +/-6 dB. Dimensions are 24 x 18 x 15; weight is 51 lbs.
Price: $495.00.

AFI-VD is similar to the TA-12C above, but it is a three-way system using a 18-in. cone woofer, 12-in. cone mid-driver and 5 x 6 horn tweeter with a crossover at 125 Hz and 3.5 kHz. Impedance is 4 ohms and the frequency response is 40 Hz-19 kHz, +/-3 dB. Dimensions are 42 x 20 x 20; weight is 120 lbs.
Price: $1030.00.

S-15B is a low frequency unit using either a sealed or vented 15-in. woofer. Impedance is 8 ohms and the frequency response is 40 Hz-2.5 kHz, +/-4 dB. Dimensions are 18 x 27 x 24; weight is 80 lbs.
Price: $560.00 for brown oiled birch finish; $380.00 for black texture paint.

S-18B is the same as the S-15B above, but it uses an 18-in. woofer and has a frequency response of 40 Hz-500 Hz, +/-4 dB. Dimensions are 29 x 22 x 20; weight is 40 lbs.
Price: $680.00 for wood finish; $480.00 for paint finish.

Peter Hammar

The Hollywood Exposition: For Us And About Us

Learn more about the non-profit Hollywood Expo!

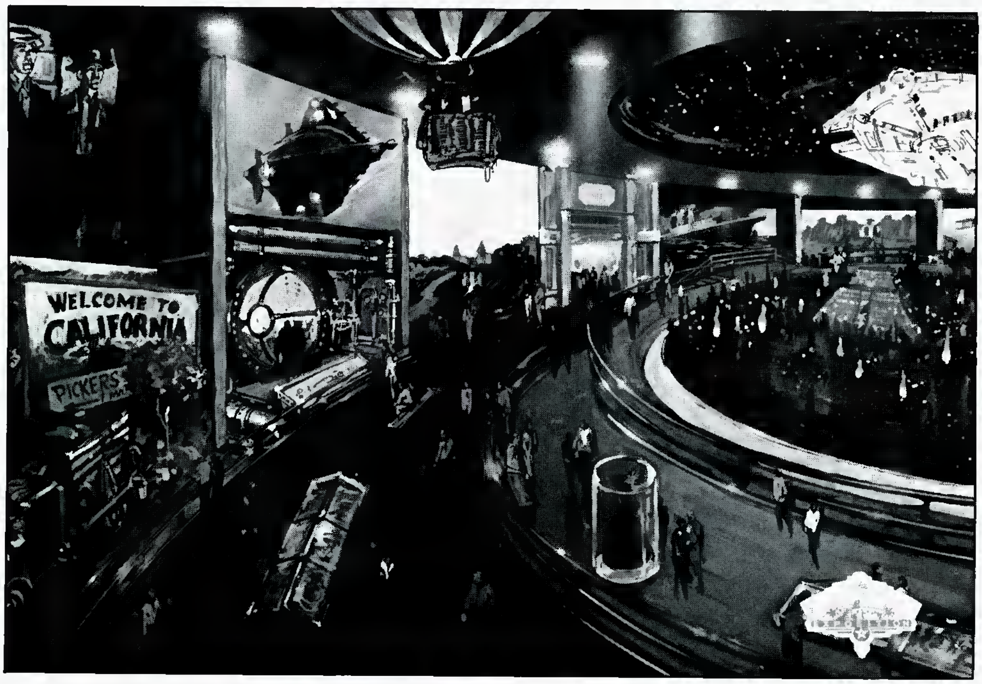


Figure 1. An artist's conception shows the main hall of the Hollywood Exposition, off of which visitors will find four main pavillions representing the allied fields of film, television, radio, and recording.

MUSEUM is a dirty word for many people. As kids, many of us were dragged through those boring, dusty places by well meaning relatives. Today, I hope most of us know better. The word *museum* really means "place of study." To muse. To think.

Museums should be an important part of our fast moving culture, because they can help us put our lives into perspective. Properly done, historical exhibits let us recognize our accomplishments so we can repeat them, and understand our failures so we don't repeat them.

The Gap: We have some good museums in the US—national, state, local, and private—covering vital areas of our lives, such as the automobile, the railroad, the computer, indigenous cultures, and pioneer settlement, to name just a few. All are important, though all certainly suffer from under-funding.

Notice the one museum area I failed to mention: **our field**, the business of entertainment media and technology! I challenge anyone to show me an even reasonably well-funded, non-profit, comprehensive major **exhibit of media history** in this country. Currently, there is not a single existing museum facility in the US that has a realistic chance of staging a **complete and permanent** exhibition of the technology that has become our new cultural heritage of electronic entertainment—records, tapes, films, radio, and television.

An Answer: We have had nothing like that in this country—until now, that is. Due for completion in the early 1990s, the new **Hollywood Exposition** in Los Angeles will offer the nation its first permanent and comprehensive exhibit of the birth and growth of all areas of entertainment—audio recordings, film, radio, television, and video technology. Four major pavillions filled with dynamic displays will lead visitors through a century of technical progress of the machines of entertainment, both production and playback.

The non-profit Hollywood Expo is wonderful news for our industry. The project enjoys major, proven financial sup-

Peter Hammar is the consulting curator for the Ampex Museum of Magnetic Recording in Redwood City, CA.

Editor's Note: The author's statements represent only his own personal views, and do not necessarily reflect the opinions of any institutions mentioned herein.

port, with almost $1 million already received in seed money from the state of California. The museum has made realistic plans to raise fifty times that amount through local, national, and international sources, mostly from private industry, individuals, and foundations. Importantly, the Exposition has become a major part of the revitalization of the Hollywood urban area. Already, developers and investors have included the Expo in the Hollywood redevelopment plans they have officially submitted.

The Hollywood Exposition will cover both consumer and professional entertainment technology. For example, visitors might learn about radio in the early 1930s by seeing a mock-up living room from the time with appropriate consumer products playing period programming. In addition, they would see a complete 1931 radio studio in action, broadcasting the programming using 1930s state-of-the-art equipment and life-size figures representing the stars of the time. Old-time media is not the only area covered. This is "living history," which means that the Expo exhibits also depict the present as it blends with the past and future of our business.

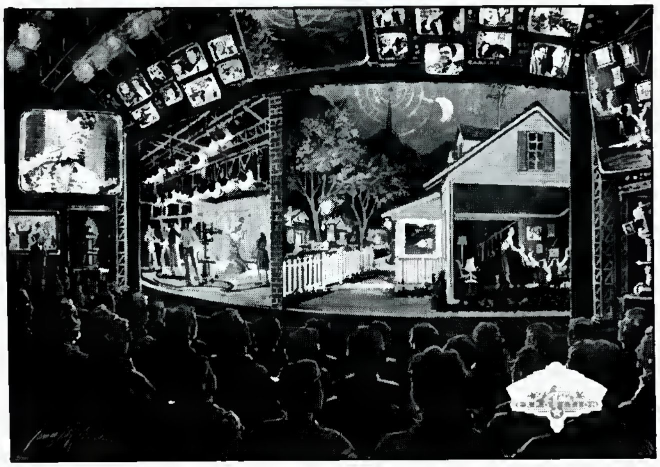


Figure 2. Inside a theme pavillion, Hollywood Expo visitors will be treated to multi-media presentations of the industry's past, present, and even future in film, televsion, radio, and recording.

The Name: Why a museum with the name "Hollywood"? The word is almost a misnomer. "Hollywood," in fact, connotes "entertainment." Naturally, with its Los Angeles location and base of support, the Expo will feature much about Hollywood and its industry. But beyond geographical Hollywood, the Exposition will display the most complete collection of nineteenth and twentieth century entertainment technology ever assembled in one place.

The Hollywood Attic: The Expo will offer something expecially exciting for all media professionals. Plans now include an adjacent "attic" or public access storehouse to display in a very simple manner the thousands of machines, parts, and related documents not on public view. Clearly, *any* museum, no matter how large, cannot possibly display every important artifact. Yet, when the time comes, this museum will certainly be inundated with donated machines, parts, and their documentation.

Most of the Expo's thousands of artifacts eventually solicited will never physically fit in the permanent exhibits in the four pavillions. Museums are usually forced to put their unexhibited pieces out of sight in remote warehouses. But thanks to the "attic" concept, media professionals in the next decade will be able to make the pilgrimmage to Hollywood and its Attic to see, touch, and even operate the machines that helped shape their careers and their lives.

Delay Your Equipment Donations Just Now: Currently, Expo director Phyllis Holzman says she cannot accept equipment donations for both logistical and financial reasons. Partly to avoid mistakes of the aborted "Hollywood Museum" of the early 1960s, Holzman plans to arrange for warehouse space and curatorial services only when the Expo has adequate financing to develop these functions. Acquiring and properly curating artifacts are items high on the Expo's list of early priorities. In the meantime, Expo people hope potential equipment donors will be able to keep and protect their holdings until the museum has a qualified curator and Class A warehouse, where donated artifacts can be properly received, evaluated, catalogued, and stored.

