March 1987

Recording ENGINEER/PRODUCER

T.M.

The Technical Journal for Audio Professionals

Circuit Principles
Page 60

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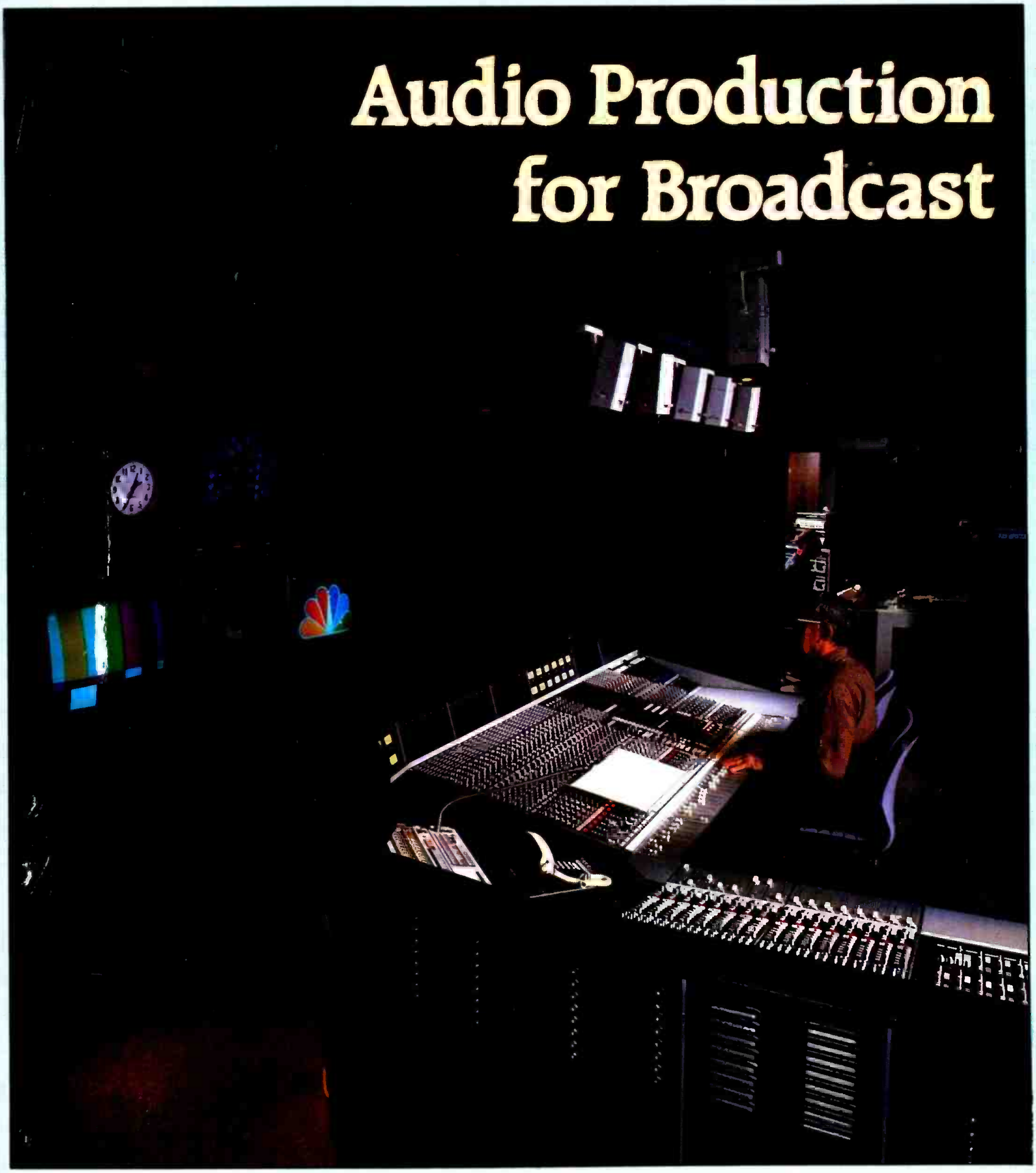

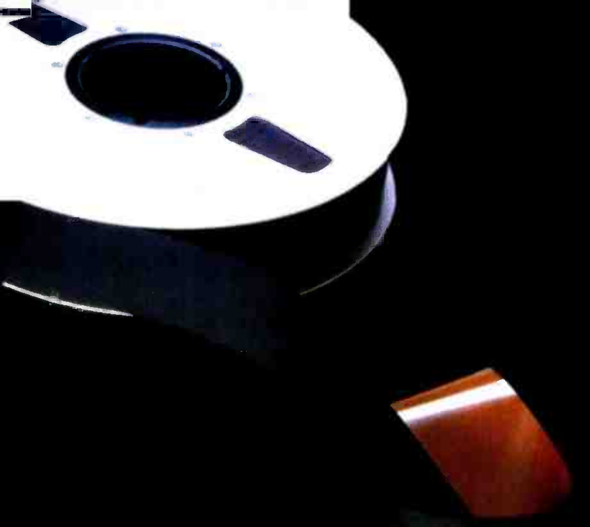

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Finally the true potential of the digital studio can be realized. No longer are you limited to storing and retrieving digital data on media designed for outdated tape technology. The Direct-to-Disk Multi-Track Recording System by New England Digital uses multiple, high capacity, winchester hard disk drives for data storage.

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Start with the industry-proven Synclavier Digital Audio System. Now available with a 32 track digital sequence recorder, 32 megabytes of high-speed RAM and on-line storage support for up to 2000 megabytes of sampled sounds.

The Direct-to-Disk System can be added at any time. Operation is simple! The system is controlled by the Synclavier's keyboard control panel. The easy-to-use interface provides all standard tape recorder functions, and more!

The finest quality 16-bit A/D conversion processes and output filtering technology available are combined with variable "stereo" sampling rates of up to 100kHz to offer audio fidelity unequalled by any other system.

The Direct-to-Disk System stores large volumes of digitally coded information on formatted winchester hard disks. Once stored, this information can be accessed randomly at any point in the recorded program material. This random access technology provides virtually instant rewind and sophisticated editing features that would be impossible using conventional technology.

When comparing the Direct-to-Disk System with standard tape-based digital recording there is a dramatic difference. For example, the Direct-to-Disk System does not need error correction. Its negligible error rate contrasts sharply with tape-based digital recorders which require error correction software to compensate for error rates of up to 180,000 bits per hour. This dramatic difference in data integrity illustrates New England Digital's commitment to quality and audio fidelity.

Expanding the system is simple. Start with as few as 4 tracks for overdubbing vocals or live instruments onto your Synclavier sequences; add on more tracks and recording time as needed. With configurations of up to 16 tracks and almost half an hour of recording time large multi-track projects can be easily completed. With the Synclavier's advanced hardware and software architecture, you always have the option to expand.

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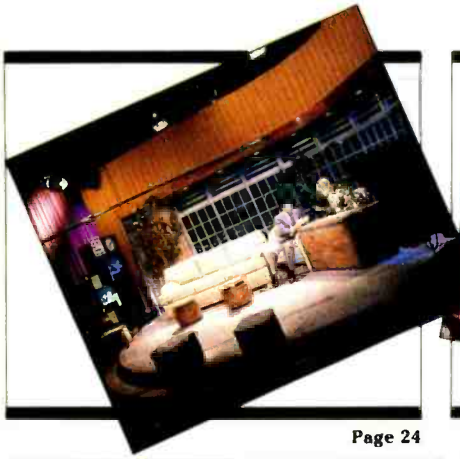

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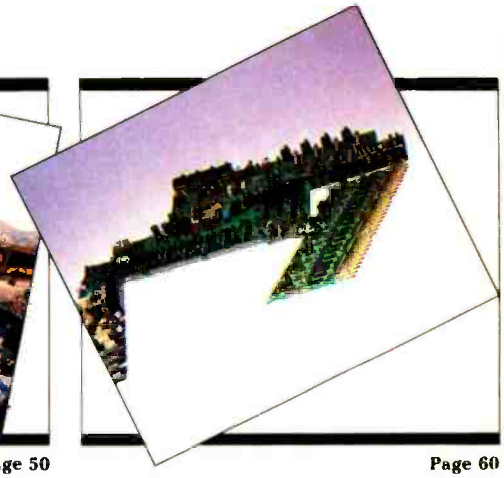

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On the Cover:

NBC-TV Studio 4, Burbank, features an LEDE control room design by Chips Davis and an L-shaped SSL SL6048 console. Photo by Jackie Sallow and courtesy of Solid State Logic.

RECORDING ENGINEER/PRODUCER—Volume 18, No. 3—(ISSN 0034-1673) is published monthly by Intertec Publishing Corporation, 9221 Quivira Road, P.O. Box 12901, Overland Park, KS 66212-9981. Second-class postage paid at Shawnee Mission, KS and additional mailing offices. POSTMASTER: Send address changes to Intertec Publishing Corporation, P.O. Box 12901, Overland Park, KS 66212-9981.

Post production is a race against time.

If you work with equipment that slows you down, the competition will pass you by.

So you'll be glad to know that Tascam's engineers have created the very first racing machine for audio post production.

Introducing the 24-track Tascam ATR-80 Recorder/Reproducer.

Our new machine is built for speed. The kind of speed that makes posting a lot more productive.

To start off, Tascam's exclusive samarium cobalt motors enable the ATR-80 to shuttle up to 380 IPS. So you'll never again have to wait for the audiotape to catch up with the videotape.

You'll save editing time, too, because unlike other audio recorders, the ATR-80 allows edit previews.

Which means no more retakes.

And no more punch-in problems. In an engineering first, we devised a set of microprocessors that makes editing gapless *and* seamless. Your edits will be perfectly transparent, as well as surprisingly quick.

Overdrive for Overdubs.

You can accelerate your overdubbing, thanks to the lead we've taken in head design.

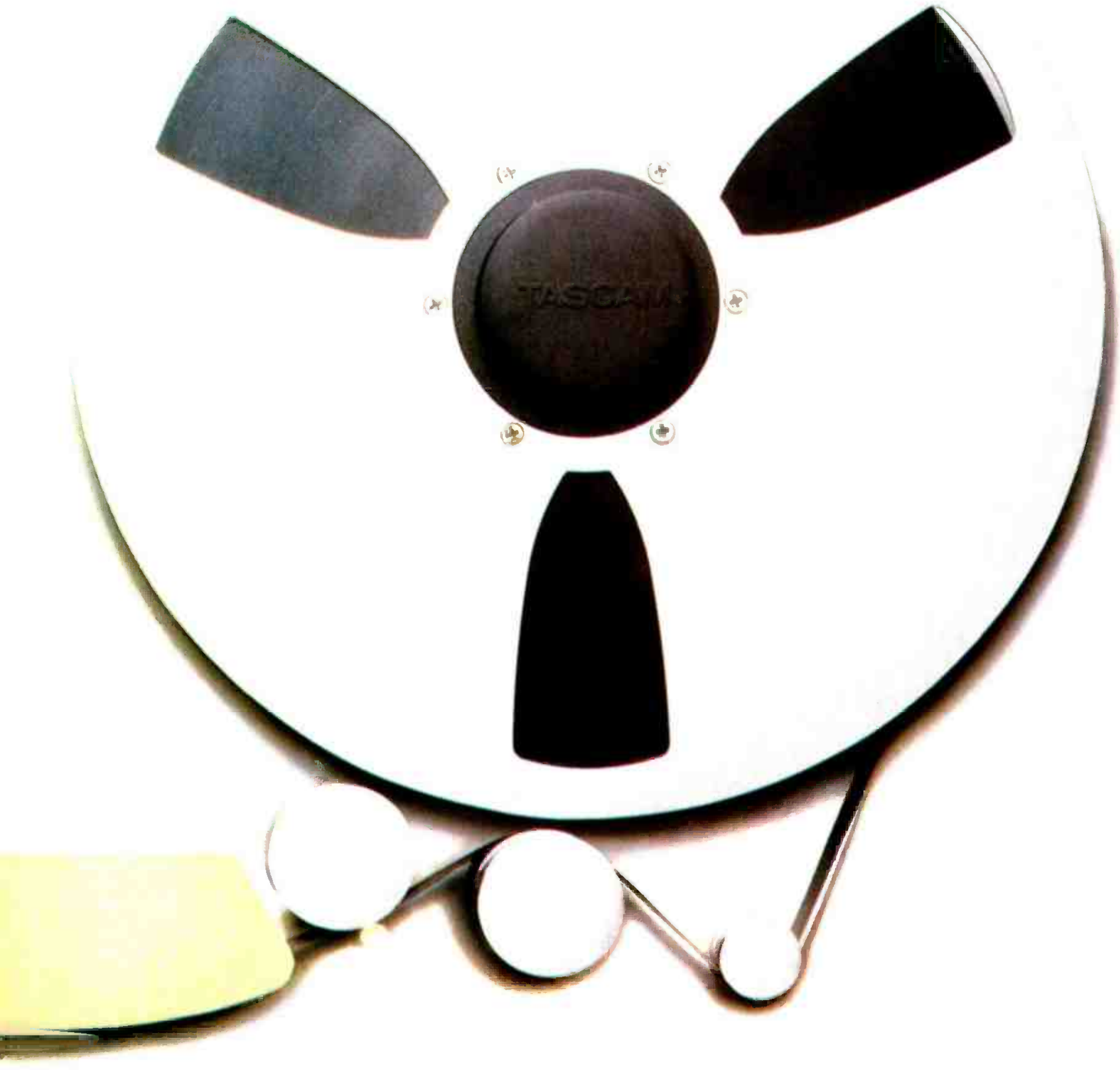
The ATR-80's uniformly responsive heads allow you to make final EQing decisions right in the Sync Mode, without having to rewind and check the sound from the Repro head.

Our heads also give you crosstalk rejection that beats most others. So you can use the track next to time code for music, instead of wasting it as a guard band.

A 32 track version of the ATR-80 will be available soon.   Dolby SR is a trademark of Dolby Laboratories Licensing Corp

And now there's a faster way to finish.

TASCAM



Quicker editing and overdubbing would ordinarily be enough to distinguish a new machine. But the ATR-80 doesn't stop there.

For instance, it interfaces instantly with any existing noise reduction system (including Dolby SR). And it accepts 14″ reels, so you can work non-stop with an hour's worth of tape.

Faster in the Long Run.

We even built speed into the ATR-80's maintenance systems: you can replace its modular power supply in just minutes if anything goes wrong.

Of course, that's very unlikely. Because like every Tascam component, the ATR-80's power supply has been built to the industry's most rigorous standards. And designed with the benefits of 30 years' experience and field-testing.

We're so confident of our quality, in fact, that we'll come to your studio and give the ATR-80 a complete check-up. While it's still under warranty.

So call us for the Tascam ATR-80 dealer nearest you. Then look one over and start it up.

You'll finish in the money.

TASCAM

Circle (5) on Rapid Facts Card

Editorial

Audio Specialties

The shifting fortunes of the recording and production industry have resulted in some fundamental changes now beginning to permeate the studio community. As we all know, most record labels face some stiff competition for the consumer dollar, and are displaying a moderate degree of caution in funding projects with new or less established artists. The bottom line is that we probably cannot expect to see a major expansion in conventional session activity within the short term.

The good news is that many other alternative avenues have been steadily opening up for studio operators, including audio production for broadcast, video sweetening, corporate audio, multimedia presentation and film scoring. Diversification has become a way of life for a growing number of facilities in the major production centers around the country. And studio owners that have been able to adapt to such changing operational requirements, and have managed to attract engineering and production staff versed in the ways of specialized audio, seem set to flourish.

Studio managers now cultivate a circle of contacts at not only record labels but also advertising agencies, corporate clients, film and television production companies that need the type of audio sessions in which their respective facility has demonstrated an expertise.

The downside, however, is that the era of diversification for studio operators (which can be summed up in terms of: "Not putting all your session eggs in one record-industry basket") might not be such good news for facility staff.

Despite a studio owner's intention of widening the client base, very seldom does this mean that the staff have to handle just about any type of session. With the exception of larger facilities running more than two or three rooms, the majority of studios are looking to focus their attentions on only one or two market segments.

These days "diversification" usually translates to: "Let's move away from one competitive and/or not so lucrative area into another." And if the studio can corner the market in providing such expertise (or at least become reasonably successful in the process) so much the better for all concerned.

Once the facility has reoriented its client base toward a new area of interest, staff at these studios have to familiarize themselves with the specific production skills needed to handle such sessions. I wonder, however, at the long-term implications for engineering and production engineers in general.

It used to be a fact that the majority of engineers could pretty much turn their hand to just about any type of audio session that came their way. In the days before MIDI, multimachine synchronization and audio/video interlock, consoles and control-room hardware were a great deal less complex. While I'm surely not advocating a return to those "small is beautiful" days of only a decade ago—Eighties technology has, without a doubt, provided us with a number of innovative and creative production tools—specialization may reduce an engineer's options.

Consider, for example, the new breed of engineer that specializes in electronic-music production. This might possibly be a personal career orientation; having maybe played in a band, he or she may decide to concentrate on the recording rather than performance side of music. Unquestionably, such a combination of musical and technical background can prove extremely useful on today's high-tech sessions.

Alternatively, such an engineer might find that the studio he or she is currently employed is making the conscious (and timely) move toward EM sessions, and discovers that such a reorientation coincides with his or her own particular career direction. The same applies to staff engineers at studios courting jingle and commercials clients, or those looking to attract video sweetening sessions.

But, if we can learn anything from the immediate past, it would be that nothing is forever in the recording and production industry. Today's specialization could very well turn out to be tomorrow's obsolescence. While I would be the first to concede that the production categories listed here look set to remain viable for the foreseeable future, there are many others that may easily pass into obscurity.

Aside from conventional disc mastering, soon to be superceded by CD and the coming generation of R-DAT and optical storage technologies, all-digital consoles and recorders could mean changing futures for analog maintenance staff and second engineers.

My advice would be to keep yourself up to date with advances in audio and video technology, and possibly try to sit in on a few session outside your immediate experience. Keeping your options open—not to mention experiencing first hand the demands of different types of production techniques—will allow all of us to remain competitive in todays rapidly changing industry. R·E/P

Mel Lambert

Mel Lambert
Editor

News

Recording engineering classes offered at UCLA

The UCLA Extension Department of the Arts will offer three recording engineering courses during the spring quarter, starting with "Digital Audio Recording: Principles and Practices," meeting on Wednesday, April 1 thru May 27. Independent engineer Van Webster will lead discussions and lectures on digital recording principles, techniques and equipment, emphasizing aesthetic considerations in music production. Fee for the class is $215.

"Recording Engineering Practice II," meeting at a 24-track digital studio on 12 Wednesdays, April 1 thru June 17, will cover recording engineering techniques and studio procedure, studio organization, live recording, overdubbing and mixing. Organized by engineer Geofrey Sykes, the class fee is $890.

Multitrack Editing for Audio Recording," a weekend class on April 3-5, will cover editing of pre-recorded material on 8- thru 24-track. The fee will be $150.

For further details, call 213-825-9064.

Sanyo to set up CD manufacturing plant

The new company, Sanyo Laser Products Corporation, will be based in Richmond, IN, with a projected start date of June 1, 1987 for Compact Disc and optical-disc manufacture. Annual capacity is expected to exceed 5 million discs during 1987, reaching more than 15 million by 1989.

Having begun pressing CDs in June 1983, with CD-ROM manufacture coming on-line by March 1986, at Gifu, Japan, Sanyo now presses more than 15 million discs per year for more than 15 U.S. record companies, and more than 30 European record companies, in addition to its Japanese customers.

New engineering group at Otari

According to company president, Jack Soma, the engineering support group will have several tasks: to study the applications of Otari machines and orient them to fit the U.S. market and the particular customer needs; to study and implement any required machine modifications; design accessories; make software changes and updates; and to design quality control and service guidelines for new machines.

The new group is described as a natural evolution of Otari's R&D activities that were established four years ago.

"We fully expect them to make an important contribution in many areas of Otari's engineering and marketing efforts," Soma says.

Seminar in audio recording

The University of Iowa's eighth annual seminar in audio recording will be June 15-26, with Professors Stanley P. Lipshitz and Lowell Cross as principal instructors. Professor Lipshitz, one of the most highly respected and widely published researchers in the audio field, has appointments in both applied mathematics and physics at the University of Waterloo, Canada. Lowell Cross, professor of music at Iowa, is well known to RE/P readers for his reviews and articles on studio microphones.

Topics to be covered, all of which will

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RECORDING ENGINEER/PRODUCER is edited to relate recording science to recording art to recording equipment, as these subjects, and their relationship to one another, may be of value and interest to those working in the field of commercially marketable recordings and live audio presentation. The editorial content includes: descriptions of sound recording techniques, uses of sound recording equipment, audio environment design, audio equipment maintenance, new products.

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INTERTEC PUBLISHING CORPORATION

News

involve demonstrations of equipment, include stereophonic and Ambisonic microphone techniques; microphone evaluations; noise reduction systems; digital processing; and digital/analog and digital/digital comparisons.

For more information contact Professor Lowell Cross, Recording Studios, School of Music, The University of Iowa, Iowa City, IA 52242-1793; 319-335-1664.

Stramp USA formed to distribute European products

The new company has been established as the U.S. distributor for products designed and manufactured by the West German firm, Audio Vertrieb Peter Strueven GmbH.

The Stramp CP-1 mixdown automation system is the first product to be introduced by Stramp USA. The computer-based MIDI automation systems, which consists of the Realcon software package and Stramp CP-1A hardware units, will automate up to 64 console channels with no modification to existing equipment, the company says.

Managed by Cliff K. Petroll, Stramp USA will initially sell directly to the end-user while a dealer network is being set up. The company's address is: 7 Charles Lane, New York, NY 10014; 212-929-9479.

Beta-test sites purchase CMX CASS-1 editing systems

Four production facilities selected as beta-test sites for the CASS-1 computer-aided sound sweetener have purchased their systems, according to Larry Weiland, CMX vice president and director of marketing. The facilities are: Century III, Boston; Studio Tempo, Montreal; One Pass Productions, San Francisco; and Streeterville, Chicago.

"There is always some risk in introducing a new technique to the field of post-production," Weiland says. "Beta-site acceptance validates our belief in the CASS-1 concept."

The system is designed for post-production facilities that offer audio sweetening and audio-for-video services. The system automates up to 32 faders of a VCA-equipped console, as well as six tape machines.

Sonopress opens CD plant in West Germany

Projected annual production capacity at the new plant is 25 to 30 million CDs, making the operation the second largest in the world, according to Uwe Swientek, general manager. The plant initially will employ 200 people in CD manufacture, with employment rising to 300 by 1990.

The new plant will produce CDs for both Bertelsmann-owned music companies and outside firms.

Currently, there are about 15 Compact Disc plants in the United States, Europe and Japan, with a combined annual capacity of approximately 200 million CDs. Europe leads with 85 million, with U.S. capacity at about 60 million and Japan's at 55 million. It is estimated that total annual capacity will rise to about 460 million CDs in 1987 and 500 million in 1988.