Big Job: We are talking here about a vast quantity of machines and material to put on display in a meaningful way—as well as to realistically fund and endow for the future. The management of the Hollywood Exposition clearly understands the enormity of the task they have undertaken. Most importantly, they are carefully and quietly making their way through the financial and political mine fields that crippled previous "Hollywood Museum" efforts.

While expansive in its scope, the Hollywood Exposition does not see itself in competition with other media museums or archives—planned or existing—around the country. Most people who support preservation efforts in this industry hope that other public and private media museums and archives will see the Expo as an *asset* to their *own* work raising funds and the public's consciousness of the importance of preserving twentieth century entertainment technolgy.

Anyone interested in more information on the exhibit and what help the Expo will need in the future should contact the office of Ms. Phyllis Holzman, Director, Hollywood Exposition, 5555 Melrose Avenue, Hollywood, CA 90038, telephone: (213) 468-5494, or contact this author at the Ampex Museum of Magnetic Recording, 401 Broadway, Mailstop 3A-14, Redwood City, CA 94063, (415) 367-3127. ■

FROM THE EDITOR

db Magazine has been working closely with Peter Hammar and others in California to create a museum that would significantly include the history of pro audio. We are aware of large collections of fine artifacts, including John T. Mullin's fine collection which has been exhibited at past AES Conventions.

We've also visited and reported on the Ampex Museum, which Peter Hammar has created, and talked with Ampex, 3M (where Jack Mullin worked for many years) and others. Each has been urged, and has responded positively, to support what is now about to happen in Hollywood.

All this is by way of saying that our industry needs a museum. We need to be able to give to future generations of audio professionals a strong sense of where their profession has come from. Only with that strong sense of past can a better future develop.

At this time, however, we must echo Peter Hammar's statements of what to do if you have a personal collection you want to donate. Don't ship anything! The Hollywood Expo is not yet prepared to receive anything. But let Peter or us know what you've got, and that will start the ball rolling. As we learn more about the progress of the Hollywood Expo, you will read about it here. LZ

Nicholas Pasquariello

Chris Newman: Recording The Feature Film

Feature film recordist Chris Newman tells us about recording sound for A Chorus Line: The Movie.

In 1968, an obscure documentary sound recordist by the name of Chris Newman was changing planes at the Los Angeles airport returning from a shoot in India, when he was paged over the pa system. The operator instructed him to call Haskell Wexler. Recalls Newman, " I called Haskell Wexler, expecting this to be some kind of fantasy come true. Wexler said, 'I heard a lot of things about you. I'm doing a movie in Chicago; it's called *Concrete Jungle*. Are you interested?' As he was finishing, I overlapped him saying, 'Yes, I'm interested.' I would have done it for nothing. It was a feature film! It was Haskell Wexler, who was a hero to me."

The picture, later retitled *Medium Cool*, was Newman's first theatrical feature job, and proved to be one of the more successful experiments in the newly invented genre of "docu-dramas." *Medium Cool* did indeed mark the point at which Newman made an enormously successful transition from documentary sound recordist to theatrical feature film recordist. In sixteen years he has supervised the location and studio sound recording of twenty-eight feature films under directors the caliber of: Coppola, Forman (three pictures), Friedkin (two Pictures), Ashby, Passer, Pakula (three pictures), and Attenborough. His peers have publicly acknowledged his achievements as an audio craftsman by awarding him three Academy Awards for Best Sound for *The Exorcist*, *Amadeus*, and *Fame*.

Chris Newman describes his early interests as science, engineering, technology, and music. In the late 1950s he was majoring in Metallurgical Sciences at the Massachusetts Institute of Technology. Newman had spent twelve years of his youth studying piano. Initially, he says, he used recording merely as a means to support himself while making short films. (By his own admission, Newman is largely self-taught in his profession.)

In the early 60s, he found himself so successful at recording work that he abandoned filmmaking entirely to devote himself to audio work. There followed television and documentary recording jobs such as *Brimstone: The Amish Horse* (for Disney) and an independently produced documentary on Ravi Shankar shot in India. When he returned from making this film he met Haskell Wexler.

Now Newman has made the popular Broadway stage musical, *A Chorus Line*, into a feature length motion picture under the direction of Richard Attenborough (best known for the film *Ghandi*). *A Chorus Line* was shot in a theater—the Mark Hellinger in midtown Manhattan. As a result of this fact alone, Newman was faced with many interesting (often unique) sound recording problems.

Among the topics Newman explored are: how recording a motion picture in a theater compares with recording on a sound stage; some of the special challenges involved in working with dancers, e.g., use of radio mics, "earwhigs" and boom mics; some of the pros and cons of looping as compared to live recording; some of the advantages of using original talent versus stand-ins in recording Foley effects; as well as his goals as a sound recordist.

db: Would you describe some of the problems you encountered because you were shooting in a theater instead of on a sound stage?

CN: There were mostly problems associated with background noise. The stage was such that there were skylights up above, which were almost impossible to seal up, and noise comes in in any case.

There were many shots in the script that called for the street doors on the side of the stage to be open. Well, fortunately, the cameraman only required a light effect out there. And (director) Richard Attenborough never really saw people making elaborate entrances from the street. So they built a little house out onto the sidewalk—they got permission from New York City—with double insulated walls, stuff like that. And there were lights inside that house. But the truth of the matter is that no matter how well you construct something like that, there's a point of diminishing returns. And you cannot shut out the noise if there's a lot of it. It varies from day to day. Wednesdays are the worst days because they're matinee days. (The Hellinger Theater is located in the theater district of Manhattan.) And you could tell, if you listen in the background, what day the stuff was shot on.

db: That didn't go into the picture, though?

CN: Well, with most of the stuff we were able to either fit lines in from closer takes, and some stuff was looped. And in some cases we were lucky because quiet stuff played far away from that open door.

We also had to contend with dimmer banks that were mounted high on the wall controlling a myriad of lights; and they had fans which were thermostatically controlled. But the electricians were very cooperative. They reset the thermostats so that the fans didn't go on until much more heat built up; and the insulator construction people insulated underneath.

It wasn't perfect and I think we knew going into it that it wasn't going to be like a sound stage. But I thought the tracks that we produced were pretty good, quite good, as a matter of fact. If we had been on a sound stage it would have been much easier but other kinds of problems would probably have arisen. Yet certainly the one enormous advantage you have working on a sound stage is that you never have any serious background noise problems. You might have all kinds of other problems but you don't have background noise.

And that's one of the big problems in mixing a feature film: you try to set up a situation where there aren't jarring changes of background from cut to cut and scene to scene. And the easiest way to do that is not to have any background noise to begin with.

We could have elected to wire people with radio mics and we would have cut down the background substantially. But most of these kids are dancers. Most were wearing very tight outfits, leotards of one kind or another. In many cases they segued from speaking into dancing. And I thought, considering that there were sixteen principals plus Michael Douglas, that it was absolutely the wrong way to go in terms of technique. I thought about it for a long time.

db: Why was it the wrong way to go?

CN: Because it's too inhibiting. If you're shooting on a set, money and time are considerations. One of the things that I get paid for, in addition to making a good sound track, is to execute (my work) pretty quickly. And, if I'm up there constantly fiddling, adjusting, fixing, specing out faulty noise, moving a transmitter because the camera sees it on somebody twirling around—as the boom man (two boom men, in this case) also is; that's a waste of time.

It unsettles the artist. It makes everybody very nervous and it's a negative way to go. If the people were wearing conventional costumes, then it would have been a very serious consideration. And we did use radios occasionally, when we had to.

db: So you used boom mics entirely, then?

CN: Yes. We used Sennheiser shotguns, for the most part. We used an extra long boom made for television work, which is big and cumbersome, and always in the way, but will permit you to make a big reach—a 16-foot reach. I had two very good boom men: Vito Ilardi and Arthur Bloom, who are both first boom men. I also had a terrific playback operator by the name of Neil Fallon who actually was hired for the job before I was hired, although he would have been the person that I would have hired.

When you do a job like this, more so than on a lot of other jobs, you really need to have very skilled people helping you. The reason for that is that there are so many elements. We constantly did playback/live, playback/live; segueing in and out; music cue after music cue; changing cues, making splices and edits in the tape as we went. And, unless you really have good people like Fallon or the boom people, you can't do a good job. You look absolutely like a total fool.

db: Would you describe how you did the playback/live?

CN: A song would be sung on stage and then there would be a stop. And the actor would speak and it would be recorded live, then the music would continue. In some cases we ducked the playback, meaning we just took it out for those two seconds, and then brought it back in again.

db: Was there live music on the stage while the singing was going on?

CN: No. Unlike other movies that I've done—for example, *Fame* or *Amadeus*—there was no singing recorded live. There were lots of discussions about doing some songs live, but the decision was ultimately tempered by the fact that things would only be done live if the respective actors or actresses could not do good lip synch, meaning if they couldn't mime to their playback. Or if Sir Richard wasn't getting a decent performance built into that same aspect of it.

db: Did those eventualities ever arise?

CN: They didn't arise.

db: So let me understand this. You're saying there were no songs recorded live in the theater?

CN: That's correct.

db: So, what was recorded in the theater?

CN: Only the dialogue.

db: No dancing either?

CN: Dancing, that's a whole other story. What we did was: while the production was shooting and when some less important things were going on, I went with part of my crew to another theater with John Bloom, the editor, and Jonathan Bates, who's their supervising sound editor from England. We showed some of the cast people who were available, a video copy of various parts of the first and second edited reels of the picture. And they, in fact, did Foleys. We used a wireless headphone system. We played Q-tracks for them right from the cut sections. And we recorded their Foleys in stereo at this other theater. And those Foleys will be used in the picture.

db: What was on the video copy they were looking at?

CN: They could see themselves dancing.

db: So they did Foley to their own dancing?

CN: Yes, exactly. And even they had trouble remembering their routines! So, if we had brought new people in to do the routines it would have been that much more difficult.

And we were lucky enough to find a theater where the acoustics were very similar to the theater we were shooting in.

db: Was this an accident; how much trouble did you have in finding this theater?

CN: Well, we had one theater which wasn't bad. Then we lost that theater. Then we went down and looked at another theater that someone had suggested—although the person didn't suggest it based on the acoustics, but on the availability and size. It turned out to be almost an exact duplicate in terms of its acoustics to the place in which we were filming. It's called Entermedia.

So we used that theater with great success and spent two terrific days there producing all these effects, which I understand the re-recording people in England, including Jerry Humphries, the re-recording mixer at Twickenham, are just ecstatic about.

db: Have you recorded in theaters before?

CN: Yes, but I've never done an entire picture in a theater, I'm happy to say. I hope to never do another picture in a theater because it's very tricky and difficult. The noise problems are such that people look at you and say, "What

This list contains all of Chris Newman's theatrical motion picture sound recording credits, thus far released. A single asterisk (*) indicates Academy Award nominations for Best Sound, a double asterisk (**) for an American Academy Award, and a triple asterisk (***) for a British Academy Award. The name following date of release is that of the director.

Medium Cool
1969
Haskell Wexler

The Landlord
1970
Hal Ashby

Little Murders
1971
Alan Arkin

French Connection *
1971
William Friedkin

Klute
1971
Alan Pakula

Heartbreak Kid
1971
Elaine May

The Godfather *
1972
Francis Coppola

Cops and Robbers
1972
Aram Avakian

Shamus
1973
Buzz Kulik

The Exorcist **
1974
William Freidkin

The Taking of Pelham **1-2-3**
1974
Joseph Sargent

Law and Disorder
1974
Ivan Passer

Mickey and Nicky
1976
Elaine May

Who'll Stop The Rain
1978
Karel Reisz

Comes a Horseman
1978
Alan Pakula

All That Jazz
1979
Bob Fosse

Hair
1979
Milos Forman

Winter Kills
1979
William Richert

Power
1980
Barry Shear and Virgil Vogel

One Trick Pony
1980
Robert My Young

Fame ***
1980
Alan Parker

Ragtime
1981
Milos Foramn

Sophie's Choice
1982
Alan Pakula

Soup For One
1982
Jonathon Kaufer

Tender Mercies
1983
Bruce Beresford

Beat Street
1984
Stan Latham

Amadeus **
1984
Milos Forman

A Chorus Line
1985
Richard Attenborough

could you possibly be hearing? We don't hear any noise." They always say that to soundmen, anyway, because everyone knows that soundmen are totally crazed. Well, the truth of the matter is that the noise is there and you just don't hear it. It's not like going outside or being in a noisy apartment. What's maddening about the theater is that you could never tell where the noise was coming from!

db: Well, it's a huge space.

CN: It's a huge space. And, for example, the air conditioner would go on in an adjacent building to cool off a restaurant and you absolutely go nuts.

db: How many of these problems did you anticipate?

CN: I predicted the noise. My feeling was that we would get away with it and we did in all but ten percent of the cases. In most situations, in addition to having the directional microphones and people who knew how to handle them, we always had actors who were projecting with a fair amount of voice. They were playing from the stage top Zack, who sits at some distance from the stage. And we could rely on that.

We could also rely on the editorial department to take medium and close-up tracks and fit them into extrememly wide shots, where people were very, very small in the frame. And since John Bloom, the editor, was cutting as we went, we knew pretty much what we needed, what we didn't need, what was going to be fixed, what wasn't going to be fixed.

When you have that kind of grown up approach to filmmaking where no one is afraid of what they're doing, then you're in a position to deal very openly and say, "We screwed up on that one but we can use that track in there. And this one we have to loop, and this one will be okay if Rick at the transfer house noise surpresses it." It's only when you have a situation where people are very unsure of themselves or put off things, that it becomes very difficult to figure out what to do.

db: You say there was no music played in the background while the acting was going on....

CN: That's not entirely true. What happened was...I don't know if you're familiar with some of the songs in *Chorus Line*...but with some of them there are kind of "vamps" or musical interludes that go on while that are soliloquies. In some cases it's necessary for the actors to hear the tempo to get back into tempo or into key when they begin to sing. But, for the most part, since we want to try to preserve the dialogue of the soliloquies in between, we would either take the playback down to an almost inaudible level and hope that their voices would drown it out, which it did in some cases and didn't in others; or you'd take it out completely. Mike Tronick, the music editor, would count off stage or wave his hand off stage. Then he would direct the playback operator to bring the playback back in as close to the beginning of the playback cue as possible so we could keep as much dialogue as possible. The way I see it is my job is to preserve as much of the live dialogue as I possibly can.

In some cases we would use something called a "thumper," which is basically a digital metronome which, if keyed by a variety of things on the playback track will keep the tempos of the music and will produce a very, very low frequency thump on the stage. I went out and I bought some sub-woofers which were mounted on the edge of the stage. It's very upsetting for some people to listen to—it sounds like a heart beat.

What will happen is you will hear the music, the music will go off, then what you'll hear is: "Boomp, boomp, boomp." In some cases, dialogue was going on and people were dancing to the tempo of the thump. So we had to give the dancers something on the stage (to dance to) because we see their legs moving and it's very hard to fake that kind of dancing unless that have something to motivate them. On the other hand, you don't want to screw up the dialogue, totally, by having to loop a three page scene.

db: So the purpose of the thumper is to motivate the dancers?

CN: Yes, and to keep them in tempo. Then, and this is all quite hypothetical, then in the post-production, even though the thump is on the track in some cases, under the dialogue, if you put the music back in on top of the thump, the thump becomes part of the rhythm track of music.

db: You don't hear it?

CN: You hear it as part of the music so you don't know what it is. Suppose, for example, there was a solitary bassist on the stage and he was playing a very low note in tempo and the dancers were cueing off that low note. Or suppose there was a kick drum and he was just going: "Thump, thump, thump," keeping time for them. Suppose, when the thing is mixed, you bring in either a piano or a rhythm section, for example. The minute you bring any kind of music back in, you obliterate the thump. The thump becomes part of the musical elements.

db: It doesn't interfere?

CN: There are times when it worked and times when it didn't work. Most of the time it worked. We only used it three or four times. From the reports that I get it seemed to work very well. I think it's just another technique for shooting dialogue with music going on.

db: How would you decide when to use the thump and when not to?

CN: We made educated guesses. The criteria was: could we hear it and how loud was it in comparison to the dialogue. And in a couple of cases, since I was making two-track recordings because I always work with a stereo Nagra, all I was able to do was take a direct feed from the playback track and feed the playback in on another track and listen to the dialogue rehearsal and listen to the playback to see if I could hear any thump. If I didn't hear any thump or if it was way down under their voices, then we assumed it would work. And it did.

And the other criteria was whether the dancer could keep time to it. See, one of the problems of the thumper is that the attack of the thump on a big stage is not very sharp. It's not like a click track. It doesn't have the bite that the leading edge of the click has. It's very dull. And the only way to make it sharper is to raise the frequency. But the higher the frequency the more audible it is; also you have to make it louder. So somewhere there's a compromise between how high you make it and how loud you make it. Betweeen Jeffrey Monaday and Mike Tronick and all of us working with the dancers and rehearsing them, we were able to get them to be sharp enough in the background, so that they, in fact, were dancing in tempo and a dialogue was going on in the foreground.

db: What's the frequency range?

CN: From about 25 Hz to about 400 Hz.

db: What was the approach to the pre-prodcution planning on *Chorus Line*?