Syn-And-Con designer workshop

A recording studio designer's workshop will be June 12-14 at Master Sound Astoria, New York. Sponsored by Synergetic Audio Concepts and hosted by Don and Carolyn Davis, the workshop is aimed at "sharing design, construction, and proof of performance technology of recording control room and studio design."

The staff is comprised of designer Charles Bilello and Dr. Peter D'Antonio of RPG Diffusors.

The workshop will stress the importance of eliminating early reflected energy, quadratic residue diffusion and the non-statistical utilization of absorptive materials to control specular reflections in small room acoustics. The format will place a heavy emphasis on measurements, workshop organizers say. TEF measurements of acoustics, monitor loudspeakers and diffusion will be used by the instructors to illustrate lecture material.

APHEX DOMINATOR HANDS-ON —A Correction—

As part of the Hands-On review of the Aphex Studio Dominator, published in the January 1987 issue, we solicited additional user comments. All of the people we contacted via telephone were invited to review their comments prior to publication, to check for technical accuracy.

One of the five users we interviewed, Andy Morris, chief engineer of Buzzy's Recording Services, Hollywood, requested that his comments not be included. Because of production deadlines, however, we were unable to omit Andy's comments from the review.

Our apologies to both Andy Morris and **RE/P** readers for any confusion that may have resulted from this unfortunate oversight—*Mel Lambert, Editor.*

News Notes

Cetec Gauss has named **AKG Acoustics** as exclusive representative for Canada. AKG will act as a stocking distributor for all Gauss loudspeakers, high-frequency drivers, horns, systems and recone kits. AKG Acoustics' main office is located at 601 Milner Ave., Scarborough, Ontario M1B 1MB; 416-292-5161.

A/V Technology International has been appointed exclusive distributor in India, Central America and South America for **Aphex Systems.** The company has also acquired exclusive distribution rights in the Far East for **Eventide,** and will be handling international sales and distribution on an exclusive basis for **Pacific Recorders & Engineering Corporation.**

Electro Sound has been named as exclusive sales agent for **TTL USA,** and will represent the company's automatic cassette loader, tape twist detector and cassette stacker stamper through its existing sales and distribution network.

DOD Electronics Corporation has appointed **Music Associates,** Salt Lake City, UT, as manufacturer's representative for the states of Colorado, New Mexico and Wyoming.

Cerwin-Vega has relocated to 555 E. Easy St., Simi Valley, CA 93065; 805-584-9332.

People

The Audio-Video Systems Division of **Ampex Corporation** has appointed **R. Bland McCartha** as director of marketing. In his new position, McCartha will oversee all aspects of product and applications marketing, as well as technology planning. Prior to this appointment, he was division manager of applications marketing.

Martin Gallay has been named executive marketing counsel at **Westlake Audio**. He will have primary responsibility for expanding the marketing and distribution of products from the company's manufacturing group, particularly monitors and loudspeakers. Additionally, he will coordinate the company's external and internal marketing communications, media relationships, convention and trade-show participations, plus vendor and customer relations. Founder and **RE/P** publisher until its sale in January 1986 to Intertec Publishing, Gallay served previously as its publisher emeritus. R-E/P

In The Past We Had A Big Advantage Over The Competition.

Now We've Got A Small One.

Until UREI's 813 Time Align® Monitor entered the studio, speaker systems had become a "smear" on the industry. A "time smear," in which high and low frequencies subtly assaulted the ear because they arrived out of sync. The results were general listener fatigue and unrealistic sound, particularly on lead instruments and vocals.

The UREI 813 solved the "time smear" problem with Time Alignment™, unifying sound into a single point source. This dramatic breakthrough, along with other major technical advances, soon established the 813 as the industry standard.

Now UREI introduces less of a good thing: the 809 Time Align® Studio Monitor. The 809 delivers all the engineering depth of its big brother, but at a compact size and price that's ideal for small control rooms and near-field applications.

UREI's 809 features a remarkable, all-new 300mm (12") coaxial driver that achieves a true one-point sound source, superior stereo imaging, and tight bass. It incorporates a unique titanium diaphragm compression driver that unleashes unequalled high frequency response.

The 809 has exceptional power handling capabilities, high sound sensitivity, and low distortion. It accomplishes precise acoustic impedance matching and smooth out-of-band response with UREI's patented high-frequency horn with diffraction buffer. And its ferrite magnet structures assure the system's high sensitivity drivers will not degrade with time and use.

UREI's Model 809 Time Align® Studio Monitor. Smaller package. Smaller price. Same impeccable "813" sound quality. See how it fits into your studio today.

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Circle (8) on Rapid Facts Card

Letters

Small is beautiful?

From: Thad A. Edwards, Matthews, NC

With reference to David Scheirman's article on Jimmy Buffett's "Floridays" Tour, published in the October 1986 issue of **RE/P**, I'm glad to see that you're finally writing articles about some of the better organized, small sound companies. I am not using the expression "smaller" sound company in a negative sense, but in the terms that Sound Image, the company supplying hardware for the tour, has found and conquered its market, and still continue to better themselves.

I have watched this company for many years, since the "Silverfish" days, and I am very impressed with what it has been able to do in a short period.

One of the company's biggest assets is not the equipment but the personnel that have been with the company and the Buffett organization for many years. Sound Image has been as much of a reason for Mr. Buffett's success as his many wonderful albums. I enjoy going to a Buffett concert to see what new developments the guys have done.

I do not say this because I work in the contemporary Christian music industry, but because Buffett concerts sometimes save my soul after a long tour of duty, as we call it.

I want to close by saying keep up the good work, and maybe you could consider doing an article on sound systems in the CCM industry.

P.S. Next time try and get Sound Image owner Ross Ritto to smile! Also let Mr. Buffett know he has one vote in the Bible Belt for president.

Connecting Equipment

From: Chris L. Wood, Sound and Fury, Ashland, OR.

Thank you for publishing Allen Burdick's article, "Interconnecting Audio Equipment," in the October issue. Allen's concise treatment of such a complex subject is impressive. We hear much about how to make sounds, but little about how to move them.

Now that *dB* magazine has abandoned its technical format, I especially hope **RE/P** will publish more on the art of installation and maintenance. RE/P

Managing MIDI

By Paul D. Lehrman

One of the most confusing aspects of MIDI is Mono Mode. Part of the reason for the confusion stems from the fact that what it does has almost nothing to do with what it's called. This can be traced back to the beginning of MIDI, when one manufacturer misread the original spec and designed an instrument around the misunderstanding but which then proceeded to take over the world. (But that's another article in itself.)

Mono Mode (or Mode 4:Omni Off/Mono, to be official about it) traditionally does two things. First, it sets up a synthesizer so that it can respond to only one note at a time—"monophonically," although the correct musicological term is really "homophonically." That's easy enough to understand.

At the same time with some synthesizers, however, it allows the synthesizer to play several notes, each with its own sound or patch. This apparent contradiction resolves itself when you understand that this second, "polytimbral" capability is only possible if the various musical lines are sent to the synthesizer *on different MIDI channels.*

So, in Mono Mode a polyphonic synthesizer is supposed to transform itself into a number of single-note synthesizers. The number is usually equal to the synth's number of voices. (For example, a 4-voice polyphonic synthesizer can handle four MIDI channels in Mono Mode.)

Unlike synths with "split keyboard" functions, in which each patch sounds only within a particular non-overlapping keyboard range, true polytimbral synths can sound all of their voices over their entire range. Obviously, one keyboard cannot play more than one type of sound (that's why organs—remember them?—have two or more manuals), so such devices are designed to be controlled by something else, like a sequencer.

Early polytimbral synthesizers were pretty simple. They assigned one voice, with its patch, to MIDI channel n; the second voice, with a different patch, to channel $n+1$; the third to $n+2$; and so on until they ran out of voices. That was about it—if you wanted to send non-note data, like pitchbend, controllers or program changes, chances were it would be ignored, or only the "basic" (lowest) channel would respond.

For a while, it seemed that nobody quite knew what to do with polytimbral synths. Many of their designers included those capabilities for use with the unit's *internal* sequencers, but didn't pay much attention to how they might be used with MIDI. Often the directions on how to set up an instrument for multichannel use with MIDI were buried in the back of the user's manual, or absent altogether.

Making things worse was that Mono Mode did (and does) different things in different manufacturer's products. Some instruments used it only for its polytimbral effect. Others didn't have polytimbral capabilities, but would go into a homophonic state, at the same time adding a legato feature. With this latter feature, if a new note is pressed before the previous note is released, the pitch will change without initiating a new attack. Sometimes a glide can be automatically inserted between the notes, for a more vocal-sounding effect. And some manufacturers included both polytimbral and legato capabilities, inseparable from each other.

Mode 4: Omni Off/Mono allows synthesizers to play several notes with its own sound or patch.

Now, Mono Mode is being recognized for the huge benefits it can give an instrument. The confusion among manufacturers has not ceased but, in some ways, has actually served to help the cause of Mono Mode because new uses are being thought up all the time.

The restriction of one channel/one note is becoming a thing of the past: keyboards and keyboardless, rack-mountable boxes are showing up that can split their four, eight, or 12 voices among several non-contiguous MIDI channels, with different numbers of voices being assignable to each. One channel might be set up to respond homophonically with legato, another might be homophonic but with retriggered attacks, while a third might be fully polyphonic.

Some of the units will even decide automatically how many voices to make available for a particular sound at a particular time by a process of intelligent "channel-stealing." Patch, controller, and even system-exclusive information can now be sent to each of the channels independently.

For MIDI guitar players, even if they are not interested in polytimbral capability, this last capability is of primary importance. With the current crop of guitar-to-MIDI convertors that send information from each string over a separate channel, players can now bend a single string, or apply a little left-hand vibrato, without hearing the whole chord fly off.

Many sampling synthesizers use Mono Mode in combination with split keyboards to provide real-time access to a huge number of samples; some sampler's memories can be divided up into 128 or more parts. Although gaining access to them all on one channel would be difficult (you'd need at least a 10½-octave keyboard), if several channels are used simultaneously, there's no problem. In addition, some such units with multiple audio outputs use Mono Mode to assign different samples to the different outputs, so they can be processed and mixed separately.

Even MIDI-controlled effects are getting into the act. Although there aren't any effects units on the market yet (though I suppose it won't be long) with multiple MIDI-addressable effects memories, at least one manufacturer is using Mono Mode to allow multiple "patch maps" to be set up on different channels within a single device. Which means that a patch number coming in on one MIDI channel will have a different effect than the same patch number coming in on another channel. In other words, patch 16 on channel 1 may call up a gated-plate program, but patch 16 on channel 2 will call for a panned-echo program—and if you want to get the gated-plate program from channel 2, you need to send it patch 47.

This function can be useful when you have several synthesizers feeding the same effects box, a situation that would be encountered more in live performance (where the keyboard guy normally plays only one instrument at a time), than in the studio.

Because it improves the flexibility of an instrument, at the same time driving down the price (because more functionality can be crammed into one box), Mono Mode is becoming one of the most valuable tools in the MIDI designer's and user's repertoire. It has graduated from little more than a curiosity in the MIDI specification, to an integral part of any MIDI setup.

R·E/P

Paul D. Lehrman is Boston-based free-lance writer, electronic musician, producer and a regular RE/P contributor.

Sound on the Road

By David Scheirman

For many touring sound system technicians, the loudspeaker system package that is to be trucked around and set up daily has been the focus of intense research and design efforts during the past few years. The speaker system not only dictates the "sound" of a company's PA efforts, by virtue of its design type and method of combining multiple cabinets into arrays, but it also is the primary visual reference point for both the audience and the paying client. If the speaker system sounds mediocre but looks fabulously trendy, it may often be preferred in some circles over a system that sounds great but has poor visual characteristics.

This concern over the visual appearance of large touring sound system arrays has led to a natural progression of speaker enclosure development. While many concerts 10 years ago featured speaker systems that appeared to be "piles of parts," or stage stacks and flying baskets make up of non-integrated, loose-component systems, the beginnings of today's trends toward integrated, modular speaker arrays were there to be seen.

The early attempts at modular touring speaker systems, made during the Seventies, focused primarily on identical square or rectangular enclosures. These units were sized to match available truck spaces, so that as many speaker units could be packed into a trailer as possible without wasting an inch. Cabinets such as the famous Clair Bros. S-4 became a standard. Direct-radiating speaker loading techniques enabled system designers to keep the enclosures relatively shallow.

The Tasco Harwell system attempted to integrate horn-loading techniques with modular speaker enclosures for touring. The size of the cabinets required crept up, however, and when large arrays were made up of multiples of these refrigerator-sized units, the arrays were massive and tight—combining for effective acoustical coupling required very specialized hanging and rigging skills.

By the late Seventies, several companies were beginning to experiment with trapezoidal, or "wedge cut" speaker enclosures for use in touring sound systems. The pioneering work of John Meyer helped to bring the Meyer MSL-3 enclosures into the marketplace. These compact enclosures combined horn-loading with narrow rectangular packaging techniques that enabled system designers to assemble tighter, more coherent multiple arrays. Some adventurous touring sound firms, including Ultra Sound (Grateful Dead), See Factor (Rush), and Tasco purchased and began to use these new, trapezoidal speaker enclosures.

David Scheirman is president of Concert Sound Consultants, Julian, CA, and **RE/P**'s live performance consulting editor.

Sensing a new market trend, other manufacturers began to develop their own trapezoidal speaker packages. Conservative touring sound companies cautioned that such enclosures were a poor investment—how would they pack into the trucks? Wouldn't it take more labor time to get the arrays hung? However, history has shown that there is more to a trapezoidal speaker enclosure than marketing gimmickry.

The speaker system is the primary visual reference point for the audience and paying client.

Several manufacturers have begun to build and distribute trapezoidal enclosures with a certain measure of success.

While pre-manufactured speaker enclosures of the trapezoidal variety have made inroads into the touring sound field, the ultimate acceptance of this new type of enclosure must be measured by the type of new speaker systems actually being constructed for their own use by established concert sound companies. A brief examination of all new cabinetry actually built and loaded by American concert sound companies within the last calendar year shows that systems based on both rectangular and trapezoidal enclosures are being fielded.

Newer, aggressive concert sound firms are often the first to commit to innovative technologies that represent a change in direction. There is a long list of unusual speaker system types that were assembled, used for one or two tours, and then discarded. (Remember the round white plastic tweeter arrays, 18-foot bass horn flares, all-cone "high-fidelity" systems and detachable fiberglass horn throats?)

Several new systems that were premiered during 1986 feature wedge-cut enclosures, and this seems to be a lasting development. The new Steradian system from Schubert Systems Group (so named because of the mathematical term that describes the geometric shape of a wedge-cut cabinet) toured the country with Al Jarreau. The new Phase-Loc system built by Sound Image for Jimmy Buffett's tour seems to have successfully found that delicate balance needed to satisfy truck loaders, production managers and soundmixers alike. The new trapezoidal cabinetry from Sound On Stage that has been developed for use with Huey Lewis and the News, first field-tested on that national tour in December, has stage crews, sound technicians and the band satisfied.

Perhaps the most telling example that points to trapezoidal cabinetry being more than a passing fad is the new arena system assembled in 1986 by Showco. The first major concert sound company to commit to its own design of wedge-cut cabinetry, Showco has finally fielded its new rig for Genesis and Peter Gabriel after extensive research and testing. Designed with the assistance of computer measurement techniques, the new Showco system has been demonstrating to even diehard skeptics that touring sound system enclosures don't have to be rectangular.

Why chop the sides off a box at angles? Does it *not* make the system harder to load in a truck, and are there really acoustical advantages that outweigh the inconveniences?

When a large hanging array is assembled out of square boxes, a certain amount of space is required each time the parts of the array are angled to increase horizontal coverage. By the time 40 to 60 enclosures have been flown and angled, the array itself can become quite large. This is particularly true if the boxes are deep. Rectangular cabinets are thus a component of very large arrays that have pie-shaped cavities between the rows of boxes. These cavities can make the array less phase-coherent and increase the array's overall size.

Trapezoidal cabinetry can be used to create arrays that are more visually pleasing, more compact and less prone to acoustical phasing and lobing interferences. In short, it is possible for them to both look and sound better.

It is true that some truck space is lost; at each end of a row of interlocked trapezoidal cabinets, a small amount of truck interior space is unusable. The increased performance and efficiency of the new arrays, however, seems to outweight this negative aspect in the opinion of companies now using the newer style of cabinetry. R·E/P

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Circle (12) on Rapid Facts Card

Film Sound Today

By Larry Blake

Despite the fabled simplicity and flexibility of 35mm mag film, it still brings with it—like an aging star—a useless entourage. Because working with mag is a labor-intensive, manual, redundant pursuit, it is virtually impossible to throw computers in the middle of mag-based sound editing and mixing to speed up the process.

Thus, regardless of the final form that the changes in post-production sound will take, we will have to wave goodbye, once and forever, to 35mm mag.

When will all of this start to come together? First, I don't think that random-access digital sound editing is going to be used widely in motion-picture post-production until somebody can make a system whose basic hardware costs approximately $25,000. By "hardware" I mean to include the basic multiuser computer system, with terminals but *without* storage devices. The number of storage devices will be a function of the size of the sound library at hand and whether or not all materials will be on-line.

To be more specific, I foresee that the more clever companies are going to hot-rod off-the-shelf multiuser computer systems, probably based on a latest generation chip like the 80386 or the 68020, and use them to control a room of R-DAT players and recorders. The result, I hope, will be a lean, affordable system based on the real-world needs of post-production sound today.

What about the attempts currently being made at putting computers into film sound post-production? I am leery about the future both of simple, single-user systems and the expensive (more than $200,000) hard-disk digital sound editors introduced in recent years.

Very few of the single-user systems that I've seen thus far have demonstrated an awareness of the importance of either the archival storage problem or the need for a direct path of taking edited sounds to the dubbing studio. You can lock to time code with these systems, you can slip sync, and maybe you can even accommodate picture changes. But a piddling amount of RAM, or a single hard disk accessible to only one user, is just not going to cut it in feature post-production.

Think about it: Unless all the sound editors working on a film (or at a facility) have simultaneous access to one large, central archive, many copies of the sound library will have to be made, and the editors will have to shuffle disks or tapes to audition the sounds they want. Furthermore, once an editor (working on a simple, single-user system with limited channel capacity) has placed an effect against picture, how in the world does that sound get to the final mix?

With a central archive, you have two obvious choices: the Edit Decision List is recreated and recorded on some tape medium, such as an R-DAT recorder or digital multitrack, or the sounds are downloaded to RAM or a hard disk at the dubbing stage. In both of these procedures the R-DAT players shuttle back and forth in an "auto assemble," mode not unlike the way that an off-line EDL is reconstructed during on-line video editing.

We will have to wave goodbye, once and forever, to 35mm mag for sound editing and mixing.

The difference, of course, is that here we are talking about perhaps 100 R-DAT players storing a sound library shared by many users. If the same player is required to be two places at once, someone will have to wait, although probably not for long given the rumored fast shuttle time of R-DAT transports.

Getting back to small single-user systems, I can only speculate that the sounds would have to be transferred to tape by the editor as they are cut; otherwise someone else will have to reload the chosen sounds and perform this transfer at a later time. Such systems tease us with computer technology, only to leave us high and dry back in the world of redundant, manual work that we are trying to get away from.

The expensive systems address many of the problems noted above, but do so at...great expense. Time and money seem to always go hand-in-hand, but I can't imagine that the time a company will save using one of these top-end editing systems will equal their $200,000-plus cost. How can a company justify a capital outlay that will only allow slick editing of sounds unless tied to a sister system on the re-recording stage?

Additionally, even if the system architecture supports a multiuser/central-library approach, the $200,000 I am referring to is for only *one* digital editor. Assuming that increased speed lets you cut the number of sound editors in half, you will still need at least two set-ups for a complex features film.

Now we're all of a sudden talking about megabucks for a complete facility. On a dollar-for-dollar basis, give me today's technology populated with people and Moviolas. A fully equipped sound-editing room today (filled with yesterday's technology) will cost you about $600 per month.

I hate to sound like a broken record, but some of these companies, in spite of the high cost of the systems they propose, haven't spent enough time addressing the issues of mass storage and re-recording.

There is no need to spend such sums of money to be able to edit sounds in a random-access format. Which brings me back to the potential of R-DAT as discussed in my December 1986 column: I know of no other storage medium that is as cost-effective on a per-stereo-minute basis. Furthermore, because R-DAT can be used for recording, editing and mixing, the hardware design for a from-scratch facility will be simplified.

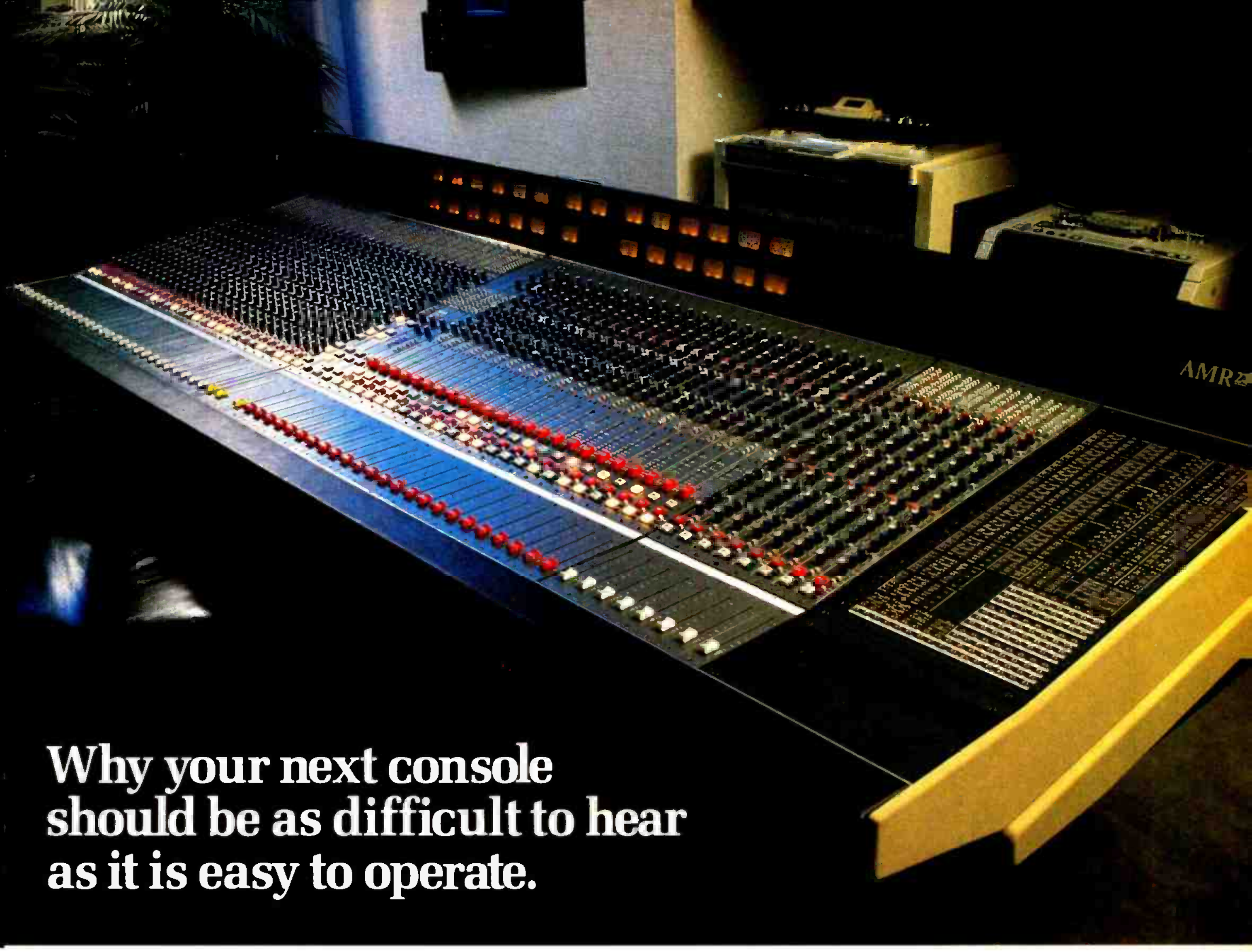
I hasten to add the obvious: I am talking strictly of the *potential* of R-DAT, based on the design parameters I have seen. Although no one I know of has worked with R-DAT systems in the interlocked editing mode that I presume will work like a charm, my big toe tells me that the format is going to fly.

If it does fall on its face, there seems to be great potential from DRAW systems that will allow users to master their own CDs with relatively simple and inexpensive recorders.

One last thing I'd almost be willing to make book on: Don't look to the major studios for leadership here. They are just too conservative and set in their ways. CinemaScope—in 1953—is the only recent major technical development sponsored by the Hollywood establishment. All the others that I can think of—Cinerama, Todd-AO and, most recently, Dolby Stereo—came about without initial big studio participation.

Along these same lines, I'm willing to bet that the next generation of sound editing and re-recording equipment and procedures will be introduced by a hungry independent facility, possibly one that doesn't exist even today. I'm not being coy here, keeping tight-lipped about some dream facility that is planned. There has been much talk from various quarters but, as far as I know, nothing has been built. R·E/P

Larry Blake is RE/P's film sound consulting editor.



Why your next console should be as difficult to hear as it is easy to operate.

The studio is more complex and less forgiving.

Electronic production techniques using MIDI and SMPTE sync require more control than a "wire with gain" can provide. But as functions and components accumulate, the console's signal path has grown more complex, and its audio performance has suffered. On analog recordings, higher levels of crosstalk, noise and intermodulation were an acceptable price for additional control. On digital multitrack, however, these flaws become glaringly obvious.

Crosstalk blurs the stereo image.

Now that digital recorders have virtually eliminated crosstalk, this is an especially annoying problem. *The AMR 24 matches the channel separation performance of digital multitracks* because it employs balanced buses that eliminate crosstalk the same way mic inputs do. This radical design approach takes full advantage of digital's more coherent stereo imaging.

Balanced buses also eliminate the intermodulation that plagues the sound of conventional "virtual ground" mix amps. *The AMR 24's noise floor is constant whether you route one input to a group, or thirty six.* So you can concentrate on the music without distractions from the mixer, even on digital multitrack.

Features shouldn't degrade audio performance.

Automation widens creative possibilities — and narrows the margin for console error. For example, FET mute switches that are "silent" individually can produce audible glitches when grouped. The AMR 24's carefully controlled switching time constants eliminate this problem.

Every circuit in the AMR 24 has been calculated with equally close attention. Each stage has at least 22 dB of headroom; total dynamic range is over 100 dB. Even so, *unused stages are bypassed to produce the shortest effective signal path in every operating mode.*

Perhaps the AMR 24 is a product of extremist engineering. But as we see it, optimum audio performance, not simply a revised layout, is what makes a console automation - and digital-ready.

The feel is familiar, the functions are unprecedented.

The AMR 24 facilitates innovative production techniques within a classically split configuration. Master Input Status switches select mic inputs or line returns on all input channels simultaneously. In its mixdown configuration, the AMR 24 will handle up to 60 tracks, because the 24 Track Select switch changes the monitor returns to line returns normalled to your second 24 track (or to synchronised "virtual tracks" from synthesisers and samplers). The monitor returns have aux buses, solo and mute, plus four bands of EQ and long throw faders, so this flexibility is achieved with no loss of audio quality. For additional effects returns, the Fader Reverse function creates an additional 24 patch points through the cue send faders.

Imaginative design and uncompromising construction give the AMR 24 flexibility and sonic transparency that represent clear achievements: especially clear on digital recordings. For all the facts on this innovative console, send your business card or letterhead to:

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Living with Technology

By Stephen St. Croix

As part of what I do for a living, I find that a great deal of detailed information concerning new and potential technologies is available to me. As I consult and design in several other fields besides electronics, information of advances in each field comes to me almost every day.

This broad-band input sometimes allows me to detect a development trend that may not be obvious at all, or one that simply isn't obvious. . .yet.

The industry intelligence to which I have access is carefully divided into three main categories:

1: Information that I use directly in my business.

2: Information that I don't use, but which is too highly classified to discuss.

3: Information that I can pass on to the readers of **RE/P**.

Sometimes I don't have to extrapolate or project the impact that new technology will have on the pro-audio business; somebody drops by and shows me the new toy they are working on for next season. All I have to do to be correct about that one is to tell the significant techno-stuff, and hope that the company actually puts it on the market. Such information falls into category No. 4.

From categories three and four, I choose what to include in this column. I must be careful when I select to cover a particular subject area. If I choose to predict the existence and/or importance of a specific technical trend or product, I cannot discuss a target of say, two years into the future, because you will have forgotten everything I said by the time the trend or product appears, which means that the column I wrote was no fun for either of us.

On the other hand, I cannot talk about anything that is truly current, because this column is subject to at least a 2-month delay between my deadline and **RE/P**'s actual publication date. Again no fun for either of us if you already have one of the devices that I'm happily dreaming about existing in the future.

As a result, I have to be as accurate as possible, while projecting about three months ahead just to talk about something as if it existed today (that's your today; my three columns ago).

• We will soon witness the emergence of what may, in fact, be the next step in the evolution of power to the people. This next generation of equipment will simultaneously put into the hands of production and recording engineers, and studios, more power and control than ever before; simultaneously taking you into the future, while at the same time bringing back one part of the past (and which I, for one miss dearly).

Stephen St. Croix, RE/P's technology developments consulting editor, is president of Lightning Studios and Marshall Electronic, Baltimore.

One well-known company active in the signal processing field has seen through the intermediate solutions to providing power and speeding forward into the past. By doing something that can only now be done, a piece of gear exists that uses brute force computer power to provide all of the myriad things to which we have become accustomed, *and* something new: *Individuality.*

The device comes equipped with a compiler so you can develop from scratch any type of DSP effects.

A new high performance digital processor and reverb unit is upon us. Yes, another one; this one, however, does some real neat stuff. The device comes equipped with a compiler designed to run on the computer of your choice, so that you can develop from scratch *any* type of DSP effect(s) you can think of; then, simply compile and upload them into the processor. [*Editor's note:* Compilation is the process of converting a high-level language that a human can understand into a high-speed machine language optimized for a particular microprocessor.]

Now, for the first time in decades, your facility can be unique in the way the best studios used to be. You can offer high quality special effects or processes that *no* other studio has; your own sound effects that are yours alone. Just like the good old days.

As this concept matures, and more control is made available to the end user, the potential to create a truly unique effect increases. The time is fast approaching when the competing studio next door may own the exact same piece of equipment, hear your effect and then spend hours turning knobs and pushing buttons trying to match it. They will not be able to duplicate your effect, however, simply because their machine is not running your DSP program. The bottom line will be individuality that anyone can hear.

I do not believe that this will be the model for every toy in the studio of tomorrow. I do believe, that equipment offering power and individuality such as this will be represented, in some way or another, in every serious audio production facility for a long time to come.

As a research tool, such a processing system is particularly interesting, because it operates fast enough to compile and send a parameter change made at the terminal to the main unit within a few clock cycles, without interrupting the audio currently being processed by the main unit.

• Speaking of novel user interfaces, another company will soon market a multiband equalizer with a minimum of front panel controls, looking suspiciously as if it will require a computer to realize its full potential. Actually, it needs only a conventional composite video monitor, which you simply plug into the video out jack on the rear. Using the unit's front panel controls, you modify and commit to memory various multipoint EQ curves. All the required data is displayed on the monitor as a 15-band graphic equalizer (one third-octave frequency centers will also be available). With more than 100 internal memories and full MIDI control, the device should prove useful in a wide variety of audio situations.

This microprocessor-controlled equalizer is an interesting design approach not only because no additional computer is required to enter or display the data, but because the device will sell for hundreds, not thousands, of dollars and yet offer high quality audio performance. Quite a difference from four years ago, when free standing programmable EQ sounded so bad that it didn't even matter that the human interface was unusable.

• If somebody doesn't come out soon with a low cost, high quality 16-bit direct-to-disk stereo recorder based on the Macintosh or Atari, I will. It should have complete time code interface, and full-on screen editing, all the way down to the waveform. The currently available Atari-based unit starts out fine, but, by the time you put a little 20Mbyte disk on it, it becomes too expensive.

There is no question that at least a meg or so of buffering and a lot of housekeeping are required to achieve direct-to-disk recording with current generation consumer PCs and hard drives. But why should the basic system be priced right where you would want it, when the disk interface costs twice as much as the rest of the system? R·E/P

SPARS On-Line

By Gary Helmers

What must a recording studio consider when developing a broadcast-related clientele for commercial voice-overs, jingles, program narration and soundtracks? With broadcasters gathering this month for the NAB Convention in Dallas, we thought that some tips would be useful for audio folks interested in entering this growth market.

There are many factors to consider when a studio decides to pursue advertising production and multi-image soundtrack business. Producers from advertising agencies and production houses require the same high quality equipment needed for album dates. Multitrack consoles and 24-track machines are the norm nowadays for ad production. You may be able to start with an 8-track facility but, as business expands, you should be ready to make the jump to 16 or 24 tracks. Digital may provide an edge in the major markets, but is not essential in most cases.

Control rooms for production work should be equipped with a pair of high quality monitors, small bookshelf speakers and some kind of limited frequency response monitor, such as the cue speakers found on many tape machines. It's desirable to check the mix balance on a variety of speakers. Producers will want to know how the spot will sound when they present it to clients on a good system, and also what it will sound like on a car radio.

A selection of outboard gear is important because you'll sometimes be dealing with tapes recorded in poor environments on antiquated equipment. Your job will be to incorporate these client-recorded gems into their production, and make it sound like a million bucks. Obviously, a parametric equalizer or two with very fine control adjustments can make life a lot easier.

Many productions call for a bewildering array of acoustic ambiences as an actor supposedly tries some product in a multitude of environments. Reverb units capable of very short decay and delay times are extremely useful for establishing the type of environments in which radio and television spots take place. Some kind of chorus and flanging device is handy when a robot or computer delivers the ad's message.

Some new tools you'll have to purchase include sound effects and production music libraries. Sound effects libraries can be bought for a 1-time charge, whereas music libraries charge on either a per-cut basis or an annual blanket fee that allows unlimited use of the library. Expect to pay $1,000 to $3,000 per year for unlimited use of a good music library. Most libraries provide a variety of payment options. Of course, you can charge your clients per cut to use these libraries.

Don't be tempted to scrimp in this area. You can draw new multi-image production business by simply having the best selection of music in town. You may want to provide a room at no extra charge for clients to preselect their music and SFX elements prior to sessions. This saves them time, which they appreciate, and it allows more sessions to be booked.

There are many factors to consider when a studio decides to pursue advertising and multi-image soundtrack business.

Many multi-image productions are of the 1-projector variety. For playback in the field, this type of show uses a standard audio cassette with mono program on tracks 1 and 2 and a 1kHz 450ms tone burst on either tracks 3 or 4, or both. The tone burst triggers the slide projector automatically as the sound track plays. The producer will expect you to have a generator capable of generating the correct tones. You can record the program on the top track of a 2-track machine, and put the 1kHz pulses in sync on the bottom track. This tape becomes the dubbing master. Consider handling the duplication of cassettes yourself, so that your hard-won client doesn't have to go elsewhere to get his dubs made.

The larger multi-image clients will request custom music tracks. This occurs when the budget allows for a multi-projector setup (six to 24 projectors all run by computer), and the client wants a great soundtrack scored to the narrator's words. Usually audio/visual houses require the audio to be recorded on tracks 1 and 2 of a ¼-inch 4-track tape, and either time code or some other clock signal on track 4. They take this tape back to their shop, plug the clock track into their computer and record the data to run the projectors onto track 3 of the 4-track master. Guess who they'll call to run the the safety of the finished show? Buy two 4-tracks or get friendly with a rental company.

Most open-reel dubbing for release to radio stations will be done at 7.5ips mono or stereo, without noise reduction. Many radio stations are still using the old Ampex operating level of 185nWb/m or the MRL standard reference level of 200nWb/m. This means be conservative with your dubbing levels. If you send a hot 320nWb/m tape to a radio station, chances are you'll hear about it in a rather unpleasant way. Video production facilities use 15ips tapes with or without time code, depending on the production. Also, make a point of getting to know the people at your local production facilities.

Another interesting feature of production is the talent you'll be recording. It's not uncommon to work with the best voices in the business (you know, guys who start talking and the VU comes up to zero perfectly; no breath noises, hisses or other annoying sounds). And on the same session the producer will bring in his daughter and three friends who are barely able to speak to provide the dialogue. You must be prepared to work with trained as well as untrained actors and actresses. They will sound quite different at first, and you'll be expected to work your magic and make everyone sound wonderful.

A couple of really fine limiters with built-in de-essing will help immensely in this sort of situation. And don't forget the noise gates (at least four) to quiet down breath noises between sentences and other noises such as page turns.

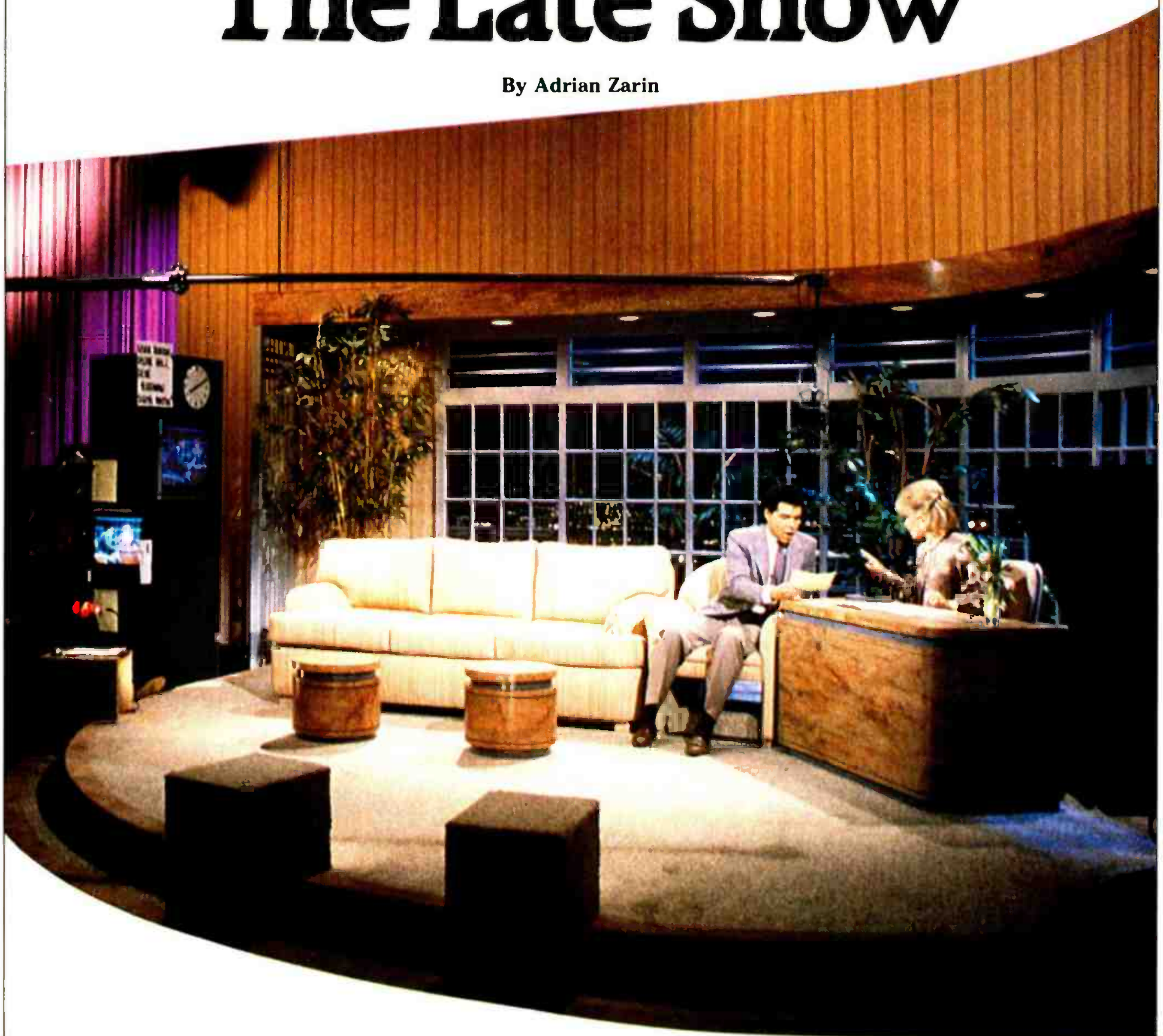
A great deal of emphasis is being placed on the ability to handle sound-to-picture or audio sweetening these days. If you are building a new control room with the intent of adding this capability later, put a lot of forethought into the design of the room. Video gear is acoustically noisy! It's advisable to put a wall between video machines and your work environment. Plan to install a good 25-inch video monitor between your main speakers and run conduit to it. Your wiring may change as business and technology evolve. Definitely plan on stereo audio, and you will also need synchronizers and controllers.

We'll continue this discussion in next month's column, and provide some tips on studio acoustics, scheduling, finances, talent and studio promotion as it relates to the world of broadcast production. In closing, I'd like to thank Joe Sheets, chief engineer at Alpha Audio, Richmond, VA, for his invaluable help in preparing this article. R·E/P

Gary Helmers is executive director of the Society of Professional Audio Recording Studios

Stereo Music Production for "The Late Show"

By Adrian Zarin



As audio production for stereo broadcast becomes increasingly important for networks and independent television stations, we take a look behind the scenes at the micing and production techniques used during the live broadcast of one of the few shows to place an emphasis on stereo music.

Can television audio benefit from the recording industry's capacious accumulation of technical knowledge? By now, most professionals involved in either or both fields would answer that question in the affirmative. And, in recruiting key audio personnel directly from the record business, Fox Television's *The Late Show Starring Joan Rivers* has put this widely held theory into nightly practice.

Mark Hudson, the show's music director, brings considerable television *and* record production expertise to the show, culled from his years as a member of recording group/comedy team the Hudson Brothers and as a producer, songwriter and musician in his own right. Thom Wilson, the show's sound designer and music mixer, is no stranger to the recording studio either, having engineered and produced numerous album projects.

Both Hudson and Wilson have an ongoing professional association with producer Phil Ramone, who also contributed some ideas to sound design for *The Late Show*. But the goal of everyone involved with audio for *The Late Show* is perhaps best summed up by Mark Hudson.

"Right from the start, we all said, let's make this sound like a recording session—every night."

The audio sources involved in the production of *The Late Show* can be broken down into three discrete areas. The first of these is music from the house band (The Party Boys and the Tramp), and includes opening and closing themes, segues into and out of commercial breaks, play-ons and play-offs for guests and accompaniment for guest vocalists. Self-contained guest bands comprise another music source; and production audio makes up the third area.

The latter category includes a variety of vocal and dialogue microphones: two boom mics, a desk mic, two hand-held floor mics, two wireless mics and dialogue mics in the band area for Mark Hudson and the Party Boys' resident saxophonist/comedian, "the Tramp" (Beverly Dahlke). Additional production audio sources include audience mics, audio outputs from a VTR used to replay film clips, etc., and the outputs from two Studer telephone interface units used for phone interview segments.

This multitude of source signals (more than 330) is organized into several basic mixes: the broadcast mix, which is sent out to affiliates across the country (and which is itself comprised of house band, guest band and production audio submixes), monitor mixes for the house and guest bands, and a house PA mix for the studio audience.

The main audio control booth at Fox Television Studio 4, used for the live stereo broadcast of The Late Show starring Joan Rivers, *features two consoles on L-shaped Neve model 4875 board (left and center) at which assistant engineer Ron Worrel handles the master stereo broadcast mix and the submixing of dialog sources. At the Yamaha PM3000 console (right), production engineer Thom Wilson handles mic and line-level sources from the house and guest bands. Inputs to the larger submixer section include desk and audience mics, boom mic, radio mics and audio outputs from various VTRs. The Soundcraft series 200D above the side console accommodates additional inputs.*

The house band: micing and acoustical considerations

Coming from his background in record production and engineering, Thom Wilson admits to having mixed feelings about the high ceiling and ambient acoustics in Studio 4 on the Fox lot, where *The Late Show* originates. He was particularly apprehensive about the highly reflective house bandstand, located stage left on the show's set.

"When we first went in there, it seemed like the inside of a huge airplane hangar," he recalls. "This is the sort of ambience I've always liked on records I've done in the past. But what we have on *The Late Show* is a much different sound approach: more contained and overtly 'sophisticated.' "

Working in consultation with Phil Ramone, Wilson developed several ideas for controlling sound reflections and microphone leakage within the bandstand area. One of these was to fill the area beneath the bandstand with absorptive acoustic foam, to reduce unwanted resonances. Wilson also decided to partially isolate the drum kit. On either side of the kit—which sits in the interior angle of the L-shaped bandstand—he positioned a 2-inch plexiglass gobo.

"I was a bit worried about being allowed to do things like this," Wilson says. "Television audio people I'd spoken to complained that they're not even in the back seat, in terms of the consideration they received; they're in the rumble seat!

"But, from the beginning, this show has taken a *different* approach. They've given me a lot of latitude. At first, the plexiglass around the drums met with a little resistance from people who said, 'We've never done that before on a television show.' I told them, 'That's *exactly* why I want to do it!'

"In the end, they went for it; and I admire them for being open to try it."

In devising a microphone setup for the house band, Wilson had the challenge of creating a stereo image on instruments such as drums and piano, while working in a tight space and combating leakage from other instruments and on-stage monitor cabinets. On drums, much of the stereo image comes from a pair of Crown GLM-200 cardioid lavalier mics used as overheads.