CN: It consisted of exactly the sort of discussion we're having now. It consisted of discussing with Sir Richard what we are going to do live, and what we are going to do with playback. It consisted of discussing with Sir Richard and the music people and Geoffrey Hanaday all of these things; we are going to do this dialogue live. We are going to do this dialogue to playback. We are going to do this line to playback. We are going to try to use the thumper there. We are going to try to use earpieces on "Hello 12," because people have imaginary thoughts when it's not a question of tempo. It's a question of mouthing words while other people are speaking.

I must say that in every case Attenborough was very patient and very considerate of our needs when we used the earpieces, the so-called "earwhigs," which we used a great deal on *Amadeus* and *Fame*. They are little inductive earpieces; they go in the ear canal and you run a loop of wire around the stage. The wire becomes an analog of a speaker. Anything induced in the loop is picked up in the earwhig.

db: When did you use these earwhigs?

CN: It's all right to use the thumper if people are dancing but if people are singing...let's say you have a situation as in the song "Hello 12," let's say some people are singing while people are speaking. In that case it's not possible to do a playback and a live dialogue recording unless one wants to dub the dialogue. If you use the earpieces, then the people who are singing can hear that they have to mouth the words to create the impression that they're singing and you can still record the dialogue.

And then we decided to use them; I think we put out twelve or fourteen and that's quite a large number. We only did it once, on that song. We had planned to do it on that song. I had seen rehearsals of that song and knew pretty much how it was going to be shot, and I knew how many earpieces we'd have to have with us. But that usually happens when you use the pieces. We set up; some of them didn't work, for some of the people it was too loud or too soft. It always takes a few minutes to sort that out. And Attenborough was amazingly patient. Another director would have said, "Forget it; let's move on." He said, " Okay, let's take a few more minutes." I told him what the problems were. And the few more minutes paid off.

db: Well, the point is it's intended to save you money somewhere else.

CN: Yes. And also a certain amount of intangibles because if you cost out what looping costs versus prodution time, there is no question that looping is cheaper than production time. But there are other elements involved: if you break your neck and have actors break their necks to produce really terrific perfomances of the moment, why are you bothering if you're going to loop. So, somewhere there is a balance between how much you loop and how much you do live depending on the rigors of the show, the actors, and the production requirements.

db: How much looping did you do?

CN: Less thanten percent. Most of the time we do less than ten percent; you just have to fight. You can't be a soundman and do location work and not be assertive and not fight for good sound tracks. ■

Sammy Caine and Peter G. Marzulli

The Alan Parsons Project Concepts In The Studio

Alan Parsons and Eric Woolfson on the recording of Stereotomy.

Photo by David Garr.

IN THE EARLY 1970s, the music industry experienced a proliferation of so-called concept album projects. These albums—Pink Floyd's *Dark Side of the Moon* and Yes' *Close To The Edge* being among the most well known—are somewhat more uncommon in the high-tech 80s. Most groups are content to have airplay, or to be "true to their art," and only a few are still conceptualizing these now uncommon concept albums.

Perhaps the most well known creator of concept albums is and always has been the Alan Parsons Project.

The core of the Alan Parsons Project, producer/engineer Alan Parsons and executive producer Eric Woolfson, first met at London's legendary Abbey Road Studios in 1973 when they were in different rooms, working on different projects at the time. Soon after they met, Eric began to act as Alan's manager. But this relationship shifted in a creative direction, as both were interested in pursuing more creative and imaginative projects. Since then, the Project has released ten albums in ten years and have become cult heros to some.

Yet, exactly what makes up the Alan Parsons Project?

In essence, it's a cross section of the best English studio musicians alive, a plethora of lead singers—which sometimes includes executive producer Woolfson—and the partnership of Eric and Alan.

More importantly though, Alan is not (primarily) a musician. He is the production and engineering genius behind the Project—as well as a few other notable records, among them Pink Floyd's *Dark Side of the Moon.* This album, as well as being a classic in its own right, holds the distinction of remaining on *Billboard*'s "Top 200 Albums" chart for over twelve years—longer than any other record.

However, in spite of all that success, the faces of Alan and Eric are unfamiliar to most—fans and critics alike. Sure, there have been videos, but neither Eric nor Alan have appeared in any of them. In fact, this was only the *second* time the Alan Parsons Project was in the US to do interviews.

We spoke to them just after the release of the tenth APP

Sammy Caine and Peter G. Marzulli are engineers/producers and principal owners of the New York-based Fantasy Productions.

album, *Stereotomy*, which, at press time, continues to climb the charts.

db: Are you (Eric and Alan) partners?

Eric Woolfson: It's a fifty-fifty partnership. The Alan Parsons Project is a combination of Alan's production and engineering and my compositions—although we each encroach into the other's area. The demarkation line is really the glass in the studio—he's on one side and I'm on the other.

db: You were among the first artists to use digital recording and you probably had one of the first Fairlight units...

Alan Parsons: We had one of the first units in the UK. Peter Gabriel also had one of the firsts.

db: What made you first decide to use digital recording? There really wasn't a lot to listen to and say, "Wow, this is the way to record."

EW: The first album we recorded digitally was *Eye In The Sky*, but we only really discovered the real benefits of digital when we recorded *Stereotomy*. That's when we found out what the *real* benefits are.

db: And what is that?

EW: We had always noticed previously that when you recorded a track, there was a certain magic about it. When you got the right take something happened, something lively—something alive. And as you did overdubs and went over and over, by the time you mixed the thing, it never quite bore any relation to that early excitement. When we worked with digital, we found out that there were no changes at all, all the excitement stayed. And we got to realize that there is actually a physical deterioration on that tape. On analog tape, each time it goes over the heads, something is lost.

So we discovered through actually working with digital that the real benefit is that you never lose the excitement. And that is something that until people try it for themselves, they're really not going to realize. Obviously, you don't get this by just mixing on digital, its got to be a full digital recording. And that makes the compact disc a thrill, from the sound point of view.

db: So how do you feel about the people that put down digital? The ones that claim that the digital medium doesn't provide the "warmth" they are looking for.

EW: Well, if they insist on recording something in analog so they get the warmth, let them record analog and then transfer it to digital. You aren't going to lose anything. The advantages really outweigh the disadvantages.

It is also a fear of the unknown and it is irrational. Ultimately people will start to realize the advantages. It's just resistance to change, people just don't feel comfortable with new technology.

db: Let's talk about the actual recording. How many overdubs do you usually do?

AP: Oh lots, but the basic foundation and rhythm section would already be there. Usually we say, "Okay that's it, everybody go ahead and double track now."

db: Do you use MIDI extensively?

AP: No, not really. We used some MIDI. We use the Phrophet 2, Emu II, PPG, and a few others. And we use a lot of clock driving on some of the synthesizers, but we really don't use a lot of MIDI.

EW: If you use too much MIDI, you actually make the sound smaller.

AP: There is the right way and the wrong way of hooking up MIDI equipment. You shouldn't daisy chain, you should go through a "through" box and use a star configuration.

EW: What I like is to use a MIDI pickup on the grand

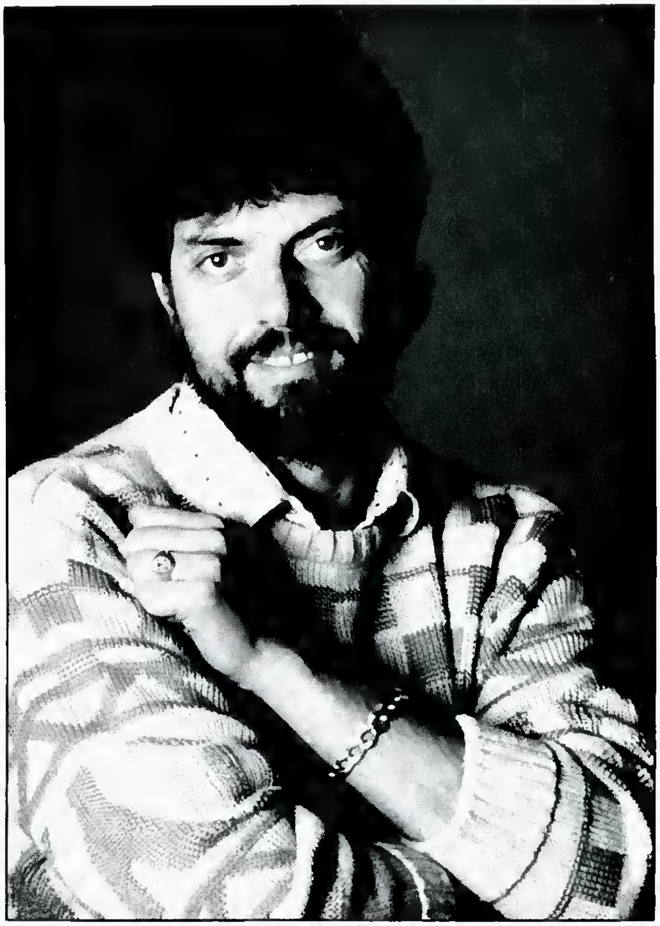


Photo by David Gahr.

piano. I can just sit and play the piano and I can trigger whatever I want. A big problem I always found when playing an acoustic piano along with a backing track is that it had to be so muffled for sound separation purposes so I could never quite hear what I was doing, acoustically. And I'm not too keen on being surrounded by headphones all the time either.

db: What about outboard gear?