"It's a micing technique I learned from Bill Schnee many years ago," Wilson says, "and which he learned from Glynn Johns.

"What you do is position one mic right over the center of the kit, pointing down at the snare. The second mic is placed above the area immediately behind the floor tom, pointing toward the snare, so that you have almost an equilateral triangle. The mic over the center of the kit is panned to the left and the mic in back is panned to the right. It gives you a wonderful stereo image."

Crown GLM-100 omni lavaliers are also used to close-mic the rack toms. In a medium where appearance counts, Wilson finds these small microphones very useful.

Adrian Zarin is a Los Angeles-based electronic synthesist, composer and free-lance writer, and regular contributor to **RE/P**.

Michael McDonald during a recent musical guest appearance on The Late Show.

"They're very unobtrusive," he says. "We use them very successfully as audience mics too. There are 12 of them hanging above the audience area. They sound good as well. On percussion, they capture the transients and even have the bottom end you need."

Wilson uses an AKG C452 on the snare, a Shure SM94 on the floor tom and an AKG D112 on the bass drum.

"It's sort of a cousin to the AKG D12, which is a mic I discovered one night at [LA rock club] The Whiskey. The D112 has a large diaphragm, which gives us a *very* good kick-drum sound."

Unlike the drum kit, which is at least partially isolated, the band's Yamaha grand piano sits in close proximity to guitar amps, monitor wedges and other sources of sound leakage. Here, Wilson has settled for what he describes as a "quasi-stereo image."

"On the piano, we have a Helpinstill pickup, which I pan to the left, and which covers the entire keyboard. And there's a Crown PZM positioned toward the side on the high end of the piano; I pan that to the right.

"But there is so much leakage into the piano, that I'm not approaching it as I would a normal recording studio stereo piano micing situation. I'm basically just getting the notes in there to be heard.

"The Helpinstill works fairly well, but it's got a limited frequency response; it only goes out to about 8kHz, and there are a lot of overtones that aren't making it. The PZM sounds better, but I get the entire band in it.

"If there's anyone out there who has an isolated piano micing setup, I'd like to know about it."

Micing on other instruments is more straightforward. Each of the four saxophones in the woodwind section is miced with an AKG C460B; Shure SM94s are used on the trumpets and trombones. Percussionist Steve Reid has four Shure SM94s on his setup, the outputs from which are combined in a percussion submix using a Yamaha 8x2 mixer and sent to the principal mixing consoles.

All other band instruments are taken direct, including three electric guitar setups (Hudson, Steve Dudus, Bruce Gaitsch), Jimmy Johnson's electric bass rig, and all electronic keyboards played by Hudson, Gaitsch and Randy Waldman. Although the guitars and bass are taken direct, the players use personal backline amplifier cabinets to monitor their instruments on-stage.

Additional monitoring is provided by a system of custom Audio Tek wedges and Galaxy Audio Hot Spots. Distributed through this system are four different monitor mixes: one for the rhythm section; one for the brass; one for the sax section; and one for Hudson, Waldman and Gaitsch.

"In addition to that," Wilson adds, "we have other Hot Spots on stage for monitoring the program audio, i.e., the boom microphone, Joan's desk mic, etc. We also have a couple of Hot Spots back stage and in the producer's area [directly in front of home base, the desk position located stage right].

"We're always on the verge of 'feedback hell.' There's always a *lot* of monitoring going on."

All audio signals generated by the house band are routed to a 40-input, 8-bus Yamaha PM3000 console, which is located one level below Studio 4 in the production booth. This area also houses the show's main Neve production console (which is manned by production mixer Ray Vaca, who alternates with Ron Worrel) and several submixers. But it's on the PM3000 that Wilson assembles the broadcast mix of the house band that goes out on the air, and the four monitor mixes routed to the bandstand.

"It's not the scenario I would have thought of," says Wilson. "But it's a situation I more-or-less inherited when I took over as music mixer. I've been trying to push for another system, so that I can have a dedicated monitor mixer upstairs; but I'm having problems getting that OK'd.

"When I first began mixing the show, the monitors were coming off an output matrix from the Yamaha. Whenever anything changed in the mix, it altered what went up to the monitors. I immediately remedied that situation by making the monitor mixes pre-fader. So, for the most part, the monitor mix doesn't change. It's just a balance and I leave it alone. Some of the guys play louder on some nights than they do on others, but that's the only aspect of the mix that changes."

The house band takes up a total of between 33 and 36 inputs, which means they can be accommodated conveniently with the PM3000 with inputs to spare for guest vocalists.

Wilson's mix of the house band is sent, in turn, to the principal production console. External signal processing devices are added to the mix via three, 12-input Yamaha M512 submixers. The effects complement consists of eight Yamaha SPX-90s and five Yamaha REV7s, which Wilson divides between the house and guest bands.

"The ability to recall reverb and delay programs rapidly makes the 90 and REV7 particularly useful," he says.

"I really need that ability to recall stored programs. With the guest bands, especially, I rarely have time to really go hunting for echo sounds. I usually use a gated-echo setting on [house-band drummer] Vinnie Colauita's kit; or, actually, different gated-echo programs for the snare and the toms. And I have several delay and reverb programs for things like the saxes, or Mark's vocals. Depending on the needs of the song, I'll obviously use different programs.

"For guest bands it's basically the same setup. When Eddie Money was here, for example, we had a big gated-echo sound on the drums. I used another device for

Close-up detail of micing used to cover The Party Band, which featured saxophonist Beverly Dahlke, a.k.a. "The Tramp" (left).

echo on a vocal duet with Eddie and Ronnie Spector. And I got a slap setting on a third device to really give it that Phil Spector 'Wall of Sound' effect."

Mixing for stereo

In building his mixes, Wilson says he remains acutely aware of the dbx/Zenith MTS (multichannel television sound) format and its practical limitations.

"At best, there's only 30dB to 36dB of stereo separation for the viewer," he explains. "And that's assuming everything is perfectly aligned on the receiving end, which is not always the case.

"In terms of stereo panning, I find I've brought things in toward the center more. I've realized that what I'm working with here is a kind of 'glorified mono,' rather than *true* stereo. I'm trying to get it to sound 'punchy' in both mono and stereo formats."

The engineer monitors the air mix on a pair of small Westlake BBSM series monitors.

"Stereo imaging is crucial and difficult," he says, "due to the fact that you are going through that stereo generator [for MTS broadcast]. Because of the small

Compromises of Stereo Music Production for Television: A Conversation with Mark Hudson

As The Late Show's *music director and bandleader, Mark Hudson often finds himself playing a mediator's role. For him, the Party Boys & the Tramp occupy a "middle ground" between the Johnny Carson and David Letterman bands: Doc Sevrinson's Vegas-style outfit, vs. Paul Schaffer's 4-piece rock and roll band.*

"Joan Rivers' personality falls somewhere between Carson and Letterman and we get an amazing variety of musical guests on the show—from ex-Plasmatic Wendy O. Williams to Vic Damone. The band has to be able to accommodate all of those styles.

"My concept was to put together a great rock-and-roll rhythm section with horns—not a big band as we know it, but one that is able to cover that territory as well as the rock material."

From an acoustic standpoint, however, Hudson's large group does present some problems.

"Monitoring is difficult when you're dealing with a lot of musicians in a small area," he concedes. "We're working it out slowly but surely, though. The key is the drums. If the horn players can't hear the drums, for example, they tend to lag behind. So we make sure there're plenty of drums in everyone's monitor.

"The studio's high ceiling is another area of concern for me. I'm trying to get a baffle dropped over the whole band area, so that the sound doesn't keep going up. But here's where the politics of my job comes in. Before we can do something like that, I have to convince the people who say it won't work visually."

In addition to mediating between divergent musical styles, Hudson often has to reconcile the demands of picture vs. sound. In both these areas, he benefits from the fact that quite a few of the band's instruments are taken direct. Specifically, this includes the Party Boys' collection of electronic keyboard instruments. Chief among these is a Fairlight CMI series III, which is fed to the broadcast and house PA consoles via the instrument's composite monaural output. The Fairlight's principal role is to provide string voicings and other orchestral timbres (tympani, etc.) not available from other band instruments.

Yamaha YCAMS equipment makes up a significant portion of the remaining synth/keyboard gear used onstage. Keyboardist Randy Waldman and guitarist Bruce Gaitsch each play a DX-7. Hudson plays a third DX-7, which has been customized by Chicago technician Don Johnson to store eight banks of 32 voices each on-board. Mark also uses a Yamaha KX-100 MIDI keyboard controller and TX-816 rack. His principal guitars are a Fender Stratocaster and a vintage Rickenbacker 12-string, which go through a Yamaha G100 backline amplifier. The set-up also includes a Yamaha SPX-90 effects unit for processing all these instruments.

"I was using a different amp for a while," Hudson says, "but it didn't seem to be as bright as the Yamaha. One of the biggest problems with television, musically, is that it's all midrange—put a guitar and piano together and it's death. I'm trying to separate those sounds by making the guitar crisper. I think that, with Thom Wilson's help, we're going to be able to expand that [apparent] frequency range a bit."

While the show is in progress, part of Hudson's role is to coordinate the Party Boys' music with all the other elements of the show. Accordingly, he's fitted with an earpiece that ties him into the talkback system used by the director and camera crew.

"I'm hearing that continually," he explains. "It's murder when you're trying to count down a song or play a tricky passage!"

A small footswitch-activated condenser mic, mounted on his music stand, enables Hudson to speak to the director and camera crew, allowing him to arrange spontaneous sight gags involving the band and to transmit other messages. He also has a small video monitor on the bandstand, which, among other things, facilitates smooth, tight endings coming out of commercial breaks.

"I never liked the way they did it on The Tonight Show. *Not to be negative about Doc, but I never cared for the way the band would just kind of fall apart when they came back from a commercial break. I wanted it* tight.

"During our first run-through shows, I did it like that and it frightened Joan; it was too abrupt. They told me: 'We've got to do it like the Carson show,' and I said, 'Oh please, not *that.'*

"I've now worked out a method where—when I see we're coming out of the break—I bring the band down in volume on a count of four. And then—still on beat—I cut them off. It's a smooth ending, but not quite as abrupt as before."

Small, tactful compromises like this are a matter of daily routine for Hudson. But musical and audio quality are two areas where he tries to avoid compromises.

"I can recall doing TV appearances with the Hudson Brothers. It would sound pretty good in the room; but, when you went home and listened, it was terrible. All you'd hear was one tambourine and one guy singing—all out of balance.

"Because of that kind of situation, a lot of acts have tended to shy away from live television. What I'm trying to do is regain their confidence and get them to do the show by offering the best players and the best audio possible."

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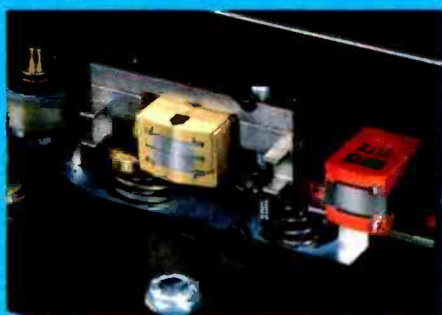

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Video master control at Fox Television Studio during rehearsals for "TLS."

speaker in most TV receivers, I really have to know what's going out."

Occasionally, Wilson will sum his left and right output buses and monitor monaurally on a single Auratone 4C Sound Cube.

"But I really just spot-check it," he says. "I'm primarily mixing for the upper and middle range of playback systems. I'm *not* mixing it for the worst case."

Guest bands

In terms of micing and mixing techniques, guest bands on *The Late Show* are handled in much the same way as the house band. This, however, is limited by two factors: first, there are fewer available inputs on the broadcast mixing consoles used for guest bands; and, second, there is considerably less setup time for a one-night-only guest band, as compared with the house band.

In order to anticipate the needs of guest bands, Wilson tries to do as much advance coordination as possible. But even with solid preparation, the show's daily schedule can result in one or two challenges. The group's equipment typically arrives at noon, and the goal is for Wilson's crew to set up and do a sound check by 3:30 p.m., when director David Grossman begins the camera block.

"Once that begins," Wilson explains, "David has to hear everything. He can't just listen to the drums or the bass alone while he's getting his shots lined up and figuring out what he's going to do to make things work visually. That means if I don't get a technical sound check [i.e., before 3:30 p.m.], I have to get it together in two or three run-throughs. If David's happy with the visuals, I get two. If he's having a problem, I get a break and we have to go through the song three times.

"We generally keep the songs to about three minutes, which means that I sometimes get nine minutes to make the mix happen.

"From 5 p.m. to 6 p.m. is mealtime; according to union rules, we can't do any work then. The house band has its rehearsal between 6 p.m. and 7 p.m. and from 7 p.m. to 8 p.m., the audience is loaded in.

"So, once five o'clock comes around, the way it sounds is the way it's *going* to sound on television. But, rather than compromise, I approach everything as though I were making a record. We all just have to work a lot faster than we would in the recording studio."

The first 10 to 12 inputs of the Neve production console are reserved for guest bands. Wilson generally uses these for bass, drums and vocals. An additional 16 inputs are provided by a Soundcraft series 200D, which acts as a submixer for the Neve. All told, some 27 inputs are available for the guest band broadcast mix. As a result, although Wilson uses many of the same micing techniques, occasionally it becomes necessary to abbreviate some of them. He may use his usual dual-overhead mic configuration on the drum kit, for example, but use fewer close mics to supplement it.

"When Jeffrey Osborne was on the show," he says, "the band had a lot of equipment, so I ended up handling the entire drum kit with four microphones: kick, snare and the two Crown overheads. I used the overhead configuration to capture a basic stereo image on the kit and the close mics for the transients and maybe a little bottom end."

Guest bands are set up on movable risers, which are rolled into their center-stage position when it's time for the band's performance. Microphone and direct box cables are split and routed to several destinations. Guest-band signals are routed to the downstairs production booth, to the house PA board and to the monitor console located backstage and manned by Chris Smith. The monitor mixing console is a 40-input Soundcraft series 800B, which formerly served as the principal house PA console. (That function was recently taken over by a 40-input Yamaha PM3000, which is identical to the board Wilson uses for his broadcast mix of the house band.)

When a guest vocalist performs with the house band, a slightly different monitor mixing procedure goes into effect, as Wilson explains: "Chris takes splits [from the vocal microphones, which are sent to Wilson's broadcast console] so that he can put monitor wedges on the floor for the artists. I then send him a composite mix of our band from one of the output matrixes on my console. He blends that mix in his monitor console, along with the guest vocals and background singing.

"Even then, I'm doing the monitor mix that goes to our band and he's doing the mix that goes to the guest artists. And all of this, of course, also goes to Mike Abott, who does the house PA mix."

Studio sound reinforcement

The mix position for Studio 4's house-sound system is located at the rear of the upwardly sloping audience area some 20 feet above stage level. This station includes a 24-input Soundcraft series 200B submixer that accommodates guest bands. As mentioned previously, the principal house PA mixer—which handles all other audio inputs—is a Yamaha PM3000.

When *The Late Show* first moved into Studio 4, Wilson determined that the existing house-speaker system—consisting of eight cabinets—was inadequate for the show's musical needs.

"The system was all midrange and it sounded horrible," he says. "It was underpowered, so that when we turned it up to the point where people could hear it, it was distorted. The existing thinking was that the studio audience was a secondary consideration, as compared with the 20 or 30 million people viewing at home. I tried to modify that thinking, because the house audience sets the tone of the whole show. If the sound is not working in our room, it's *not* going to

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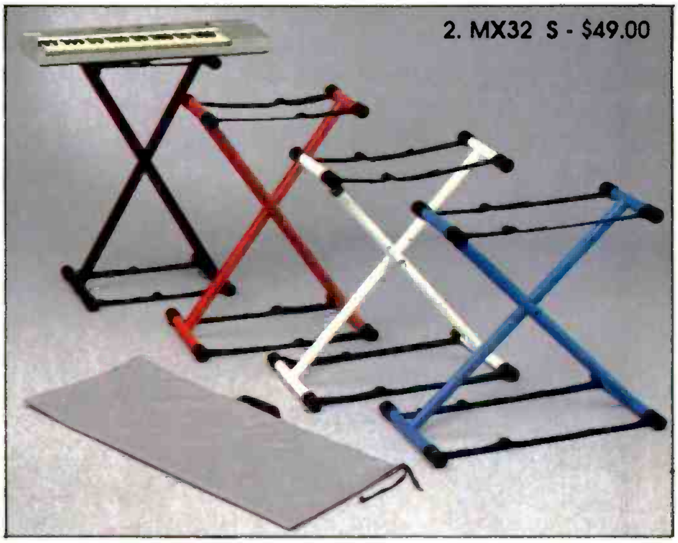

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The audio production crew responsible for "The Late Show" *include (clockwise from left): Jeanette Sutor, who handles stage duties, including band micing; Ron Worrel, who handles the dialogue submixing and master mix; Brock Miller, who is responsible for carts and tape duties; Michael Abbott, who handles house PA mixing; John Emerson, a boom and micing assistant; Chris Smith, the on-stage monitor mixer; Yasmin Muniz, boom operator; and Thom Wilson, audio production supervisor and music mixer.*

work in someone's living room. As a result, we reached out a little more on the PA."

The existing system of Altec cabinets was augmented with six Audio Tek custom, full-range cabinets and two Audio Tek custom subwoofers.

"The fear was that we'd be walking the line a little too closely, because of the audience mics and the boom mic. When you run the PA too hot, it sounds very 'hollow.' My idea was not to run it as loud, but just to disperse more transducers, so that you could move more air."

Even at these relatively low levels, the danger of feedback remains formidable, and precautionary measures are necessary.

"For example," Wilson says, "we EQ the boom mic differently in two different positions. One of these is the monologue position [center stage and very close to the audience] where the mic is quite near the PA system. The other is the desk position, or what we call 'home base.'

"We split the boom-mic feed and bring it up on two different channels, each with different settings on a sixth-octave equalizer."

Transmission: procedures and problems

The show's final broadcast mix, which is output from the production booth's principal console, is routed to another studio in the Fox compound, Control 3, where commercials are inserted. From there the signal goes to Fox's satellite uplink for on pass to East-Coast affiliates via AT&T Telstar 301. This live transmission begins at 8 p.m. There is also an 8:30 p.m. delayed feed for affiliates who do not want to interfere with their 11 p.m. news programs. The delayed feed is recorded on an automated Beta cart system using special controlling software that makes it possible to play back the first half-hour of the show while continuing to record the second half. When delayed playback of the first half is completed, the second half is ready to play back. For the West Coast feed at 11 p.m. PST, the show is pre-recorded on a Sony VPR-2P 1-inch VTR.

Reception of the live transmission at various downlink sites has proved to be a delicate and difficult business on occasion. Much of this would seem to stem from calibration problems.

"There's a lack of setup time in many cases," Wilson says. "What I discovered during my first week here—when we were getting disparity reports from different places—was that a lot of stations come on the network only 10 minutes before we go on air. I had always assumed there would be more time to transmit reference tones to them, so that zero on my console would translate as zero for them. That seems logical, doesn't it?

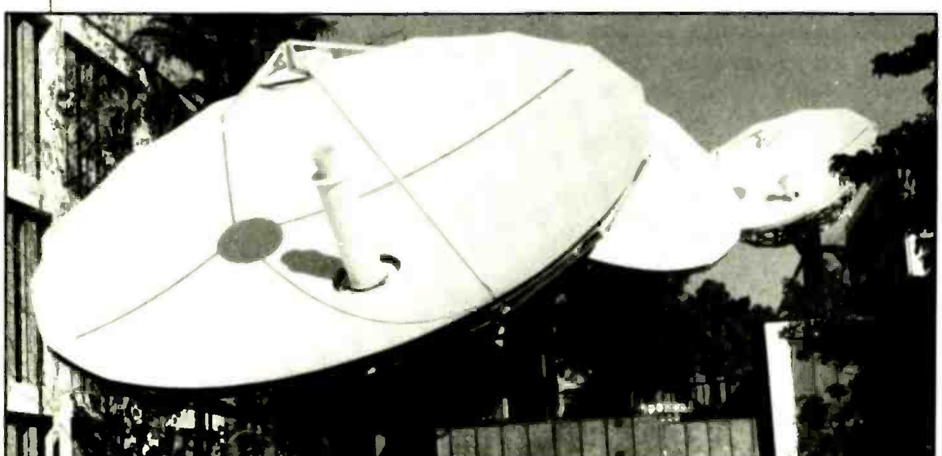
"But one night, we shipped a tone to New York on the satellite and found there was 4dB to 5dB of discrepancy. And this was about five minutes before we went on the air! We had had some complaints about audio 'splattering'—which, in essence, is overdriving the VTRs on the receiving end. And all it turned out to be was a calibration problem—*their* zero wasn't *our* zero VU."

"All you can do is put out the best signal you know how to, and hope that it's picked up properly on the receiving end. But you never know. I found out that some affiliates tape the show as it comes in from the satellite, so they can broadcast it whenever they want to later on. Who knows what kind of equipment they're using and what's happening to the audio.

"What I'm trying to do is get a sort of grass-roots movement going, so that we can have a show recorded at all these different affiliates and then have them send it back to us. This way we can hear what it is sounding like on *their* end."

Despite this tangled web of potential pitfalls—or perhaps because of it—Wilson says he derives a unique satisfaction from doing the show.

"For me, it's a different world. All of a sudden there are 20 million people out there listening to a mix that's going down live. There's no turning back. It's something that combines a few elements I've always loved about records—namely, cutting tracks, where you're capturing a performance, and mixing. It's a mixture of real excitement and real terror that puts an edge on everyone's performance. And that's a feeling I had lost making records." R·E/P



Satellite uplinks on the Fox lot used to beam live video and stereo audio for live broadcast to the East Coast. Other time zones utilize tape-delayed transmissions.

Photos by Erik Heinila, Fox Broadcasting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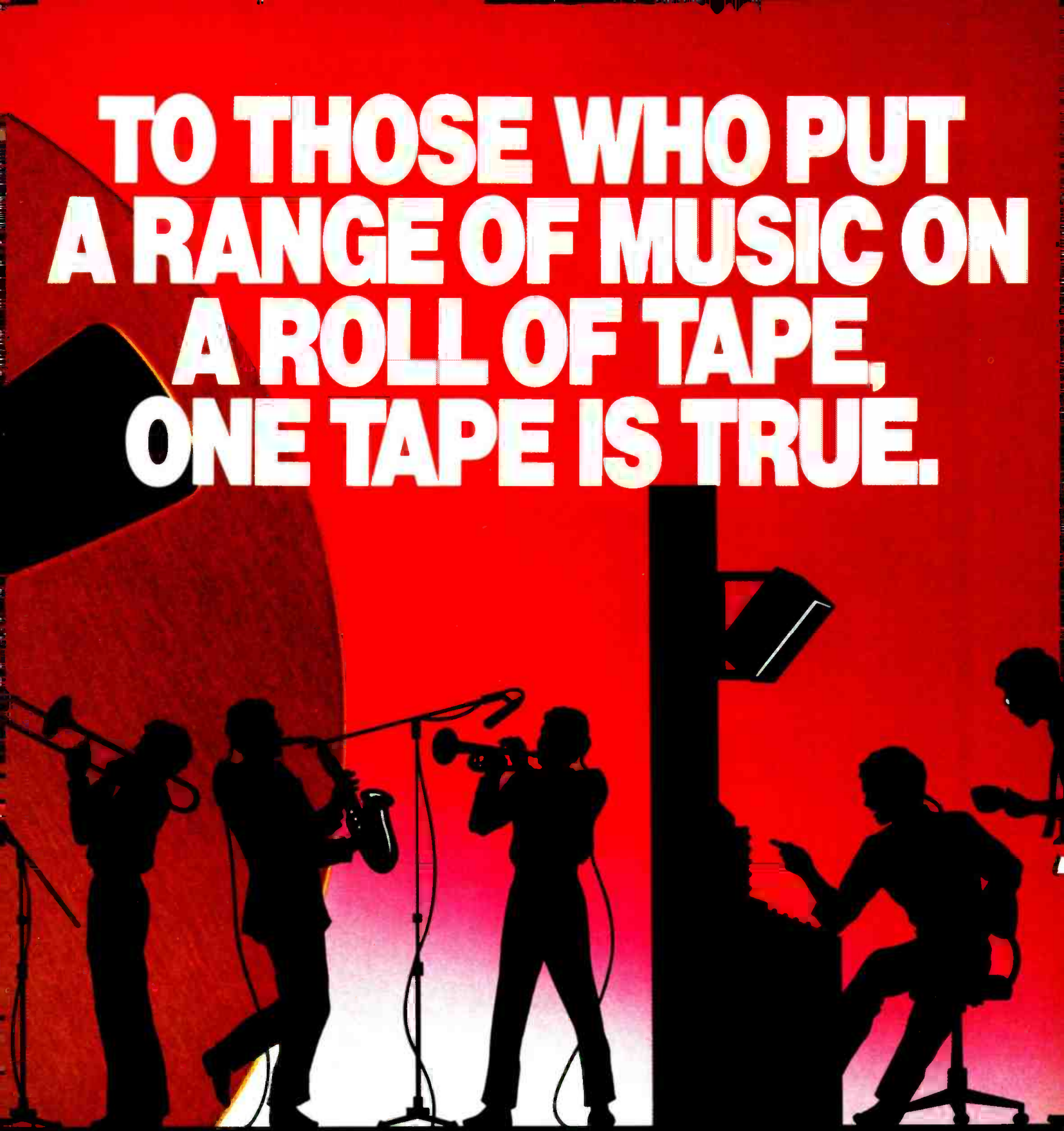
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A ROLL OF TAPE,
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Techniques for Producing Commercials and Jingles

By Terry Fryer

Electronic music composition, digital sampling and multitrack recording are finding wider application in the world of stereo commercials production. A recent 30-second Masterlock TV spot provides an interesting case example of what many consider to be a wave of the future.

The making of a television commercial involves a number of highly technical and creative stages. One aspect of commercials production that has seen the most radical change over the last decade has been the use of music and sound effects to complement and enhance the visual images.

The easiest way to bring you up to date on these techniques is to describe the step-by-step production of a 30-second spot I recently produced for Masterlock—a commercial you may have seen on the air lately. In addition to detailing the various audio techniques I used to produce this spot, I'll also describe a number of other processes that can be used to create music and sound effects for a commercial.

Terry Fryer is a leading synthesist and electronic music composer, and co-partner in Colnot/Fryer Music, a Chicago-based commercials and music production facility.

Client input

The first step in the process is always the initial client input: What does the client want? Unlike some commercial assignments, the three major directions from the client in this case were straightforward:

- Create a powerful, rhythmic track that complements the picture.
- Musically highlight two or three important visual cues.
- Create a realistic sound effect each time the locks close.

Oh, and one more thing: Do it by tomorrow!

First, of course, we had to take a look at the edited visuals for the spot. The 30-second commercial, which had been shot on film, consists of a rapid and visually dramatic sequence of scenes of people closing locks: a girl at her school locker; a guard locking a gate at night; a man closing a strong box; a young woman locking her bicycle, and so on. The spot closes with the familiar Masterlock signature—a lock being shot with a rifle—and the announcer making a terse comment about how tough the products are. In fact, the only voice-over copy comes at the end of the commercial.

The next step was to construct some type of map that would help the process of writing a piece of music to match the picture elements. The easiest technique for this is to note the position of all the

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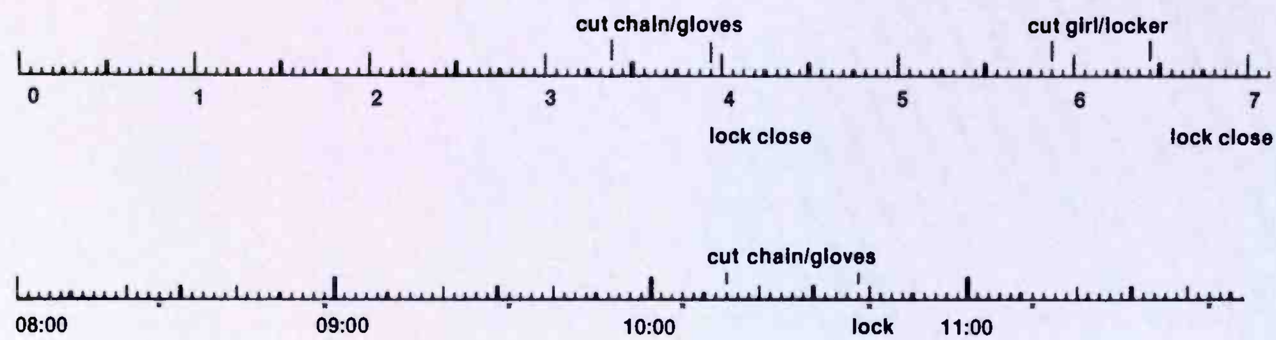


Figure 1. Cue locations for the Masterlock commercial were entered on a bar sheet representing 24fps film feet and frames (top). For example, the visual lock closure at 3 feet and 6 frames is shown as the second vertical tick mark. The same cues can be entered on a bar sheet representing 30fps video frames, as shown in the lower annotation. (Note that the time code on the supplied video worktape started at 08:00 seconds, rather than zero, hence the offset from the picture start.) Other visual cues are listed in Tables 1 and 2.

visual and sound cues. In this case, the most obvious cues were the repeated closing of the locks, as well as the cuts between scenes.

I had a secret hope that the spacing of these cues would set up some sort of rhythmic pattern that could be used to create the music and rhythm tracks. If so, the music would ride, as it were, on the rhythm set up by these cues.

Film and video cue sheets

Several methods are available to make the kind of map I needed to determine if my hopes would come true.

Method one involves sitting with the editor and writing down cue points while he or she runs the film. 35mm film is projected at the rate of 24fps and is measured and notated in feet and frames. There are 16 frames for every foot of film.

The editor has a counter on the flatbed or Movieola, which displays how many frames and feet have gone by. The method is simple: the editor stops the film at each visual cue and we write down the feet and frame location. When we had finished, the cue list for the first part of the spot looked like Table 1.

Method two is used where the commercial has been shot and/or edited on videotape instead of film. In the United States, videotape usually moves at the rate of 30fps and each frame is identified uniquely with a time code location. If you want to work on videotape, but the commercial has been shot and edited on film, ask the editor to make a transfer from film to tape and add audio and visual time code. (Because there are a number of time code formats for videotape, specify 30fps, non-drop, which is what keeps all of us music types out of trouble and in sync.)

Time code is measured in hours, minutes, seconds and frames. (Some devices also construct subframes in divisions of 80 or 100 per video frame.) Armed with a properly constructed videotape you can now repeat the process described above that I used for film (in the comfort of *your* living room, of course). A time code listing of the same audio and visual cues for this commercial is provided in Table 2.

Method three utilizes a MIDI sequencer equipped with a MIDI-to-time code interface. Ask the editor for a videocassette with time code on one of the audio channels. Connect the output from your VCR's audio time code channel to a MIDI sequencer TC input. Put the sequencer into record mode and have the composer play whatever music seems to fit in this hit-or-miss fashion, experimenting until the music begins to work against picture.

This latter technique, while having the advantage of being very simple, does require some experimentation before the composer comes up with suitable music. (If the MIDI sequencer is less sophisticated, by the way, and cannot read time code, you'll need to provide another device, such as the Garfield Master Beat or Roland SBX-80, to act as a translator between the time code from the video deck and MIDI timings.)

Whatever the method being used, you'll need to put the information relating to the visual cues into a format that makes sense. The traditional next step is to use a bar sheet, as shown in Figure 1. A bar sheet has markers that represent each foot, frame and second of film or videotape. Cues are placed on the rule at their proper locations.

The next step is for the composer to choose a tempo for the piece and place it on the top of the bar sheet. (You can maybe see where all of this is going; my hope while composing the Masterlock piece that the cues would make up some sort of rhythm, were still alive at this point.)

A click book is a tool that can be used to answer the question of whether or not the cues match up with a rhythmic pattern that would be recognized in the Western World as music. Basically a click book lists locations of beats for various tempi. In use, a composer would place the cues he or she wants to catch in a list and then search through the various tempi to find the ones that come closest to

Table 1. Annotated visual cue list in film feet and frames for the first part of the 30-second Masterlock TV commercial.

Feet/frames		Cue name
03	06	Cut chain/gloves
03	15	Lock closes
05	14	Cut girl/locker
06	07	Lock closes
08	09	Cut locker
08	11	Lock closes
09	09	Cut gun
10	04	Lock closes
11	06	Cut dog
11	12	Lock closes
		...and so on.

Table 2. Time code locations for the cues listed in Table 1.

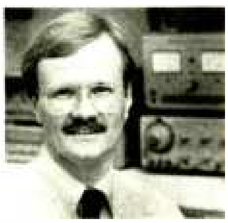
Time code location	Cue name
00:00:10:07	Cut chain/gloves
00:00:10:19	Lock closes
00:00:11:27	Cut girl/locker
00:00:12:08	Lock closes
00:00:13:21	Cut locker
00:00:13:24	Lock closes
00:00:14:11	Cut gun
00:00:14:24	Lock closes
00:00:15:17	Cut dog
00:00:15:25	Lock closes
	...and so on.







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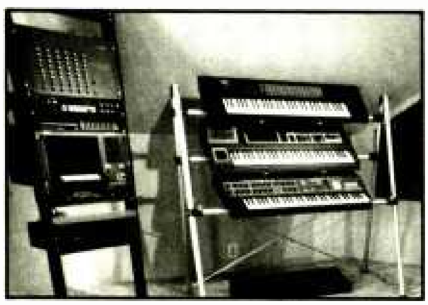
Beat	$13^3/_8$ Tempo Feet	$13^3/_8$ Tempo Frames	$13^4/_8$ Tempo Feet	$13^4/_8$ Tempo Frames	$13^5/_8$ Tempo Feet	$13^5/_8$ Tempo Frames
1	0	13.36	0	13.50	0	13.63
2	1	10.75	1	11.00	1	11.25
3	2	8.13	2	8.50	2	8.88
4	3	5.50	3	6.00	3	6.50
5	4	2.88	4	3.50	4	4.13
6	5	0.25	5	1.00	5	1.75
7	5	13.63	5	14.50	5	15.38
8	6	11.00	6	12.00	6	13.00
9	7	8.38	7	9.50	7	10.63
10	8	5.75	8	7.00	8	8.25
11	9	3.13	9	4.50	9	5.88
12	10	0.50	10	2.00	10	3.50
13	10	13.88	10	15.50	11	1.13
14	11	11.25	11	13.00	11	14.75
15	12	8.63	12	10.50	12	12.38
16	13	6.00	13	8.00	13	10.00
17	14	3.38	14	5.50	14	7.63
18	15	0.75	15	3.00	15	5.25
19	15	14.13	16	0.50	16	2.88
20	16	11.50	16	14.00	17	0.50
21	17	8.88	17	11.50	17	14.13
22.	18	6.25	18	9.00	18	11.75

Figure 2. *Three tempi from a click book, listing the film feet and frame locations of beats for tempi of 13 3/8, 13 4/8, and 13 5/8 film frames per beat. As will probably be readily apparent from Figure 3, the author selected a 13 3/8 film frames per beat tempo for the Masterlock commercial, to provide musical hits at the corresponding lock closure locations.*

The author pictured in one of a pair of production studios operated by Colnot/Fryer Music, Chicago. The consoles comprise a pair of Hill Audio 16x2 mixers for monitoring via a pair of UREI model 809s. Below the consoles can be seen the music keyboard for the facility's Fairlight CMI series III digital synthesizer, and to the left a Linn 9000 drum machine/sequencer. A separate side room houses the CMI's electronics rack, various U-matic VCRs, tape library and an Otari MTR-20 24-track.



In the rear of the room are located a separate keyboard submixer, PPG Waveterm controller and VDU (left) and the facility's main keyboard array: a Sequential 2000, E-mu Systems Emulator II and a PPG Wave 2.2.

matching the cue list. It's a tedious technique, but it can be made to work, as shown in Figure 2.

The most sophisticated and, in my book, the most fun method involves a computer. There are a number of programs available for personal computers that will simplify the process. Having been supplied with the cue list, the program calculates the tempo and generates a listing of where each cue falls. Currently available programs include Auricle, Opcode Film Composer, and Digidesign time code cue sheet, as well as custom programs for the Fairlight CMI and New England Digital Synclavier synthesizers that form part of larger computer music systems.

With these programs, a composer can step through a number of tempi to find the one that catches the maximum number of cues. The software will print out a listing that shows the location of each cue in relation to the beat. Some of the programs have time code sheets that generate MIDI commands at specific time locations.

For the Masterlock commercial, I used a custom program developed by Kelvin Palmer and myself to locate a tempo that looked good against the picture (Figure 3). A couple of viewings later it became apparent that my dream had not come true: I could find a tempo that flowed well with the picture, but there appeared to be no constant rhythm to the closing of the locks. Making a high-level executive decision, I chose to treat the lock closings as sound effects and write a complementary music track.

The music was written, the client approved it and the time came to produce the piece. Based on the client's request, both synthesizers and acoustic instruments were used. A small brass ensemble and rhythm section was paired with synthesized rhythmic elements and harmonies.

The music was assembled on 24-track analog tape on an Otari MTR-90. All synthesizer sequencing was done on a Linn 9000 referenced to a time code track using a Roland SBX-80. The sequenced parts were written and recorded first, and then the acoustic musicians overdubbed to those tracks.

During the acoustic part of the session, the 24-track tape was locked to a ¾-inch videocassette of the visuals using a BTX Shadow time code synchronizer with Softouch controller. Since the common technique is to synchronize audio and video tapes by hand, eye and ear, our high-tech solution to the problem allowed extremely accurate checking of the cues and made the client a lot less nervous about things in general.

Digital sampling keyboards

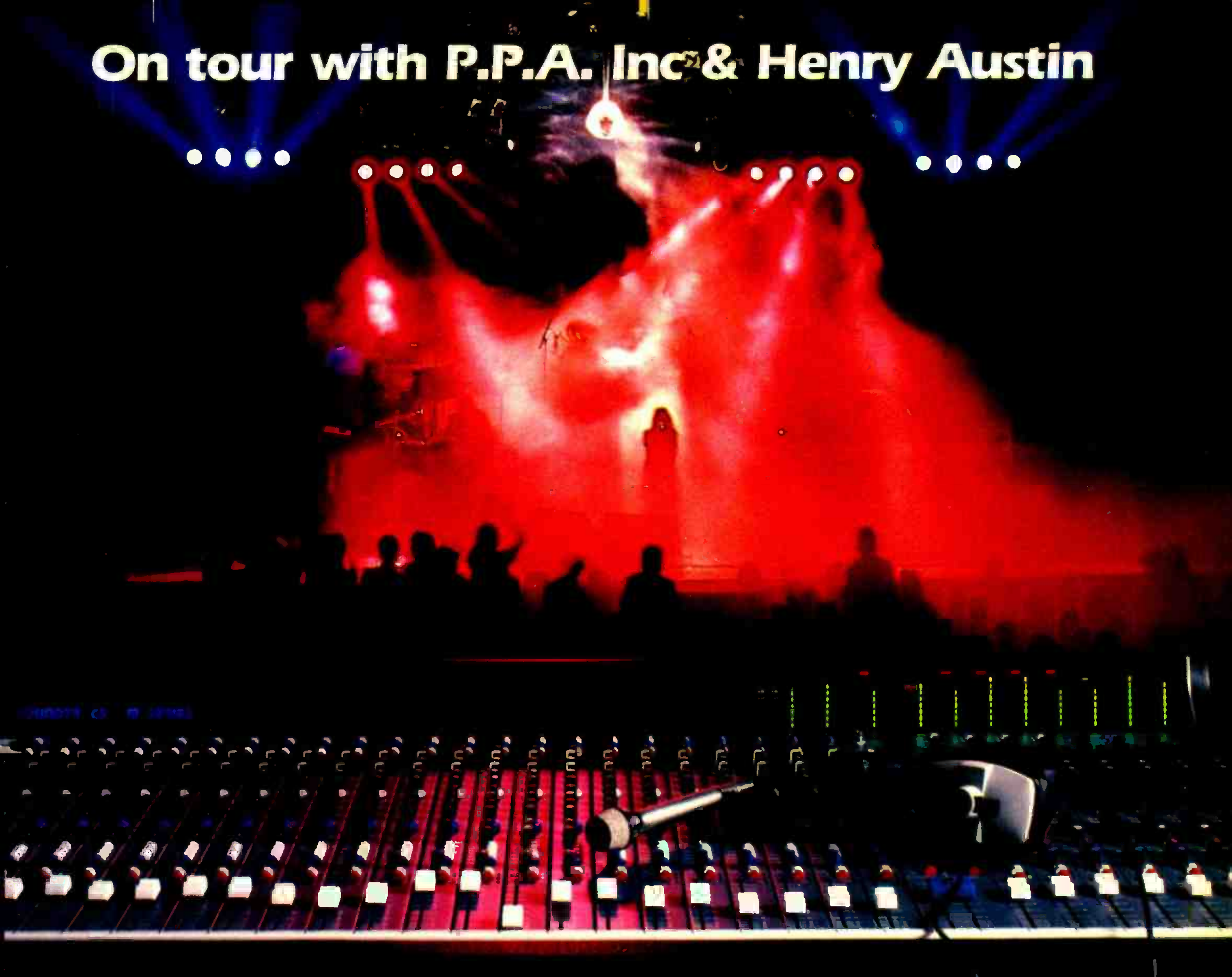
The sound effects were the next order of business. I had access to all of the locks that appeared in the commercial. The lock closings could have been located by traditional music notation and performed (i.e., just close the lock in front of an open mic) but that's not techno enough for me. I chose instead to sample all of the lock sounds into a Fairlight CMI series III digital synthesizer.

Thus armed with a voice file of different lock effects, I moved to the CMI's time code trigger page. This feature allowed me to have the instrument automatically replay the sound of a certain lock—or, at a later stage, the announcer's voice, which I also sampled into the CMI—at a selected time code location in the commercial.

The sound effect of a rifle shot used in the closing was constructed in stereo (all of the lock closing effects were in mono) using the CMI. Working from a basic library of sampled gunshots, noises of torn sheets and some other interesting textures, I transferred the various elements to 24-track, mixed and processed them in stereo and then resampled the final mix back into the CMI.

Once I'd entered the appropriate time code location and picked the MIDI event that I wanted to occur at that location, I simply rolled the videotape and watched everything fall into place.

The final step in the assembly process, adding the voice over, was handled not much differently from any other effect. The selected announcer take was presented on a 15ips mono analog tape. The


On tour with P.P.A. Inc & Henry Austin

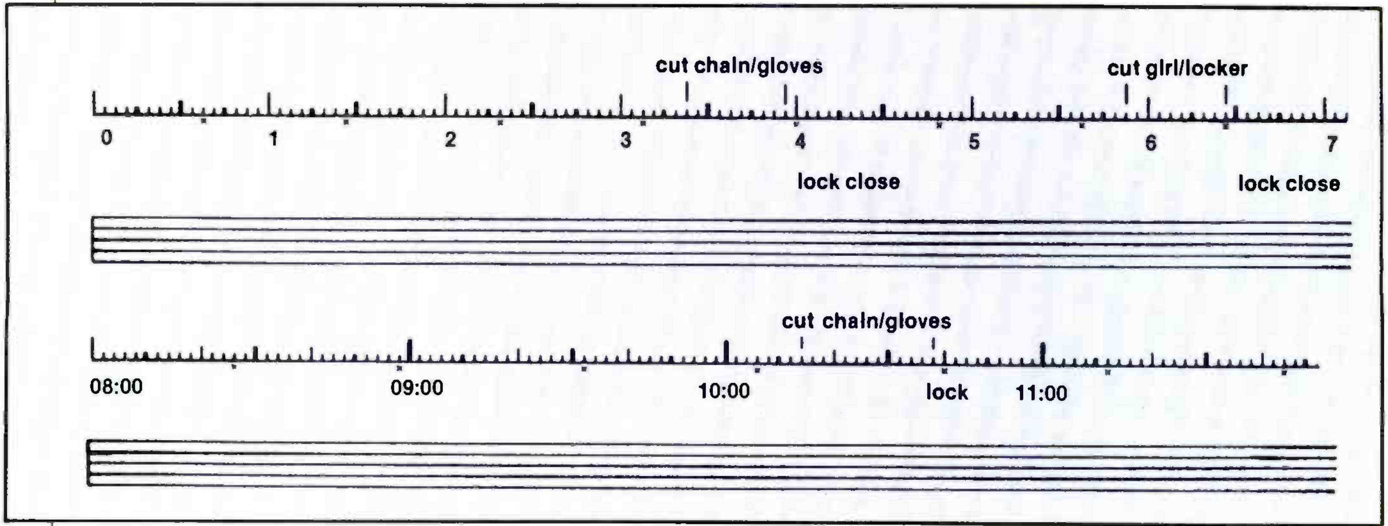


Figure 3. *Working with a custom program, it is possible to provide markings of beat locations at various film frame per beat tempos. Here, in response to the author selecting a 13 3/8 frames per beat tempo, the program has printed small "x" marks on the bar sheets at the corresponding beat locations. The top bar sheet is marked in film feet and frames, while the lower sheet is for 30fps video, as shown in Figure 1.*

A second commercial and jingle production studio is centered around a Biamp console, a Sony JH-24 multitrack and Calibration Standard Instruments MDM-4 close-field monitors.

normal production dictates transferring the tape to 35mm magnetic stock and adding and removing pieces until it syncs with the picture. Again—not techno enough for me.