AP: We use the usual array of outboard effects, AMSs, pitch changers, delay units, etc. Recently I have been getting into the Yamaha REV-7. I think it's wonderful. For the money it's just beyond belief. I'm building my own studio now and I'm not going to use anything else. I've got the Quantec, the REV-7 and the Roland SDE-3000 and that will do for me for echo.

db: What about drum sounds?

AP: Drums are Simmons and samples. There is only one track that isn't sampled, but that is the track we use for all the samples.

db: Tell us about the home studio you are building.

AP: I'm building a studio at home because I'm hoping to get involved with a lot of different outside projects. Basically, the idea of my studio is to save money on recording. I have put a lot of money into digital equipment, so I better get some of it back! I am not going to keep paying out money to studios. The only work that will be done in my studio will be things that I'm involved with. I don't want anything else going on there. It's being set up as a residential studio. It's a good sized control room, and the studio is really only an overdub room.

db: What kind of equipment do you plan to use?

AP: I'm going to use an Amek console. They've got a new series of consoles coming out and I'm going to buy their new

Tales Of Mystery and Imagination
20th Century Fox
1975

I, Robot
Arista
1977

Pyramid
Arista
1978

Eve
Arista
1979

The Turn Of A Friendly Card
Arista
1981

Eye In The Sky
Arista
1982

The Best Of The Alan Parsons Project
Arista
1983

Ammonia Avenue
Arista
1984

The Vulture Culture
Arista
1985

Stereotomy
Arista
1986

assignable console. The tape machines are two Sony 3324s, which we have already. We recorded *Stereotomy* on them.

db: Why didn't the APP ever tour?

EW: The difficulty with touring is that Alan's function is very hard to portray on stage. I mean, he would just stand there and mix for four hours. It would be pretty boring. I think there will come a time when we feel it's right to tour. Up until now we had most of our time taken up by recording, and we had a lot of contractual dates we had to meet. In the near future after our contract with Arista expires—after one more record—we may have a little bit more freedom to explore touring.

db: Alan, how did you initially start working at Abbey Road Studios?

AP: I started my first job with EMI just after leaving school. It was at their record and tape plant in Hayes, which is outside London. I worked in the tape duplication department making three 3/4-in. mono reel to reel tapes. Cassettes were just beginning to appear. While I was there I got to hear the Beatles' *Revolver* and *Sgt. Pepper* albums on the hi-fi. That was what started it all. I had to satisfy my own curiosity not only of what the Beatles were like, but what a studio was all about. So thankfully I applied to Abbey Road at the right time and I landed myself a job. I first tried through the proper EMI channels and they said, "Oh you'll never get a job there," and I just wrote a letter and then poof, I had a job.

db: What was the first Beatles LP you worked on?

AP: *Let It Be.* Most people think it was recorded *after* the *Abbey Road* album, but it was recorded over a year before that and it just sat on the shelf until Phil Spector got a hold of it. All the Beatles' work was done about a year before Phil even started working on it.

db: What was working with George Martin like?

AP: Oh, great. I've always had the greatest respect for George. I learned as much from him as I did from anybody. Not only the techniques of recording, but the way to handle people. If somebody could handle the Beatles, he must have had something going for him. And he commanded their total respect. Every one of them. That's why Paul and he are still working to this day.

db: What were some of the unusual techniques used on those Beatle recordings?

AP: There were an awful lot of tape machines running. There were machines down the corridor and wires trailing everywhere. Every available machine in the building was in use. Just for sounds. There were no digital delays back then so every sound you wanted to put a delay on was another tape machine. There was no other way around it. And the time it took to line them up was just hours and hours. Head alignment and everything. You had to line them up once every two or three hours.

db: Back to your own recording: do you usually record live? What are the techniques you use?

AP: We get everybody playing at once. We get everybody out there hammering away. You *do* end up with a sterile result if you lay down a drum machine, then try to put the drummer on top then add the bass, and then the guitar. I like to give everybody their own part that they can develop by listening to everybody else.

db: Do you do extensive pre-production work?

AP: We do none. I don't even know what Eric has come up with until day one.

db: Do you feel that there is any overlap or conflict in doing both the engineering and the producing on all your albums?

AP: Not really. I'm still the producer, although I am billed as an artist, I am really a producer. Occasionally Eric criticizes me for spending too much time on one instrument. That's the only conflict, something that I've actually composed at home and I am trying to get the same result in the studio as I did at home.

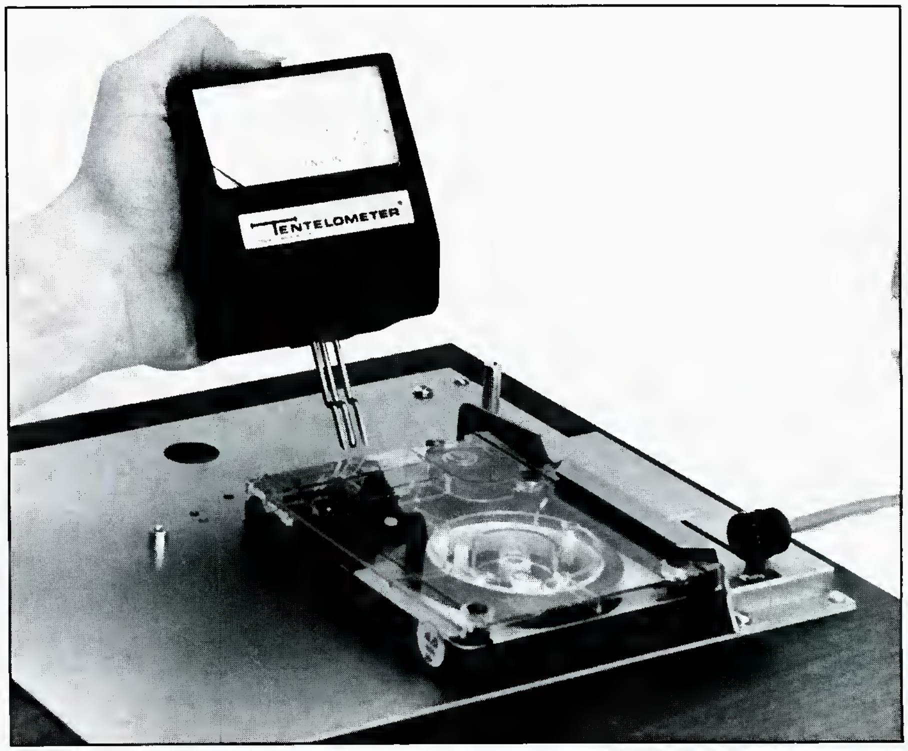
EW: There is a big advantage in that one doesn't have a communication problem of any sort between the engineer and the producer if they're the same person.

db: It has been said that production techniques—particularly yours—are similar to directing a movie.

AP: The only reason, I think, that that applies to us more than anybody, is the same way it was with Steven Spielberg's *ET*. It is Alan Parsons' project. I am sort of controlling the strings on the puppets, as it were. That's why. Spielberg doesn't appear in his films, I don't appear on my records.