The voiceover was sampled into the Fairlight CMI as a complete piece. Using the time code trigger page, I chose a starting point for the announcer. With the CMI's editing facilities it was a simple task to move the elements around until the voiceover was perfectly matched to the picture. This element was then transferred to the 24-track master.

Commercial remix

A traditional film mix approach dictates that the music be mixed separately from the sound effects and dialogue. Each of these elements is placed on a separate 35mm mag track and positioned against the film by the film editor. A mix usually takes place in a large re-recording theater where a balance is achieved by the film mixer.

Although this method allows limited control over the balance of individual elements, it is well suited to certain types of TV production.

Another method is to assemble all of the various elements on a multitrack tape. Because of its complexity and the number of voice, music and effects elements, the Masterlock commercial spot was mixed from 24-track on an automated Solid State Logic SL-4000 series console at Chicago Recording Company. To ensure perfect audio/video sync, the videotape was slaved to the 24-track.

During the 24-track remix stage at CRC, I panned the stereo music and rifle effect to their conventional left and right assignments and placed the announcer and mono lock effects in the center of the mix. I suppose that I could have alternated the lock closures left and right to add some sonic interest, but this might have been too distracting for the viewer. Also, the client wanted to keep the audio movement to a minimum, so that the visuals would carry the intended message of the 30-second commercial.

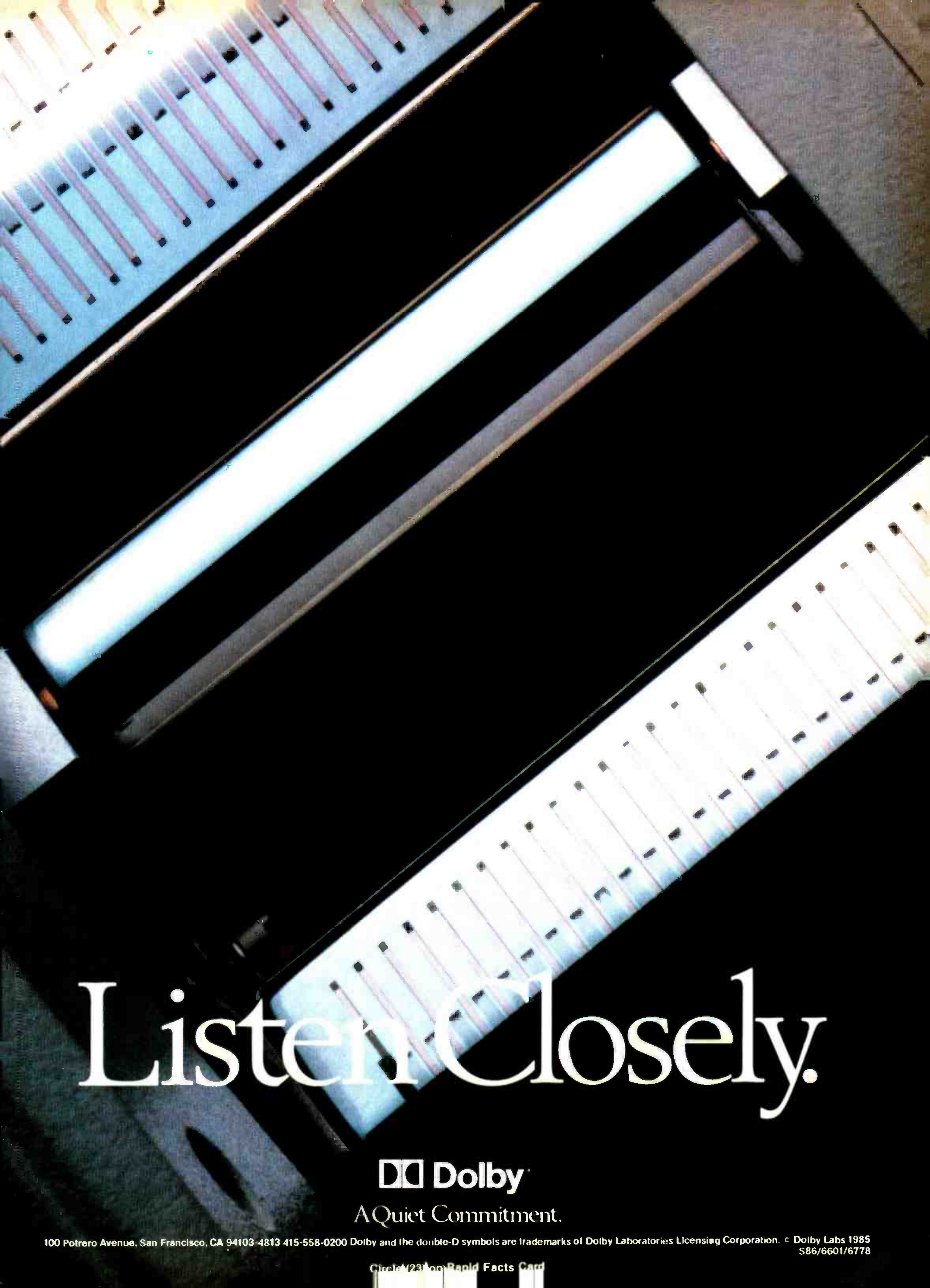
The final format of the soundtrack for the commercial was a 30ips, 4-track analog tape for stereo broadcast with left and right audio on tracks 1 and 2 and time code on track 4. A mono version was also constructed on 15ips 2-track analog with program on track 1 and a 60Hz tone on track 2. These masters were transferred at the video production house onto the final video master used to air the commercial.

So we come to the end of another day in the life of Colnot/Fryer Music, another happy client and another commercial on the air. Better living through the magic of microprocessors. R·E/P

The keyboard array at the second studio includes (left-to-right): a Fairlight CMI series II, PPG Wave 2.2, Roland SBX-80 time code-to-MIDI converter, Minimoog, Oberheim Expander, Roland MKS-80, Yamaha DX-7 and Sequential T-8 synthesizers.

In rear of the studio is a composition area, where the author can view the U-matic video workprint and enter cue locations in film or video frame numbers into a custom software package running on a PC (left). The software provides beat positions for a variety of frame per beat tempi.

Photos by Kelvin Palmer.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Circle (23) on Rapid Facts Card

Design of a Digitally Controlled Assignable Console for Recording and Broadcast Production

By Ken Farrar

As more users of recording and broadcast production consoles look to the coming generation of digitally controlled assignable designs, we present a technical overview of a recent implementation.

Multitrack recording and broadcast production consoles are becoming larger and increasingly complex, making it difficult for engineers to identify and reach the control they want. Moreover, the task of setting and resetting a console is now formidable. In reality, however, the majority of operational controls on a console can be represented for a short time on a small, assignable control panel, and all settings transferred to and from static and dynamic memory.

Our proposed design of an assignable

Ken Farrar is managing director and chief designer at Calrec by AMS, Yorkshire, England.

Section of an 112-channel assignable console for a BBC TV Master Sound Control Vehicle.

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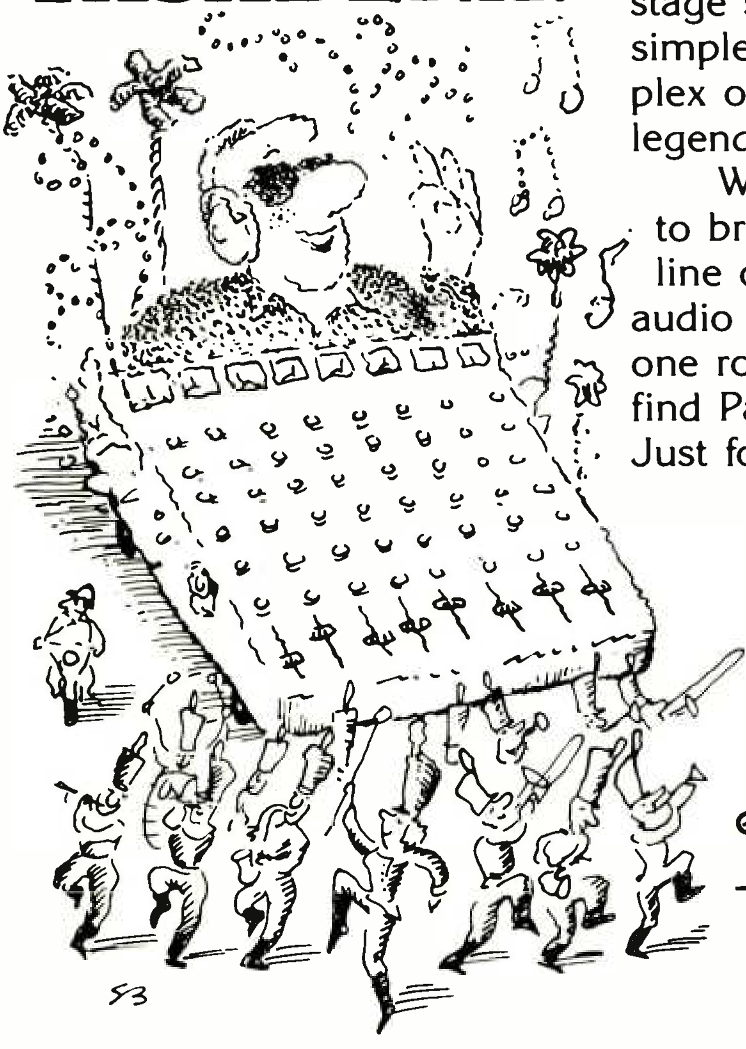

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production console comprises a compact control surface in the control room, and card frame racks sited in a technical area some distance away. The console protocol employs distributed intelligence for maximum speed of operation and security. There are two independent data links to the card frame rack, and each is easily capable of handling the full interchange of information.

This approach for an assignable system allows separate processors to take care of separate areas of the desk, thus providing enhanced security, easier servicing and bench testing, and fast response of controls such as shaft encoders.

Other advantages include a reduction in physical size and placement of better quality controls in front of the operator; complete or partial desk settings may be copied, compared, stored and later recalled as required; and equipment is easier to maintain since the use of an audio rack and cards provides a regular, familiar circuit form.

The system allows for a normal maximum of 128 fader channels, eight freely routed stereo groups, four stereo outputs and 24 or 32 multitrack groups. In addition, faders may be subgrouped in any area of the console.

Channel facilities may be controlled on one or two totally independent assignable control panels. The controls include normal input and gain selection; 4-band parametric equalization; 2-band filters; insert facilities selectable to three points in the channel; optional pan control to groups and outputs, or to multitrack; and up to 16 auxiliary group outputs.

Each section of the channel controls is individually intelligent, and faders are subdivided into intelligent blocks independent of each other. Further independent intelligent systems control all routing and housekeeping controls, plus memory, copy, test and diagnostic facilities.

The system is designed to use digital bargraphs for main, multitrack and auxiliary metering; the main metering can also employ moving-coil VUs or PPMs if required.

The system does not require a patchbay to initiate operation, because microphone-to-fader selections are software controlled and normal paths to insert devices are established in the routing procedure. (It must be realized that unlike all other settings anything patched cannot be committed to the memory system, except by scratch-pad keying.)

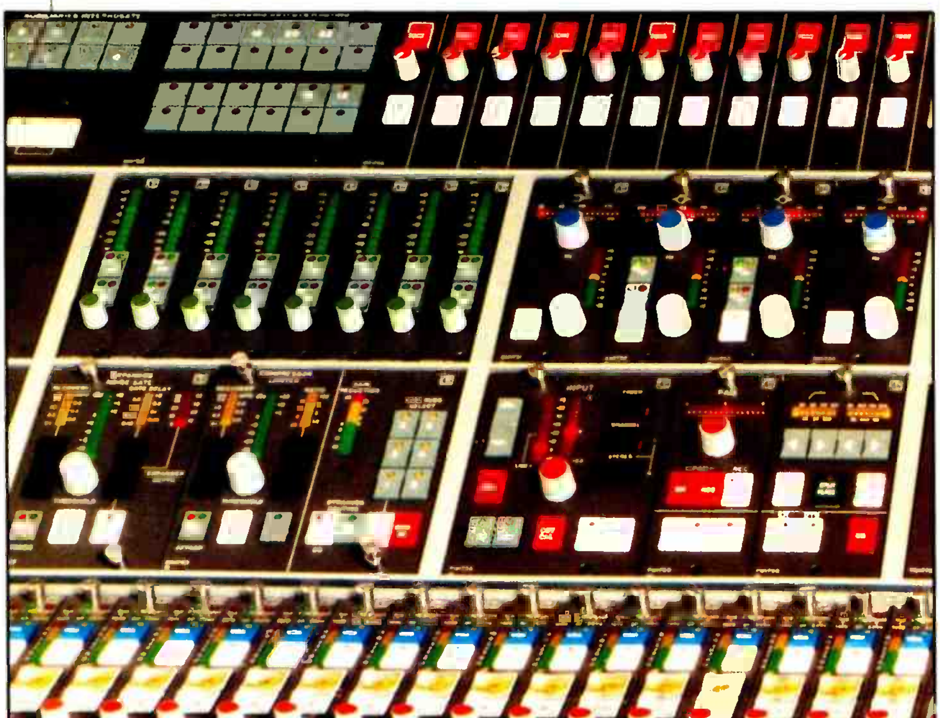
Console layout

The console is divided in three sections:

- A flat fader area that includes discrete fader controls. If assignable channel dynamics are used, a dynamics bargraph is included with each fader.
- An angled control section that includes all assignable, routing and housekeeping controls, main output controls, monitor, meter, talkback and monitor loudspeaker controls.
- A penthouse section that houses the bargraphs and other metering, plus discrete compressor/limiter modules and so on, as required.

All the fader-control modules are identical and interchangeable.

Close-up detail of the assignable control panel fitted to the BBC Television console, showing the 24-track multitrack assign buttons [top left], eight auxiliary masters, 4-band EQ section [center right], expander/noise gate and compressor-limiter section, input gain trim and pan assignments. [lower right].



Card frame racks

These racks house all of the principal analog audio circuit cards, the address decoder cards, overload reader cards, diagnostic cards and power units. They also carry the input, output and control interface connectors.

The audio cards fall into several different categories, some of which may be actively addressed by the console:

- A channel card comprising microphone/line input amplifier, VCA, pan, routing and insert amplifiers.
- An equalizer card with 4-band parametric EQ and 2-band filter.
- A dynamics card with compressor/limiter and expander/noise gate.
- A 2x8x4 switching matrix.
- An 8x3x1 switching matrix.
- A VCA-controlled output card.
- An auxiliary matrix card that supports eight channel auxiliary circuits, each with VCA and pre/post selection, mixed to one output.
- A 4x8x1 master switching matrix.
- A Line output card.

All audio circuits are held on latches that are only addressed and changed when a control change occurs. There are thus *no* cycling digital data and address buses near the audio circuits; similarly, a processor crash, malfunction, power drop out (even of a protracted duration) and so on will not cause malfunction of the latched audio circuits. A control change requires an intelligent discourse between the control processors and the audio-rack logic system.

Interconnections

Studio microphone lines are connected directly to the racks using star-quad cable and XLR-type connectors. Because channel allocation to faders is part of the desk initialization procedure, they do not route through a patchfield. Audio line, insert, multitrack, monitor and tie lines are carried on multicore cables and connectors between racks and the console patchfield.

The high-level data bus between the control surface and racks are small, multicore connections that include

20 10 6 3 1 0 1 2 3
– +
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acknowledge and diagnostic data, as well as control data.

Console operation

Complete initialization of the console is not always necessary because, if required, the console memory can hold several standard settings. Operation requires first that the operator selects the pattern of faders required for the session and marks them accordingly.

A fader may be assigned, using a dedicated Assign button or the main control panel, to an output or a channel, or cause a channel to be set as a Group. Channels may be set to mic, line or tape replay inputs. Channels can be used without faders if required, even in stereo.

Channel and group channels may be routed to other groups, outputs or recording groups with or without faders. Routing is principally achieved by touching the Assign buttons of the fader to be routed while holding down the desired group or output button(s). Faders may be VCA subgrouped irrespective of their assignation.

A VCA subgroup fader (which could equally be called a channel master) acts on the subgrouped channel faders in the same way as if it were a group fader. All other channel functions, such as cut, AFL, PFL, routing, equalization, etc, remain individual to the channel. VCA subgrouping can include the channel located behind the VCA subgroup fader, and avoids using up channels for grouping purposes.

Permanent insert facilities are provided on all outputs, whereas assigned channel and channel-group inserts may be used as separate outputs by using the Insert output function. The latter can only be connected to channels and channels assigned as a group, and selected pre-EQ, pre- or post-fader.

Two assign buttons, Main and Sub, call up the appropriate assignment control panels to the fader. Simultaneously, the designated assignment control panel repeats the fader and channel (or group-channel) number(s), and displays the assigned channel or group channel settings.

All the assigned channel functions, including gain, equalization, filter, pan and auxiliary outputs, now may be adjusted. Control is achieved via toggle-action buttons, nudge buttons or shaft encoders, with single LED or bargraph indication of the settings(s).

Generally, shaft encoders are used where a knob is the familiar type of control, such as auxiliary gain, frequency, amplitude and so on, and buttons for definite step actions. The input-gain setting, therefore consists of 6dB per step adjustment by nudge buttons that, if held down, cause continuous stepping through the range, whereas the trim control is a shaft encoder. Both ranges have a pointer-type LED display.

When a fader controls two channels in stereo, a stereo balance adjustment is possible by pressing and holding a Bal button. A second bar offset from the first appears in the trim display, which offset is then held for all settings.

The high-and low-frequency EQ bands are set for a shelving characteristic; the two mid-frequency bands have bell characteristics with four choices of bandwidth (0.7, 1.5, 2.5 and notch). HF and LF filters each have eight pre-set frequencies and slopes of 12dB and 18dB per octave, respectively.

The frequency displays are horizontal, with vertical amplitude displays below. Shelving and filter bargraphs close in from the ends, commensurate with their action in the real circuit.

Assignable panel displays are dimmed if not actually in circuit, allowing pre-adjustment before introduction at which time they brighten to normal intensity. The pan, frequency and equalizer amplitude controls are all 8-bit (256 step) controls.

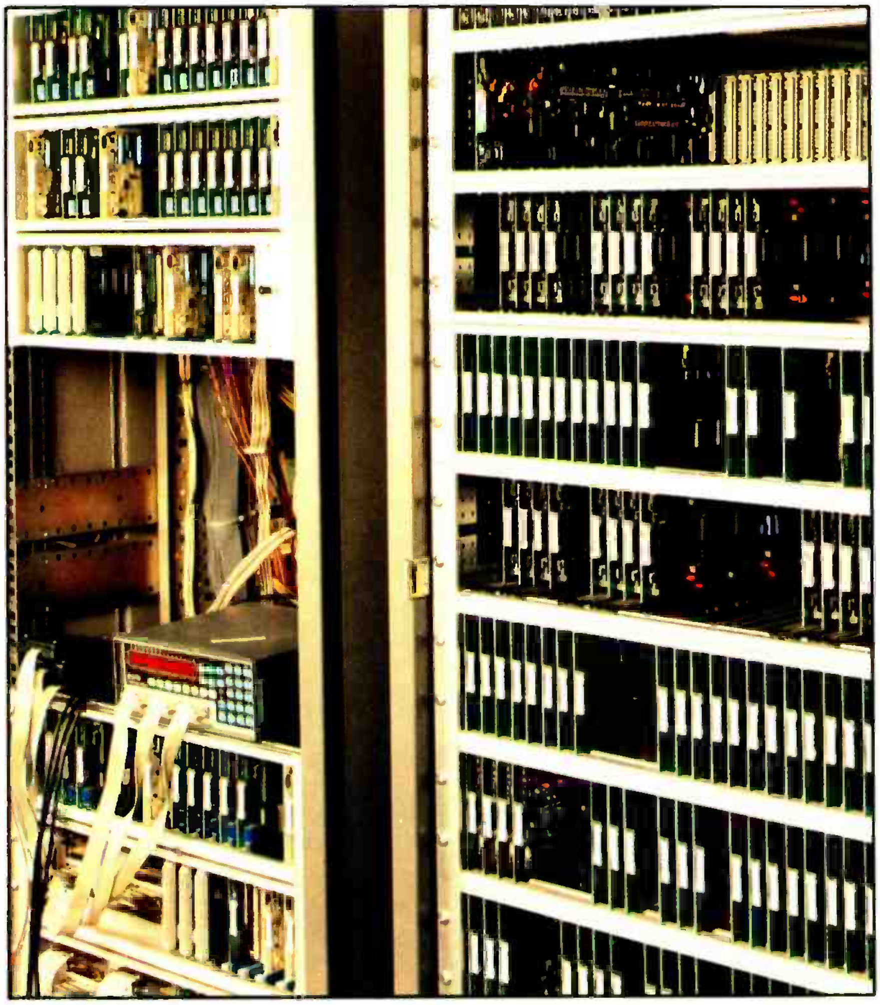
It is possible to select the eight auxiliary outputs from a channel pre- or post-fader to eight A or eight B buses, or any combination of eight (16 outputs in all). The eight A or B outputs also can be controlled on the eight Auxiliary controls. Eight dynamic bargraphs indicators of bus levels are switchable between A and B outputs.

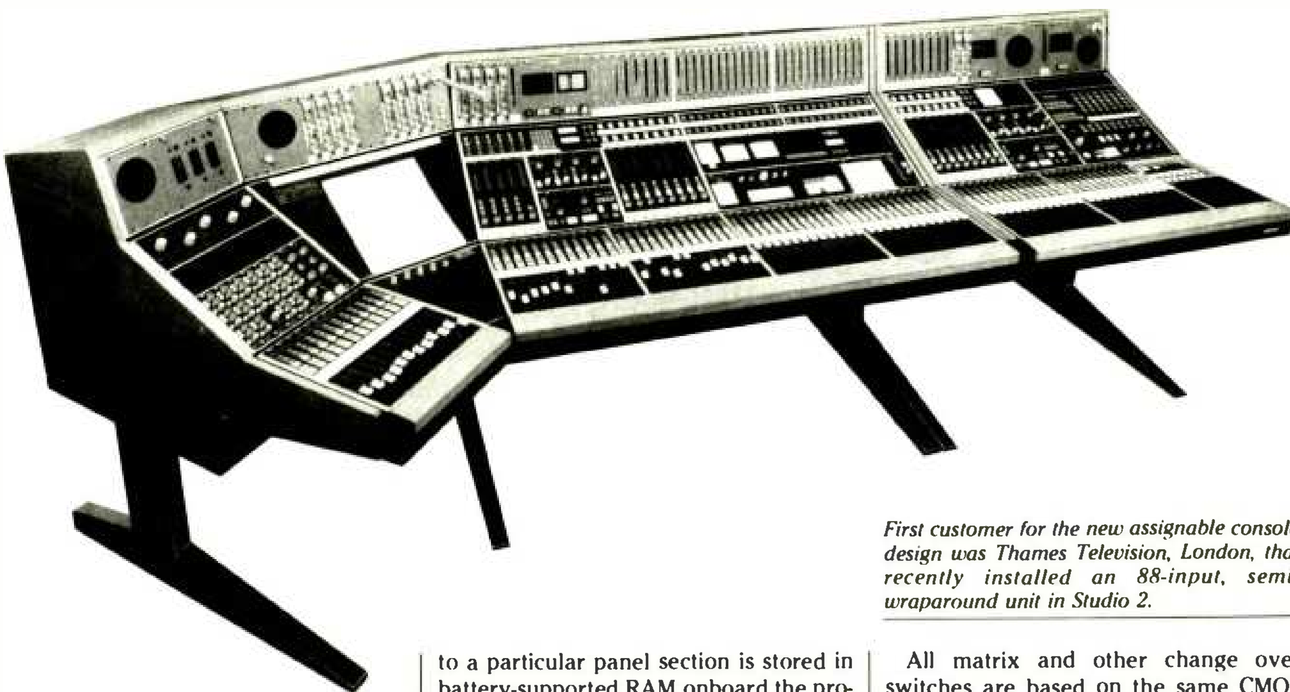
Outputs may be controlled directly on four centrally located controls with bargraph-type level displays, or by assigning a fader to any of them. An additional Main Output control affects all four stereo outputs.

Because recording group outputs usually do not require a great deal of trimming, one control strip can be provided for this purpose and assigned to any output.

Most of the monitor-select functions are presented conventionally, with one

Separate rack units house the analog audio controller cards, plus the talker/listener cards that interrogate digital data signals from the assignable control surface.





First customer for the new assignable console design was Thames Television, London, that recently installed an 88-input, semi-wraparound unit in Studio 2.

notable exception. The buttons used to route and interrogate groups, outputs, recording outputs and interrogate auxiliary outputs are not duplicated on the monitor selector matrices. Instead, a single button allows any of these buttons to change duty to monitor selectors when the sectors of the buttons called Mon (monitor) become operative.

The rest of the loudspeaker control system follows conventional practice, with provision for optional AFL override and mono or stereo listening. Provision is also made for M/S to be monitored L-R. Monitoring can alternatively be routed to a pair of smaller auxiliary loudspeakers.

One channel or group-channel may be copied to another using the Assign and Copy buttons.

The memory system comprises three complete console snapshots stored in RAM, labeled Last, Current, and Next. It is possible to move instantaneously between these settings, or to use the console as a preview (or review) of the settings, without changing the audio.

In addition, up to 30 console settings can be stored on floppy disk and pre-stacked and downloaded in sequence. The sequence can be freely changed or reversed at will.

Digital control system

Each assignable area of the desk requires the collection of shaft and button data from the control panel, and its transfer via the data for actions at the audio rack. A dedicated microprocessor control card using a Motorola 6809 collects the data from the control surface. Data collected from the bus appropriate to a particular panel section is stored in battery-supported RAM onboard the processor card.

The main interchange of information between control units and the rack is via a dual data bus, which is similar to the IEEE-488 architecture but of a more robust nature to enable the transfer of data over greater distances. Access to the bus is provided on a cyclic basis to each control unit in turn.

Each control unit, in general, will generate a package of four, 8-bit words: an 8-bit data word, a 16-bit address word and an 8-bit housekeeping word. The latter word will determine the priority of its message (VCA data taking priority over EQ or routing words, for instance), and its order in the cyclic routine.

Hence the source of the message can be determineed so that similar control units, such as equalization, can collect data relative to their partner. Transfer rates are of the order of 64kbytes-125kbytes per second.

At the rack end, similar talker/-listeners collect data relative to the purpose of the listener section of the card and to provide overload data, error and fault information for diagnostic and testing purposes.

The "impersonal" nature of the design ensures the compatibility of modules in any slot in the rack; the data on the bus and the hardware coding of the card slot determine the task that the talker listener will carry out.

The database of console snapshots is stored on battery-supported RAM, or a 5¼-inch double-sided diskette holding about 32 snapshots. Scratchpad data, such as information regarding the recording session patchcord settings, microphone distribution on the set, etc., can also be stored to floppy disk.

All matrix and other change over switches are based on the same CMOS low-noise/low-distortion analog switch capable of full signal handling to ±22dBu. Where a build-up of potential crosstalk occurs, such as in a 128-way matrix, further "group" isolation effectively reduces crosstalk within specification limits.

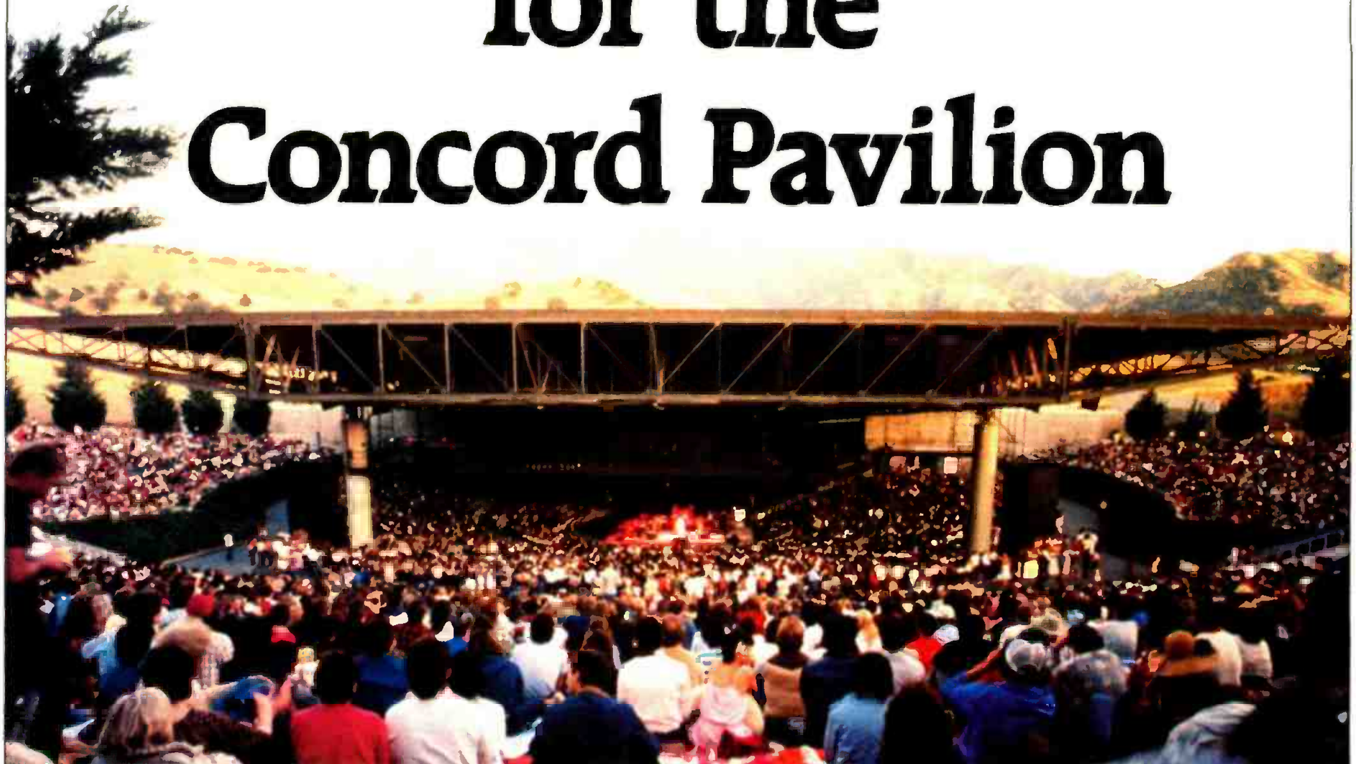
Variable-gain circuits employ VCAs controlled by 8-bit (256-step) digital data. Frequency circuits are all based on the same state-variable filter circuit employing twin VCAs to control frequency with 8-bit (256-step) resolution, except for filters that provide eight discrete settings in both HF and LF bands.

Electronic balancing is used for mic and line-level inputs, and many local outputs that will provide up to ±22dBu output level. Principal outputs are transformer-isolated and capable of ±22dBu output level.

A practical reality

To date, we have delivered an 88-input version of the assignable design to Thames Television, London for installation in Studio 2; consoles for Studio 1 and 3 are scheduled to follow. In addition, an 112-channel assignable console has been installed in a BBC Television Master Sound Control Vehicle, and which includes assignable dynamics and an assignable "wild" control in every channel. BBC Television has also ordered a 96-channel system for installation in Studio 4 at the Corporation's London Television Centre, with a similarly equipped console anticipated for the Centre's Studio 3. R·E/P

Sound System Design for the Concord Pavilion



By Mark Smith

This unique amphitheater in Northern California features a variety of sound reinforcement systems designed to accommodate a wide range of symphonic, rock and operatic performances.

Nestled in the foothills of Contra Costa county, in Concord, CA, lies an open-air arena and grassy lawns disguised as a symphony hall without walls. Known as the Concord Pavilion, the venue's specialty is the reproduction of concert hall acoustics in the open air.

The Pavilion was founded in 1973 by the Concord Performing Arts Center Authority, a coalition of the City of Concord and its Mount Diablo Unified School District. Financed by a bond issue and other donations, the 35-acre, $4.5 million facility officially opened in 1975. It has since been used as a venue for just about everything from symphonies to rock concerts, theater and spoken-word engagements. Indeed, the architectural and electro-acoustical designs implemented in the facility allow it to be used for virtually any type of performance.

Designed by the architectural firm of Frank O. Gehry and Associates and acoustician Christopher Jaffe, of Jaffe and Associates Acoustics, the Pavilion features a rear wall and two forward columns that support a steel roof projecting over a 3,500-seat amphitheater (Figure 1).

A surrounding tri-sectional 63,000 square foot lawn area is located outside of the overhanging roof, adding an additional 5,000 seats to the 270° listening radius. In typical amphitheater fashion, there are no side or front walls.

The design features two electro-acoustical systems that replace the "missing" architecture. An Assisted Resonance (AR) system is one of only four installed in the United States and 11 around the world. The Electronic Reflected Energy System (ERES) is a Jaffe design, and one of some 25 in the world. Chuck McGregor, Jaffe's consultant to the Pavilion since 1984, implemented the ERES system, while the AR system was installed during the venue's original development.

These ambience-enhancing arrays form part of a low-level system. There are, however, two systems in place: one for low-level applications, and one for high-level employment. Typically, the low-level system is used to supplement stage dynamics and so emulate an enclosed performance of symphonic sound. The high-level system reinforces the volume requirements of rock-type acts.

Mark Smith is a San Rafael-based free-lance writer, sound designer and engineer.

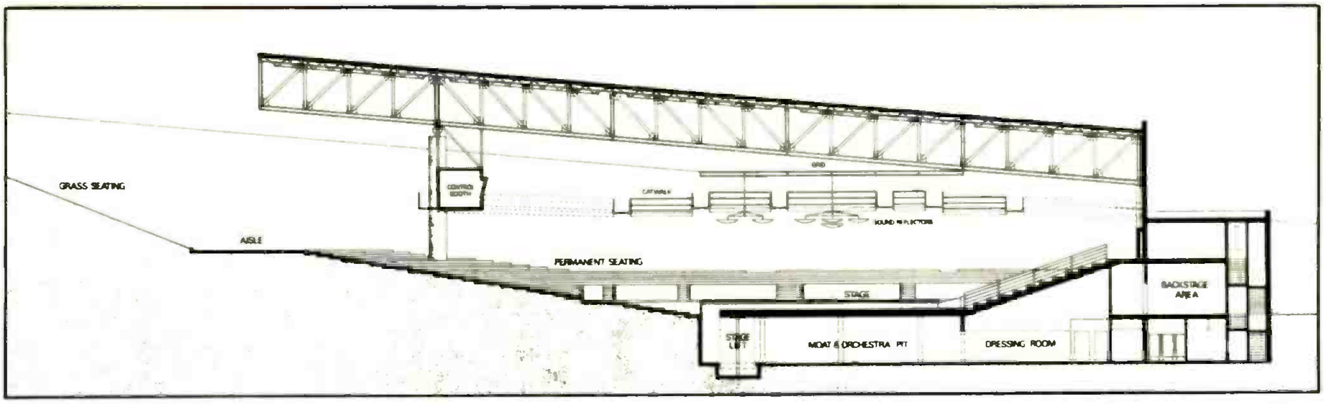


Figure 1. Section of Pavilion looking east, showing the extension of the roof area over the permanent seating areas, plus the position of sound reflectors mounted over the stage.

Both systems are piloted by the facility's house engineer, Jon Sagar.

Amphitheater structure

Centered under the steel roof is a performance area consisting of an all-wood sprung stage, with a ½-inch pine top. A curved and elevated acoustic moat serves as a stage apron and orchestra pit, as well as a micing area. Large, cylindrical fiberglass-filled drums descend by cables from the underside of the stage floor to suppress stage rumblings and first reflections that roll under the stage and through the open moat area.

Above the seating area is 144,000 square feet of reflective surface, 92,000 feet of which comprise steel beams and bracing. Some three quarters of the corrugated reflective surface on the underside of the roof is covered by 22,000 square feet of 6-inch R-16 fiberglass, thereby reducing the reverb time to approximately two seconds.

Directly above the first ring of seats, an inner catwalk circles a support area for the orchestra shell. This support area holds two stacks of three low-frequency Electronic Acoustical Rentals (EAR) subwoofer cabinets for the high-level system; the AR system speaker panels; the flying microphone used with the ERES system; and a hanging configuration of large saucer-like sound reflectors that help disperse sound outward by directing it into the forestage area.

The first (inner) catwalk (Figure 2) serves as an anchor for a series of 14 Bozak CM-209-16CH speaker columns supporting the low-level system. Four columns are concentrated behind the stage seating area, while the remainder are grouped with 10 custom-designed EAR Cubes that serve the high-level system. Both arrays cover the first seven seating rows that lay between the stage and the first concrete wall. This inner catwalk also acts as a support for five EAR 12′x3′ hanging speaker baskets, one per seating section for the high-level systems.

A second (outer) catwalk (Figure 3) is positioned in front of the second seating section and the first concrete wall, directly over the last row of seats in the first seating section. From this catwalk, a series of 15 Spectra Sonics model 3000 studio monitors that serve the low-level systems are suspended. Eight customer EAR 75° horns interwoven with the model 3000 cabinets support the high-level system over the facility's extended seating area. Also suspended from the outer catwalk are the resonator cones and microphones for the AR system.

Located at the front stage right pillar, near the rear of the seating area, is the house mix position, which connects to the stage via a 40-pair snake (Figure 4). Two isolated splits are available for the main house console and the monitor console, plus a third split for remote recording or broadcast. Six lines connect to the inner and outer catwalks, and the ERES flying mic. Ten additional backfeeds can be used as returns or sends to amps located in the overhead sound booth for the ERES and AR systems. Also, there are three available intercom channels.

Directly in line with the house-mix position, front center of the seating area, is an overhead south booth. This area houses the electronics for the ERES and AR systems, including the amplifiers and crossovers for the rest of the fixed reinforcement systems. Also located here are four Cetec Vega wireless receivers, a video monitor system (for viewing the storage area), production support equipment and a currently unused Spectra Sonics mixing console.

Positioned in equidistant circular patterns from the stage are two walls of ERES speakers and a subring of Spectra Sonics model 300 cabinets. The first wall consists of 16 ERES cabinets (Jaffe-redesigned Bozak CM-109-23s) circling the approximate center of the second seating section (Figure 5). The subring of six 3000s sits in line with the rearmost portion of the first wall of ERES speakers. The second ERES wall, comprising eight Jaffe-redesigned Bozak cabinets, sits some two-thirds of the way back over

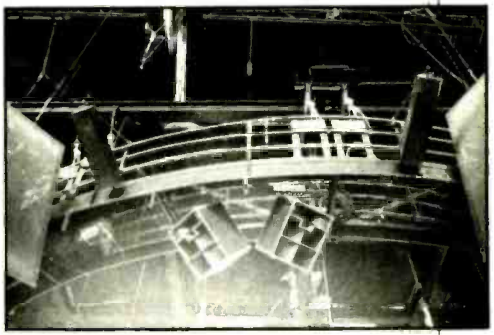


Figure 2. The first or inner catwalk supports the Bozak CM-209-16CH speaker columns. EAR Cubes (center) and EAR hanging speaker baskets (left and right).

Figure 3. The second or outer catwalk supports the Spectra Sonics model 3000 cabinets, EAR horns and the AR microphone arrays. The circular unit shown lower left is for low-frequency pickup, while the cylindrical device next to it is tuned for midrange pickup.

Figure 4. View from the left-rear area of the seating area, showing the house mix position.

this same seating area. The last row of reserved seating area lay 90 feet from the stage.

To provide sound coverage for the lawn area, 10 speaker "baskets" are suspended at the edge of the outer roof area in a circular pattern from platforms on aluminum brackets (Figure 6). Three baskets are positioned for each of the two side lawns and four are for the rear lawn. The baskets support both the high- and low-level systems that cover the most distant lawn areas located some 300 feet from the stage.

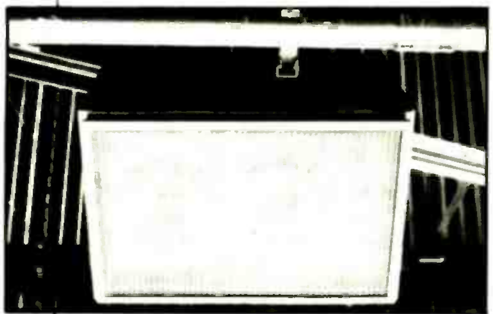


Figure 5. The ERES consists of two rings of Bozak CM-109-23 cabinets.

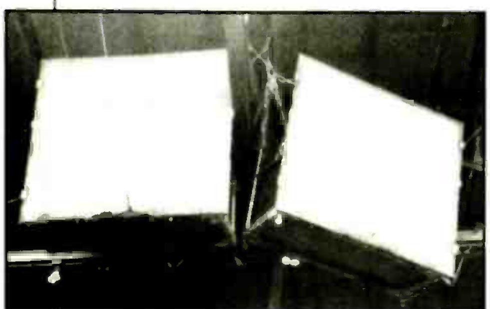


Figure 6. Lawn baskets suspended from the outer edges of the roof provide sound coverage for the distant audience members.

Figure 7 details the complete sound system currently in use at the Concord Pavilion.

Low-level reinforcement system

The low-level system consists of 90 AF speakers, 96 ERES speaker components, 224 Bozak column components (inner catwalk and behind the stage seating area), 63 Spectra Sonics model 3000 components (outer catwalk and booth ring) and 150 lawn speaker components. Total maximum power available from the amps connected to the speakers is a quoted 20,300W.

The AR system is powered by 90 AIRO-designed, 60W amps. The ERES system receives a maximum of 1,440W via 12 Spectra Sonics 120W bridged amps. The Bozak ring is powered by 14 Spectra Sonics 120W amps for a total of 1,680W. The Spectra Sonics model 3000s connect to a variety of Spectra Sonics 120W and 30W amps via active 3-way crossovers.

The model 300 monitors receive a total of 3,780W broken down to 120W bridged pairs feeding the low-end, and single cards of 30W each for the mics and highs.

The lawn system receives 8,000W from 20 Crown DC-300As, with 200W each being divided between two low-frequency, a midrange and a high-frequency band.

For mixing symphonic performances, a 20-input Soundcraft series 500 board at the house mix position is normally fed with an omni Telefunken 250 for conductor center, including Neumann M49s for stage left and right, plus a series of AKG C-451s and C-452s, Shure SM-57s and 56s and Electro-Voice PL80s for spot micing. A 48x8 Soundcraft desk is also available, although this is used primarily with the high-level system.

Assisted resonance system

The AR system was developed in 1965 by P.H. Parkin, of the Building Research Station, England, to correct the acoustics of the Royal Festival Hall, London. In 1969, Acoustical Investigation and Research Organization (AIRO) was licensed to market the AR concept throughout the world. To date, 11 such commercial systems have been implemented, the first in the United States being at the Concord Pavilion.

The principal behind the AR concept involves modifying the acoustics of a giv-

Basic Principles of ERES and AR Systems

• The ***Electronic Reflected Energy System*** *enables the introduction of electronically generated "walls" and "surfaces" into evironments that are either devoid of them or are in need of improved acoustics. The system uses pre-amplifiers, amplifiers, speakers, signal processors and other components, as well as an ERES control unit, to simulate the reflections of sound from architectural surfaces.*

In setting up an ERES system, the facility's acoustics are carefully surveyed, architectural drawings analyzed and sound tests performed. After sound absorption coefficients for air and wall surfaces and time of travel through the air are determined, the ERES control unit is then programmed with differing coefficents of absorption and physical orientation.

Single or multiple microphones connect via a pre-amp to the ERES control unit, which controls arrival time by releasing the sound of a presentation at specific locations through the facility that match the calculations for optimum acoustics. Inside of the ERES control unit signal passes through a notch filter to a series of delays. Outputs from the delays then feed power amplifiers that energize speakers positioned to recreate reflective surfaces.

• ***Assisted Resonance*** *is an electronic technique that makes "dry" or non-reverberant environments sound richer and fuller. Tests have shown that lower frequencies up to about 1kHz are responsible for the "warmth" and "resonance" that most listeners seek in a concert hall. By sustaining the volume of these lower frequencies in the areas of the hall where they are most subject to attenuation through absorption, AR duplicates the sound of concert-hall acoustics.*

Microphones and loudspeakers are installed at the boundaries of the hall, away from the on-stage performers. The microphones pick up the decay of the reverberant sound field, send their output to the amplifiers which, in turn, drive the dedicated AR speakers, thereby reinforcing the sound field. As each microphone senses a fluctuation in its specific frequency range, its associated amplifier and loudspeaker react instantaneously, adding sound level as required.

In a typical application, a mic housed in a tuned resonator delivers its output via a pre-amp to a phase shifter, which ensures that the system is in perfect phase and that any phase drift results in a decrease in gain rather than movement toward uncontrolled feedback. The signal then connects to a filter stage, which includes variable gain, Q and center frequency, followed by a fixed attentuator. The signal is then amplified and sent to the corresponding loudspeaker.

In the most current installations, a microcomputer controls and adjusts the attenuation and memorizes the system configuration.

The AR system is fine tuned using a loop break jack connected between the attentuator and the amplifier stage. By inserting a test signal (an artificially generated tone of a specific channel-related frequency) into the amplifier, phase and magnitude can be compared with the resulting output from the pre-amplifier. The phase shifter can then be set to ensure in-phase return signal, and the attentuator adjusted to provide the required amount of positive feedback to achieve the desired reverberation time.