EW: Hitchcock liked to appear in his movies in a small cameo. Alan likes to do the same thing on his records, he always likes to do a little something. ■

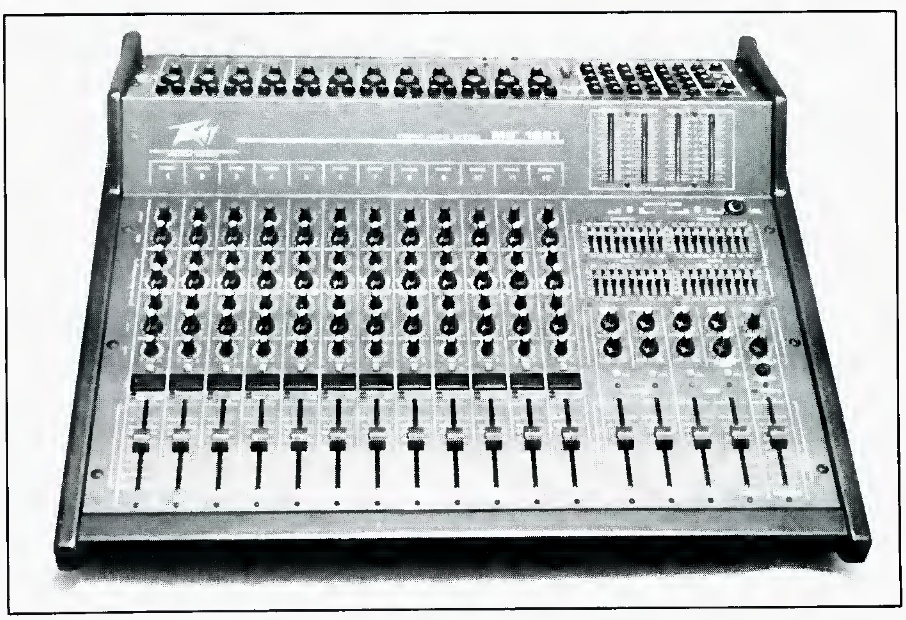
TENTEL CART GAUGE



● Tentel has developed a special "offset probe" version of the Tentelometer tape tension gauge for use with broadcast cartridges (carts). The special "offset probes" on the T2-H7-AC can be inserted over the tape inside the cart, to measure the inherent tape tension of each cart. Tape tensions above three ounces can cause splice breakage, wow and flutter due to slippage at the capstan, premature oxide sloughing, and faster head wear. Use of a tape tension gauge may put an end to air time gaps caused by cart failures. The gauge can be used with a broadcaster's extra cart machine; a low cost deck suitable for checking tape tension is available from Tentel. The T2-H7-AC can also be used for holdback and take-up tensions on 1/4-in. reel-to-reel machines allowing easy trouble shooting of dynamic problems.
Mfr: Tentel Corp.
Price: $345.00.

Circle 51 on Reader Service Card

PEAVEY STEREO MIXERS



● The Peavey MS 1221 12-channel and MS 1621 16-channel stereo mixers use the latest generation of semiconductors and innovative circuit designs. Features include: switchable, 48-volt phantom power in groups of four channels; newly developed equalization circuitry featuring sweepable (semi-parametric) mid range; two monitor and two effects sends per channel; long throw (100 mm) conductive polymer slide faders with custom contoured Soft-Touch control knobs; full channel patching capabilities and comprehensive master patch panel for total flexibility; built-in reverb and electronic delay effects; built-in electronic crossover; and four fully patchable 9-band graphic equalizers. The mixers also incorporate transformer balanced outputs for mains and monitors and have assignment capability of reverb, delay, and effect returns to Monitor A and B buses.
Mfr: Peavey Electronics Corp.
Price: MS 1221-$1,799.50; MS 1621-$2,099.50.

Circle 52 on Reader Service Card

BGW POWER AMP

• The BGW 750 Series Professional Power Amplifier is designed to specifically handle 2 ohm loads and deliver as much as 850 watts per channel IHF dynamic power. Low feedback, all discrete circuitry, and greatly increased damping factor result in improved sonic accuracy. The new 750 Series are 7 dB quieter and at the same time have 4 dB more gain than the previous models. Exclusive BGW Ultracase power devices with their heavy solid copper header provide improved thermal properties and five times greater bandwidth. The LED VU meter has 60 dB dynamic range with twenty-three LEDs per channel. Eight status indicators monitor all important functions including power line voltage. Any abnormal condition is reported on the front panel display. The Model 750D is identical to the higher priced 750E, except for the substitution of modulation and true loss of feedback clipping indicators for the display.

Mfr: BGW Systems
Price: 750E-$1,699.00; 750D-$1,499.00.
Circle 56 on Reader Service Card

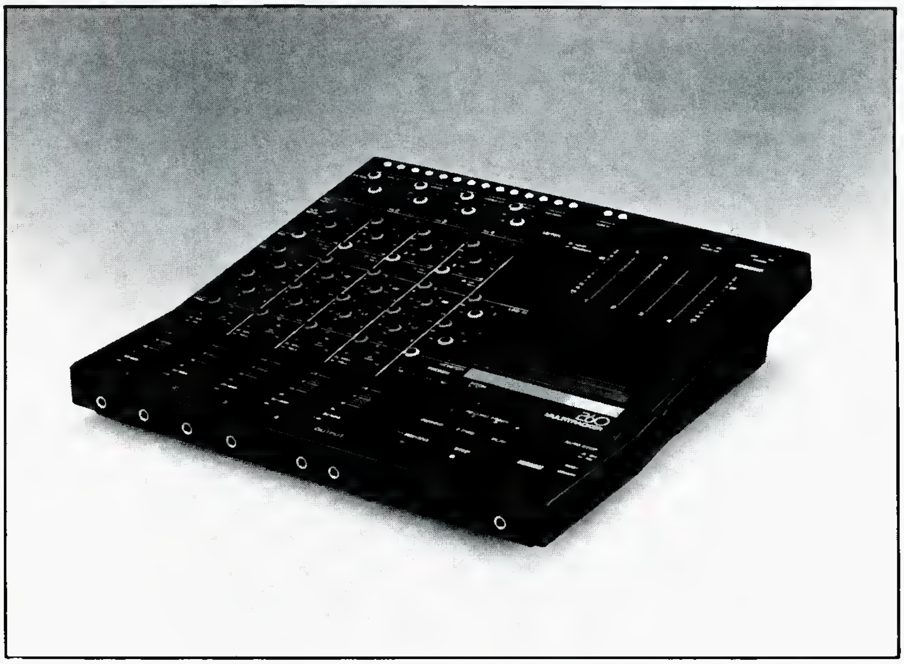
AKG EFFECTS UNIT

• AKG's Aurora Digital Reverb and Effects Unit Model ADR-68K is a powerful computer, capable of performing a variety of tasks which outboard studio equipment is used for. They include: reverb, sampling, special effects, eq, and others. The Aurora will be the object of extensive program development. Improved versions of currently popular sounds will be available, as well as unique programs breaking new ground in reverb, effects, and sampling processing. The Aurora uses a 68000 microprocessor, 16-bit analog to digital conversion and 32-bit internal precision to provide quiet, impeccable signal quality. Bandwidth is switchable between 15 and 20 kHz. The unit features: full MIDI implementation, including register recall and parameter control; 2 in 4 out design allowing you to split the Aurora to run two different programs at once; full function remote unit; and removable RAM cartridge to double parameter memory. The unit also features 16-bit, high quality sound-sampling capability. Four sounds up to eight seconds in length may be triggered independently or simultaneously (via MIDI or trigger inputs), and may be reverberated in the unit at the same time as well. It's possible to sample both a bass drum and a snare drum into the Aurora, plus have them reverberate in different rooms. The Aurora is completely software based, so that it can never be outgrown. New programs and sounds will be generated for it on an ongoing basis. There will also be a special magazine sent to all users at no charge, sharing ideas and applications.

Mfr: AKG
Price: $3,995.00.
Circle 57 on Reader Service Card

• Note: Ursa Major, who actually makes the ADR-68K, is now a division of AKG. Products may now have the AKG name on them, although existing stocks of Ursa Major products are undoubtedly on dealers' shelves.

FOSTEX MIXER/RECORDER



• Fostex Corporation's new multi-track cassette/mixer, the Model 260, features six inputs and an independent stereo bus. It has four mic/line inputs, plus two additional line inputs which are suitable for tape returns. They are also useful as stereo effects returns during mixdown. Each input channel of the 260 has a slide fader, mute button, trim control over a 50 dB range, parametric eq, track assign direct or to the independent Stereo Master Bus, two Aux Send controls for effects, and Monomix Pan and Gain controls for a separate, independent stereo mix of the 4-track cassette tape (cue). Tape speed is 3 3/4-in./sec. for extended frequency response. Dolby C noise reduction is built-in for greater headroom and noise-free recording. Other features include switchable LED bar graph meters, automatic monitor switching for true "rolling" punch-ins, automatic stop with two position memory, and convenient top panel patch points for directing signal flow quickly and easily.

Mfr: Fostex Corporation of America.
Price: $995.00.

Circle 58 on Reader Service Card

SCHOLZ GUITAR PROCESSORS

• The Rockman Sustainor guitar processor combines a wide range of ultra low noise clean and distorted sounds that are footswitchable through two independent channels. The Auto Clean enables you to go from powerful distortion to clean sound without losing output volume or high frequency response, as you do when you turn down the guitar volume. The Smart Gate automatically removes high frequency noise, adjusting its own cut-off time—quickly for staccato notes and slowly for notes held for infinite sustain—never cutting off the end of a note. The Phase Notcher allows reproduction of the phase cancellation patterns of mic'ed multi-speaker cabinets. You can record direct or go straight through a pa and still get the mic'ed sound. The multi-faceted output section features a master volume control for each channel, including a rhythm volume slider, two High Level Effect loops, and a Treble Boost slider to match the Sustainor to any amp or mixer.

The Rockman Stereo Chorus/Delay combines the Rockman Stereo Chorus sound and a delay section adjustable 20 to 200 milliseconds for echo effects. The LED indicated Sweep Speed controls the pace and intensity of chorusing, and includes the new Long Chorus effect. The Feedback slider determines the number of repeats from one to infinity or runaway. Long delays are filtered smoothly to assure correct frequency response for any given delay. Input/Output levels are matched with one slider on the front panel while adjusting the Adjustable Drive Level maintains unity gain and 90 dB signal-to-noise ratio. Together the Sustainor and Chorus/Delay offer six footswitchable controls. The units are available separately, or together with a 19-in. single rack space unit.Mfr: Scholz Research & Development, Inc. Price:Rockman Sustainor- $349.95; Rockman Stereo Chorus/Delay-$269.95.

Circle 59 on Reader Service Card

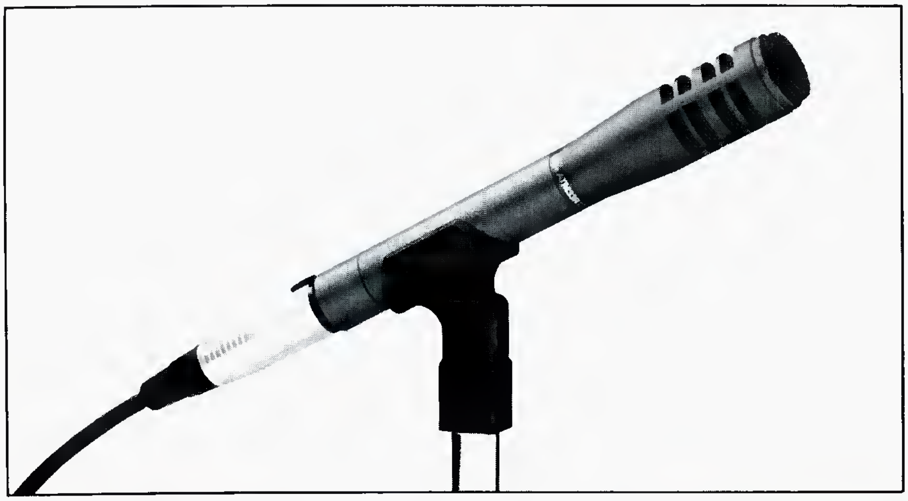
SYNTOVOX VOCODER

• The Syntovox SPX 216 vocoder/effects processor embodies many features that have only been available on expensive laboratory instruments, and some that have never been offered before. The SPX 216's speech input circuit includes a low-noise mic preamp, variable high pass filter, compressor, and a total of fourteen precisely calibrated band pass filters for combined intelligibility and smooth sound. The carrier (instrument) input may be any tone-producing instrument; the unit's self-contained voltage controlled oscillator or noise source, or the speech input itself. The fourteen carrier filters, calibrated identically to those of the speech input, are alternately connected to the left and right outputs to produce true stereo sound, even from a monophonic source. Advanced control features include single button formant shift that raises or lowers the entire formant structure; external control of the sets, compressor output and expander inputs; and footswitch- or computer-controlled external patch control to simultaneously activate all control outputs. A demonstrative cassette is available.

Mfr: Synton/Holland (Big Briar).
Price: $1,395.00.

Circle 53 on Reader Service Card

A-T CONDENSER MIC



• The Audio-Technica A-TM33R is a phantom-powered unidirectional condenser microphone recommended for recording uses, and it is also useful for broadcast and sound reinforcement applications. The reponse of the ATM33R is 30 Hz-20 kHz, and has a moderately rising high end. It is designed for inconspicuous hand or stand use and it measures 7-in. long. The head diameter is 1 1/64-in. and the handle diameter is 13/16-in. The weight is 4 3/4 ounces.

Mfr: Audio-Technica US, Inc.
Price: $250.00.

Circle 54 on Reader Service Card

AUDIO/DIGITAL DIGITAL DELAY

• Audio/Digital has introduced the ADX-2000 Digital Delay Processor and digital delay modules which automate delay control for distributed speaker systems. Sophisticated microprocessor control and nonvolatile memory storage for delay settings allow the user to quickly reconfigure the entire delay system, which may include up to six input channels, and forty output channels. The ADX-2000 is particularly useful for installations where delay settings will be changed frequently, including convention facilities, touring sound systems, and theme park exhibits. A user can key in and store settings for twelve complete system configurations, then recall any configuration for use in seconds. In addition to delay functions, the modules perform channel on/off and output gain functions. Delay settings range from 10 microseconds to 261 milliseconds (1048 milliseconds optional) in increments of one micro- or millisecond. Dynamic range is at least 100 dB (A-weighted, no companding); THD plus noise is better than 0.1%; and frequency response is 20 Hz-20 kHz. Additional modules which will further broaden the ADX-2000s capabilities are forthcoming.

Mfr: Audio/Digital, Inc.
Price: $3,500.00 for standard configuration.

Circle 55 on Reader Service Card

FIELD PRODUCTION

A special job demanding specialized products.

For your audio needs: a growing line of compact, easy-to-use FP amps and mixers.

Shure FP products are built specifically for ENG, EFP, film, and video work. They're not general audio products that "might" work on remotes. And no one offers as wide a selection with this kind of built-in ruggedness and reliability.

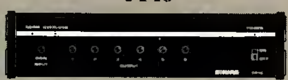
FP32

FP42

For Stereo Remotes. The FP32 Stereo Mixer is comparable in size and features to our famous FP31. Its stereo capability, light weight, easy-to-use controls and convenient shoulder harness make it the first choice of field crews. Our FP42 Stereo Mixer simplifies mic cueing, so important in situations like sports remotes. Plus it enables you to easily mix down stereo in your post production booth. It offers all the features of the popular M267 plus stereo capability and a stereo headphone amp.

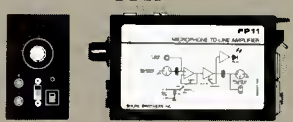
FP31

FP16



The Industry Standards. The FP31 is Shure's original field production mixer. Thousands bet their audio on it worldwide. The FP16, a one-by-six distribution amp with transformer balancing and link jacks, outperforms all competition. It's also ideal as a portable press bridge.

FP11



FP12

For Long Yardage Situations. The FP11 Mic-to-Line Amp provides freedom from noise in long line situations, with up to 84 dB of gain in 15 6-dB steps. It converts any mic to line level and includes an invaluable limiter circuit. The FP12 Headphone Bridging Amp is a must for shotgun and boom operators. It keeps them on target without need for a return line. It's ideal for multiple headphone feeds, troubleshooting, and as an intercom.

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SYNTOVOX VOCODER

• The Syntovox SPX 216 vocoder/effects processor embodies many features that have only been available on expensive laboratory instruments, and some that have never been offered before. The SPX 216's speech input circuit includes a low-noise mic preamp, variable high pass filter, compressor, and a total of fourteen precisely calibrated band pass filters for combined intelligibility and smooth sound. The carrier (instrument) input may be any tone-producing instrument; the unit's self-contained voltage controlled oscillator or noise source, or the speech input itself. The fourteen carrier filters, calibrated identically to those of the speech input, are alternately connected to the left and right outputs to produce true stereo sound, even from a monophonic source.

Advanced control features include single button formant shift that raises or lowers the entire formant structure; external control of the sets, compressor output and expander inputs; and footswitch- or computer-controlled external patch control to simultaneously activate all control outputs. A demonstrative cassette is available.

Mfr: Synton/Holland (Big Briar).
Price: $1,395.00.

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A-T CONDENSER MIC

• The Audio-Technica A-TM33R is a phantom-powered unidirectional condenser microphone recommended for recording uses, and it is also useful for broadcast and sound reinforcement applications. The reponse of the ATM33R is 30 Hz-20 kHz, and has a moderately rising high end. It is designed for inconspicuous hand or stand use and it measures 7-in. long. The head diameter is 1 1/64-in. and the handle diameter is 13/16-in. The weight is 4 3/4 ounces.

Mfr: Audio-Technica US, Inc.
Price: $250.00.

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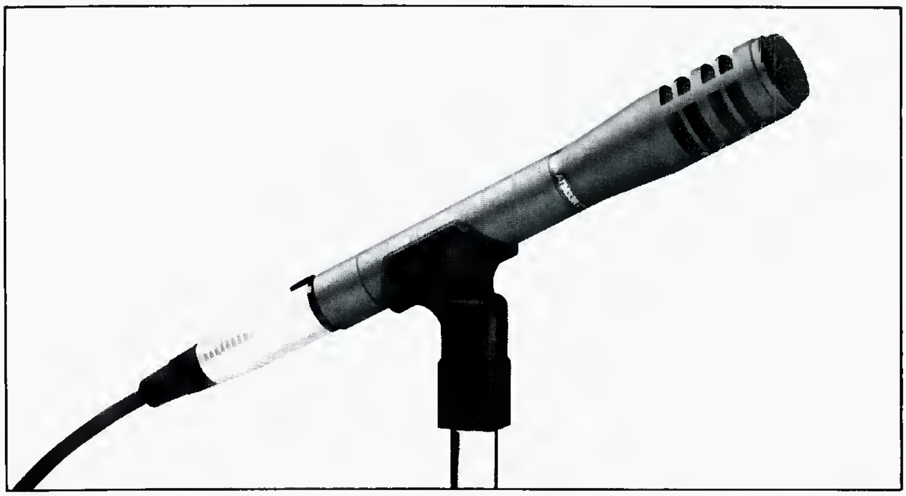
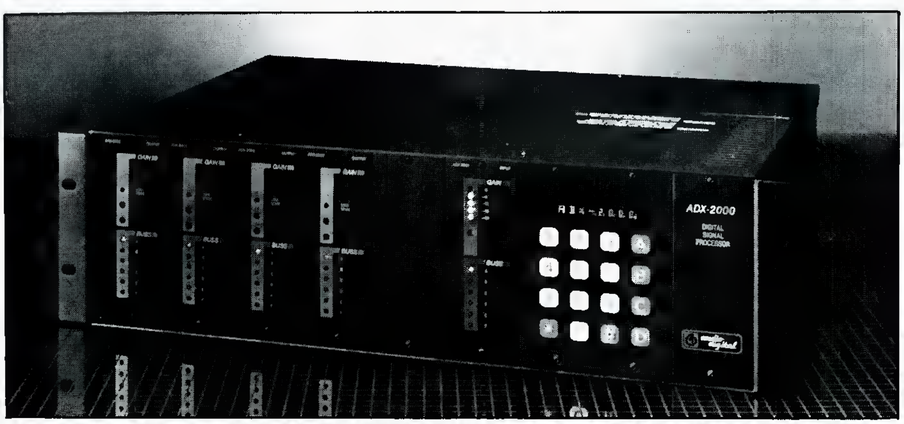


AUDIO/DIGITAL DIGITAL DELAY

• Audio/Digital has introduced the ADX-2000 Digital Delay Processor and digital delay modules which automate delay control for distributed speaker systems. Sophisticated microprocessor control and nonvolatile memory storage for delay settings allow the user to quickly reconfigure the entire delay system, which may include up to six input channels, and forty output channels. The ADX-2000 is particularly useful for installations where delay settings will be changed frequently, including convention facilities, touring sound systems, and theme park exhibits. A user can key in and store settings for twelve complete system configurations, then recall any configuration for use in seconds. In addition to delay functions, the modules perform channel on/off and output gain functions.



Delay settings range from 10 microseconds to 261 milliseconds (1048 milliseconds optional) in increments of one micro- or millisecond. Dynamic range is at least 100 dB (A-weighted, no companding); THD plus noise is better than 0.1%; and frequency response is 20 Hz-20 kHz. Additional modules which will further broaden the ADX-2000s capabilities are forthcoming.

Mfr: Audio/Digital, Inc.
Price: $3,500.00 for standard configuration.

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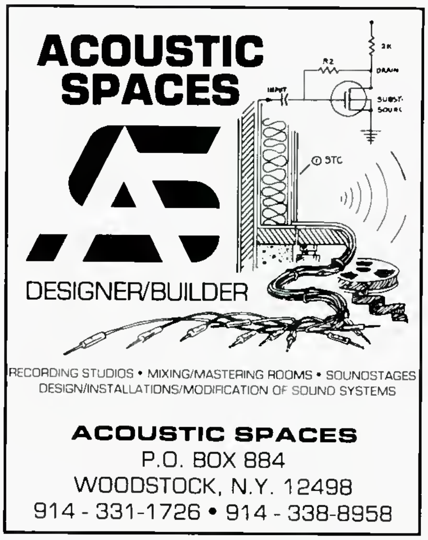
FOR SALE

HEATSHRINK available by the foot in black sizes 3/64" to 1". For more info. and prices write to **Quality Audio Products PO Box 2595 Fairlawn, NJ 07410.**



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MCI JH-24 w/Autolocator III—2 units both in excellent condition; **3M** M64 2-track; **Scully** 280 2-track. **Mitsubishi X-80** (2 units available Sept 1986). All equipment professionally maintained and in service in major studio. Broker participation welcome. **Contact Pat Scholes at: (901) 725-0855, Ardent Studios, Memphis, TN.**

RADIO TRANSCRIPTION DISCS: Any size, speed. Drama, comedy, music, variety, adventure, soaps, children's, AFRS, big band remotes, library services. **Kiner-*db*, Box 724, Redmond, WA, 98073-0724.**

WANT TO BUY

Otari MTR-90 24-track series 2 of recent vintage. **Contact Pat Scholes at: (901) 725-0855, Ardent Studios, Memphis, TN.**

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TRUTH... OR CONSEQUENCES.

If you haven't heard JBL's new generation of Studio Monitors, you haven't heard the "truth" about your sound.

TRUTH: A lot of monitors "color" their sound. They don't deliver truly flat response. Their technology is full of compromises. Their components are from a variety of sources, and not designed to precisely integrate with each other.
CONSEQUENCES: Bad mixes. Re-mixes. Having to "trash" an entire session. Or worst of all, no mixes because clients simply don't come back.

TRUTH: JBL eliminates these consequences by achieving a new "truth" in sound: JBL's remarkable new 4400 Series. The design, size, and materials have been specifically tailored to each monitor's function. For example, the 2-way 4406 6" Monitor is ideally designed for console or close-in listening. While the 2-way 8" 4408 is ideal for broadcast applications. The 3-way 10" 4410 Monitor captures maximum spatial detail at greater listening distances. And the 3-way 12" 4412 Monitor is mounted with a tight-cluster arrangement for close-in monitoring.
CONSEQUENCES: "Universal" monitors, those not specifically designed for a precise application or environment, invariably compromise technology, with inferior sound the result.

TRUTH: JBL's 4400 Series Studio Monitors achieve a new "truth" in sound with an extended high frequency response that remains effortlessly smooth through the critical 3,000 to 20,000 Hz range. And even extends beyond audibility to 27 kHz, reducing phase shift within the audible band for a more open and natural sound. The 4400 Series' incomparable high end clarity is the result of JBL's use of pure titanium for its unique ribbed-dome tweeter and diamond surround, capable of withstanding forces surpassing a phenomenal 1000 G's.
CONSEQUENCES: When pushed hard, most tweeters simply fail. Transient detail blurs, and the material itself deforms and breaks down. Other materials can't take the stress, and crack under pressure.

TRUTH: The Frequency Dividing Network in each 4400 Series monitor allows optimum transitions between drivers in both amplitude and phase. The precisely calibrated reference controls let you adjust for personal preferences, room variations, and specific equalization.
CONSEQUENCES: When the interaction between drivers is not carefully orchestrated, the results can be edgy, indistinctive, or simply "false" sound.

TRUTH: All 4400 Studio Monitors feature JBL's exclusive Symmetrical Field Geometry magnetic structure, which dramatically reduces second harmonic distortion, and is key in producing the 4400's deep, powerful, clean bass.
CONSEQUENCES: Conventional magnetic structures utilize non-symmetrical magnetic fields, which add significantly to distortion due to a nonlinear pull on the voice coil.

TRUTH: 4400 Series monitors also feature special low diffraction grill frame designs, which reduce time delay distortion. Extra-large voice coils and ultra-rigid cast frames result in both mechanical and thermal stability under heavy professional use.
CONSEQUENCES: For reasons of economics, monitors will often use stamped rather than cast frames, resulting in both mechanical distortion and power compression.

TRUTH: The JBL 4400 Studio Monitor Series captures the full dynamic range, extended high frequency, and precise character of your sound as no other monitors in the business. Experience the 4400 Series Studio Monitors at your JBL dealer's today.
CONSEQUENCES: You'll never know the "truth" until you do.

JBL

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8500 Balboa Boulevard
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FIELD PRODUCTION

A special job demanding specialized products.



For your audio needs: a growing line of compact, easy-to-use FP amps and mixers.

Shure FP products are built specifically for ENG, EFP, film, and video work. They're not general audio products that "might" work on remotes. And no one offers as wide a selection with this kind of built-in ruggedness and reliability.

FP32

FP42

For Stereo Remotes. The FP32 Stereo Mixer is comparable in size and features to our famous FP31. Its stereo capability, light weight, easy-to-use controls and convenient shoulder harness make it the first choice of field crews. Our FP42 Stereo Mixer simplifies mic cueing, so important in situations like sports remotes. Plus it enables you to easily mix down stereo in your post production booth. It offers all the features of the popular M267 plus stereo capability and a stereo headphone amp.

FP31

FP16

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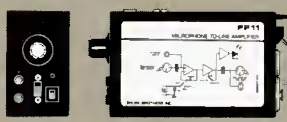

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