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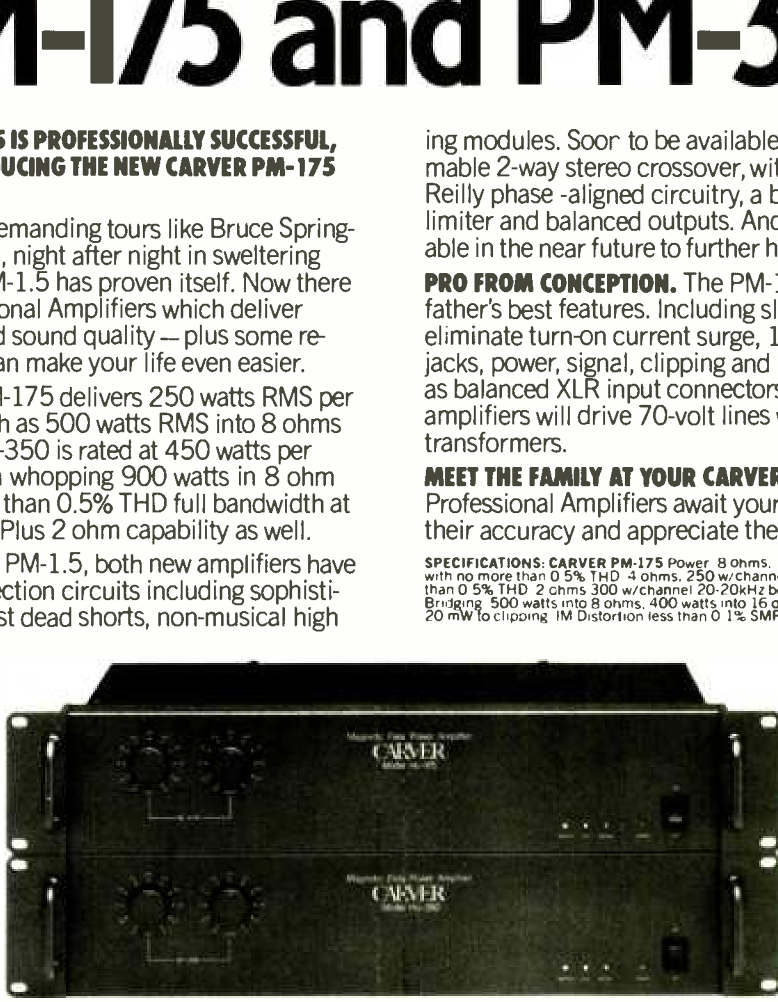

SPECIFICATIONS: CARVER PM-175 Power 8 ohms. 175 w/channel 20-20kHz both channels driven with no more than 0 5% THD 4 ohms. 250 w/channel 20-20kHz both channels driven with no more than 0 5% THD 2 ohms 300 w/channel 20-20kHz both channels driven with no more than 0 5% THD Bridging 500 watts into 8 ohms. 400 watts into 16 ohms THD-less than 0 5% at any power level from 20 mW to clipping IM Distortion less than 0 1% SMPTE. Frequency Bandwidth 5Hz-80kHz. Gain: 29 dB Input Sensitivity: 1.5 V rms. Damping: 200 at 1kHz Slew rate: 25V/ micro second. Noise Better than 115 dB below 175 watts. A-weighted. Inputs: Balanced to ground, XLR or TRS phone jacks. Input Impedance: 15k ohm each leg Compatible with 25V and 70V systems. 19"W x 3.5"H x11.56"D

SPECIFICATIONS: CARVER PM-350 Power: 8 ohms, 350 w/channel 20-20kHz both channels driven with no more than 0.5% THD 4 ohms, 450 w/channel 20-20kHz both channels driven with no more than 0 5% THD 2 ohms 450 w/channel 20-20kHz both channels driven with no more than 0.5% THD. Bridging: 900 watts into 8 ohms: 750 watts into 16 ohms. THD-less than 0 5% at any power level from 20 mW to clipping IM Distortion less than 0 1% SMPTE. Frequency Bandwidth 5Hz-80kHz. Gain: 31 dB. Input Sensitivity: 1.5 V rms. Damping: 200 at 1kHz Slew rate 25V/micro second Noise: Better than 115 dB below 350 watts. A-weighted. Inputs: Balanced to ground, XLR or TRS phone jacks. Input Impedance: 15k ohm each leg Compatible with 25V and 70V systems 19"W x 3 5"H x 11.56"D

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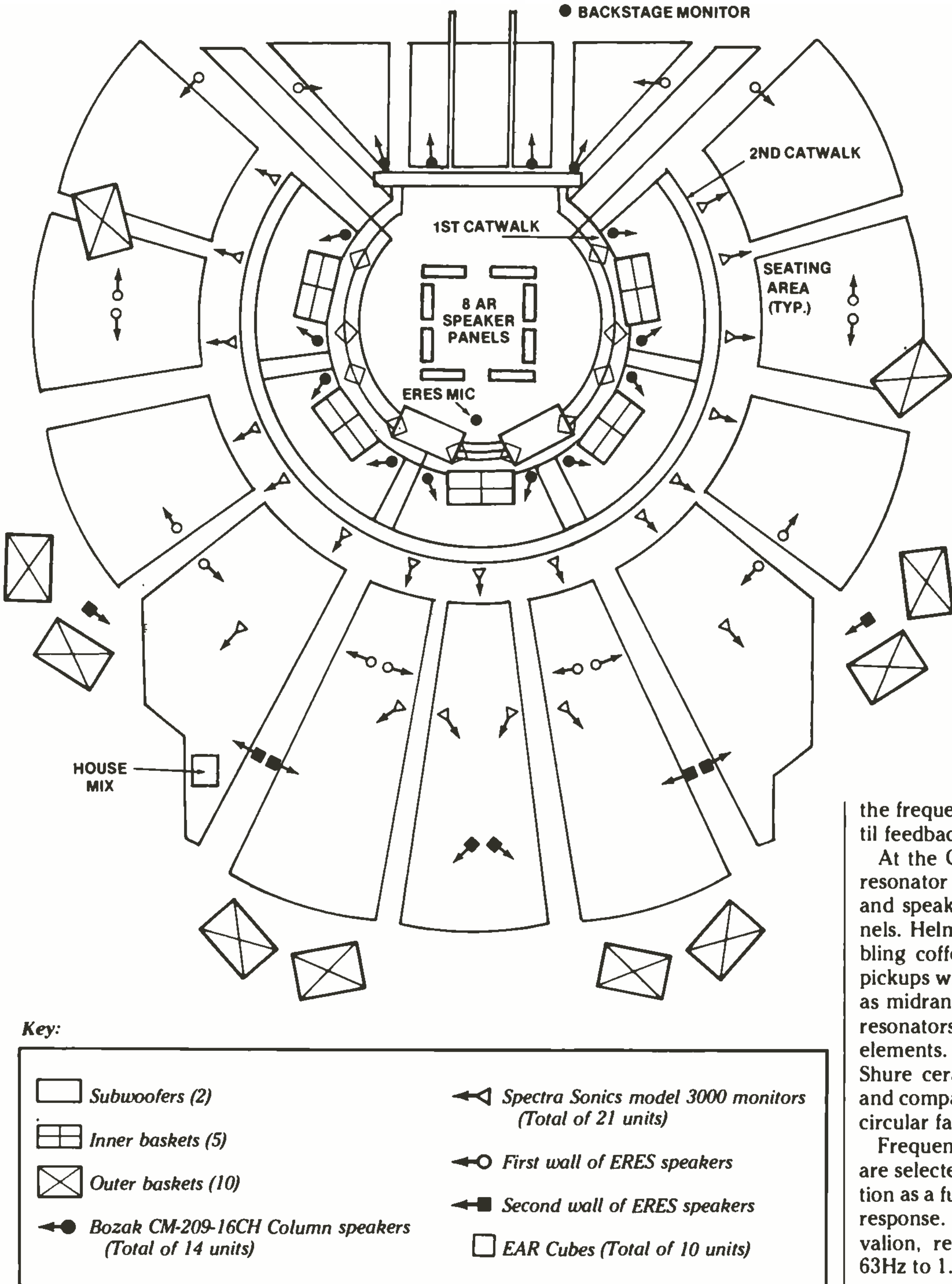

Figure 7. Detailed layout of loudspeaker components used for the various reinforcement systems.

en space, as opposed to modifying the characteristics of the sound being delivered to that space. In essence, controlled positive feedback via a bank of microphones and special loudspeakers is used to increase the RT_{60} from a value that would be suitable for speech and reinforced music, to the levels appropriate for opera, small instrumental groups, symphonic and choral presentations. Assisted resonance makes use of a comparatively large number of microphone and loudspeaker channels that are tuned to cover the frequency range of interest.

In practice, this means that a speaker is placed at a peak position in the acoustic pressure response of an area and a corresponding microphone located in a specially designed resonator at another peak. The electronics consist of a preamplifier, phase correction circuit, notch (bandpass) filtering, attenuation and amplification (Figure 8). By increasing the amplifier gain, reverberation time at the frequency of interest is increased until feedback occurs.

At the Concord Pavilion, a total of 90 resonator mics, pre-amps, power amps and speakers make up the 90 AR channels. Helmholtz resonator cones (resembling coffee cans) act as low-frequency pickups while thin parallel tubes function as midrange conductors. The Helmholtz resonators feature Shure dynamic mic elements. While the parallel tubes carry Shure ceramic cartridges. The AR mics and companion cones are suspended in a circular fashion from the inner catwalk.

Frequency intervals for the AR system are selected by using a logarithmic equation as a function of the ear's exponential response. In the case of the Concord Pavilion, resonators cover a range from 63Hz to 1.114KHz, with channels spaced at approximately 4% differentiation at lower frequencies and 2% differentiation at higher frequencies. Lower frequencies of 125Hz to 250Hz have proved to contribute most to the sensation of resonance.

Unlike some AR systems which are computer controlled and provide automatic operation, the Pavilion's manual system provides selectable "high," "low" and "off" settings per channel. Typically, sound is divided into frequencies through the mechanical resonators, as a function of their volume and construction, and then processed to the point of feedback. Attenuation is provided via a series of resistors that prevent feedback from occur-

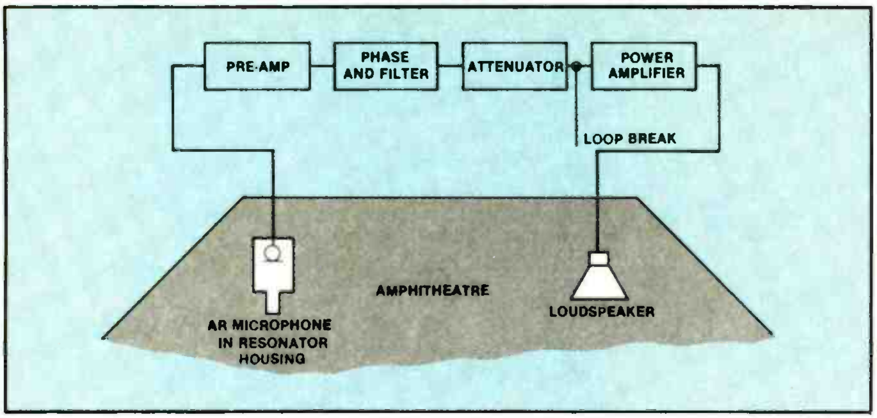

***Figure 8.** Block diagram of a typical Assisted Resonance channel.*

ring by reducing system gain below the feedback threshold.

A high/low selector allows the system operator to select a decibel level that is further tailored to the energy level being produced on stage. The high setting is typically 3dB-4dB higher than the low setting. A switch located at the mix position allows the system to be switched from high to low, or turned off.

Driving the AR system is the package of 90 AIRO-designed, 60W amps, whose true output usually hovers at around 18W-20W when driven at designed levels; feedback howl is usually introduced at around 32W. Underrated amps are used in order to maintain headroom.

The AR speakers comprise 12-inch Jensen model LMI-122s mounted four across and three high on eight infinite baffles made of ¾-inch plywood.

Located above the stage in the center of the first catwalk, this ensures the speakers with inconspicuous, centralized sound radiation.

A randomized frequency dispersion configuration, in conjunction with a patterned resonator-cone-to-speaker placement layout, ensures that mics and resonators are located as far away as possible from their corresponding speakers, and that speakers emitting adjacent frequency bands are not placed next to one another. Speakers are thus located on opposing panels in an ascending frequency progression. This predetermined placement ensures a good distribution of frequencies, as well as an acoustic mode that achieves a natural effect while avoiding a noticeable feedback loop.

The system is intended to be automatic; it needs only to be turned on and run, with some attention being paid to individual channel gain levels or system-wide high/low levels. Only occasionally does the system have to be turned off when a performance crescendo forces the system to take off, thereby unmasking its presence.

The ERES spatial system is designed to enhance an acoustic environment by electronically introducing "walls," "ceilings" and other reflective surfaces through a system of time delays. The objective behind the system is to tune the arrival of the low, mid and high frequencies of a performance, thereby simulating the presence of first sound reflections.

In implementing ERES, a facility's sound behavior is analyzed in terms of sound coefficients for air and wall surfaces, and audio travel paths. The system is then programmed to control sound arrival time by releasing the sound energy at calculated speaker locations.

By calculating reflecting patterns for various frequencies throughout the facility, ERES can be further tailored to balance localized sound discrepancies. Via matrix routing, the characteristics of a sound source that might be lost or muddled because of cross-stage frequency blending can be reintroduced locally, thereby bringing the strengths and weaknesses of a certain location into balance.

Typically, ERES microphones are placed approximately 20 feet from the central sound source. In the case of the Concord Pavilion, an AKG C451 is suspended from the first catwalk, with the mic stand pointed at the conductor's head. Output from the mic is then fed through a pre-amp, a control system, a notch filter, and then through a series of delay lines to the ERES speakers.

At the time of writing, the Pavilion supports two delay times of 72ms and 92ms, 96ms or 100ms, depending upon the performance. By the end of its 1986 season, the venue was looking to support four delay times concurrently. The delayed signals are sent to the two walls of Bozak cabinets, and the system is usually driven 6dB-8dB below the main system in order to achieve transparency (Figure 9).

ERES speakers are mounted in pairs driven by a single Spectra Sonics amp. An additional ERES feed is sent to the Bozak ring on the first catwalk plus the Spectra Sonic model 3000 rings. Currently, the direct send is to the Bozak columns located on the first catwalk, and via 72ms of delay to the Spectra Sonics cabinets on the second catwalk and those at the booth (in line with the first

Speaker Components Used at the Concord Pavilion

- ***AR System:*** *90, 12-inch Jensen LMI-122 drivers.*

- ***ERES System:*** *24 Bozak CM-109-23 cabinets, each housing two, 8-inch Bozak CM-109 and two, 2-inch Bozak CM-200 drivers.*

- ***Low Level System:*** *14 Bozak CM-209-16CH columns, each housing four, 6-inch Bozak CM-209-1 cones with midrange double-magnet speakers, and 12, 2-inch Bozak CM-200 drivers; plus 21 Spectra Sonics model 3000 studio monitors, each cabinet housing a 15-inch JBL K-130, a JBL 2307 conical horn on a JBL 2470 driver and an Electro-Voice T350 horn.*

- ***High-level system:*** 6 subwoofer cabinets *housing two, 18-inch Cerwin Vega drivers on custom-designed horns;* to EAR Cube cabinets housing a *15-inch, a 2-inch and a 1-inch JBL driver; EAR custom-designed, 75° horns with JBL 2445 drivers; five inner hanging baskets with four columns per basket (the columns in the three center baskets house two JBL 15-inch drivers, two JBL 2482s on single horns and a TAD 2-inch driver, while the columns in the two outermost baskets house a single JBL 15-inch driver and a 2-inch Emilar driver); 10 lawn speaker baskets each housing six, 15-inch Electro-Voice EV-15Ls plus four 12-inch JBL K-120 (two baskets) or four JBL 2202 drivers (eight baskets), a custom-built FM VT horn on a JBL 2482 driver and four JBL 2307 conical horns on JBL 2420 drivers.*

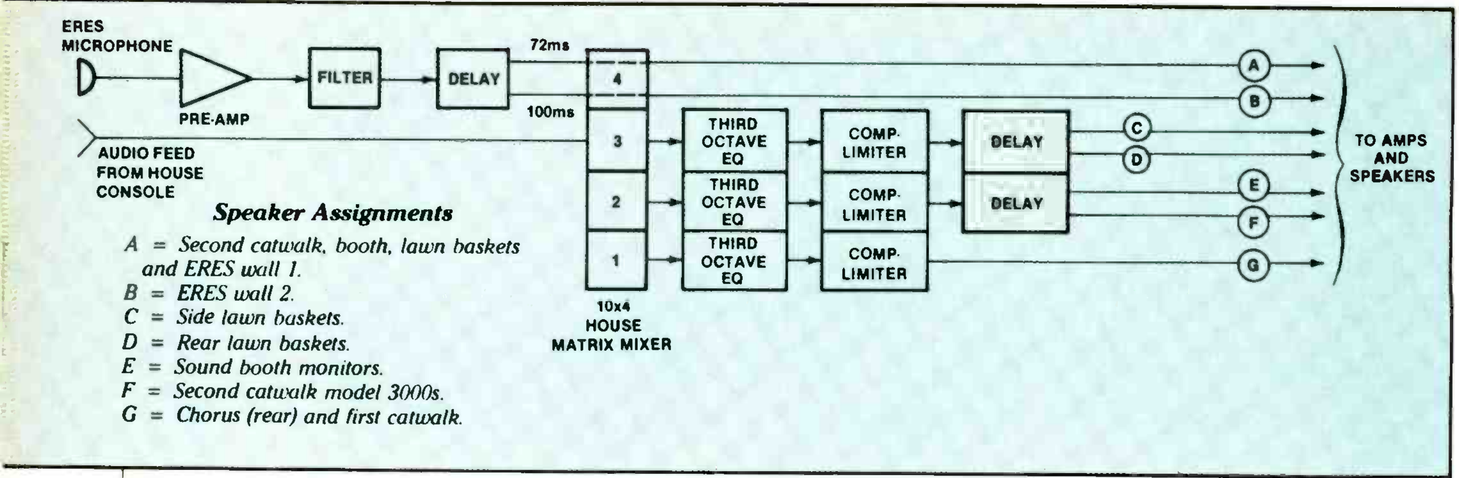

Figure 9. Block diagram of low-level and ERES systems.

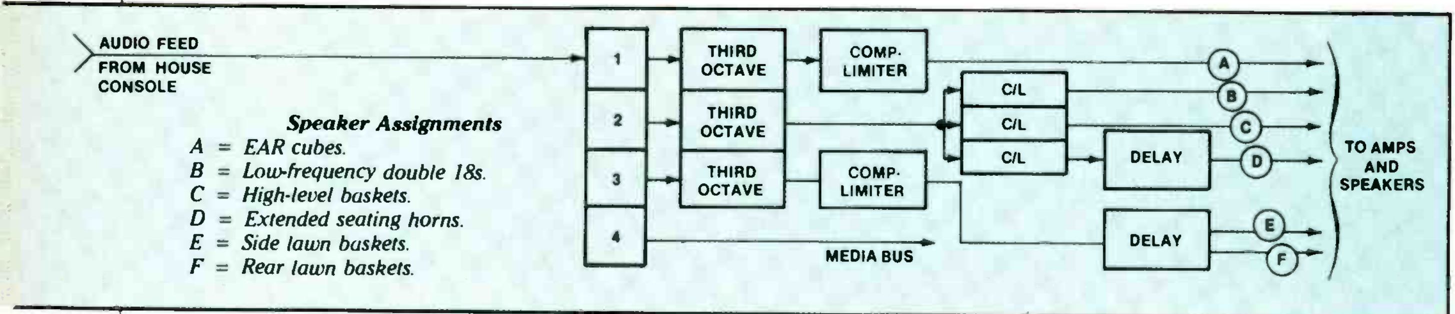

Figure 10. Block diagram of high-level sound reinforcement system at the Concord Pavilion.

wall of ERES). Also included are 10 lawn baskets and the first wall of ERES speakers. The variable second delay time is used in the feed to the second wall of ERES speakers.

The proposed configuration of four delay times will feed direct signal to the first catwalk, and ascending delay times to the second catwalk (Spectra Sonics cabinets), the booth speakers (Spectra Sonics cabinets), the first wall of ERES and the second wall of ERES, in that same delayed order.

High-level system

The high-level system (Figure 10) is predominantly leased from sound contractor Electronic Acoustical Rentals (EAR). The 22,000W system features 120 speakers incorporated in the central subwoofer cabinets, five central, custom designed hanging baskets, the Cubes, and a group of eight, 75° horns, in addition to the 10 lawn baskets. Touring sound companies commonly use the baskets in tandem with their own speaker setups, which can be mounted on pre-hung rigging points positioned between the five high-level baskets.

A customized 48x8 Soundcraft console, built originally for TFA Electrosound, is made available through EAR for bands travelling without their own board. (Optionally, a Yamaha PM3000 console is also available if the custom Soundcraft is being used elsewhere.)

As mentioned previously, the Cubes serve to cover the front seating area, while the horns help to cover the facility's extended seating area. Because the Pavilion is located directly above a residential area, sound bleed out of the venue needs to be controlled. To minimize sound bouncing off of the concrete wall above the side lawn area, and reflections rolling down the back sides of the facility, speakers and horns are angled downward.

Live performance review

A visit last year by the New York Philharmonic Orchestra served as a suitable subject for reviewing the low-level system. Twenty-six microphones were used on the orchestra, including one for the ERES system. During the performance, sound came from the model 3000s, lawn, Bozak, AR and ERES systems.

Despite the lack of a formal sound check, by the end of the first piece, house engineer Jon Sagar had the mix well underway. At first, the sound near the front of the lawn area seemed to be coming from up above, but by the end of the second piece, it appeared to be coming from the stage area. Sagar pointed out after the show that the AR system got out of control at one point during the show, but was not evident to me. Indeed, operation of the ERES and AR systems appeared totally transparent.

The sound system reproduced the performance well, with every instrument clearly audible. Dynamic response was not lost by the reinforcement systems, nor distorted by the steel super-structure or open-air design. Sound reproduction was of concert-hall quality, an observation shared by NYP guest conductor Leonard Bernstein.

For a performance review of the high-level system, a recent Moody Blues concert proved ideal. The system was driven by a Clair Bros. touring package with system engineer Gene Clair and the Pavilion's Jon Sagar working to combine the Clair add-on speaker cluster with the house systems. Clair Bros. provided four 12-foot bumper bars with three S-4 cabinets per bar.

Clarity of sound reproduction, as well as directional ambiance from the stage, were preserved during the concert. The high-level system proved to be surprisingly clean-sounding, even at the farthest point from the stage. There was no muddling of sound, and the transient peaks of synthesizers were not lost in the mix. Vocal range reproduction was superb. R·E/P

All photography courtesy of the Concord Pavilion.

Understanding Circuit Principles

Operational Amplifiers: Who's Who and How Come?

By Terry Pennington and Larry Winter

In this first part of a series written for recording and production engineers, we look at the various species of op-amps and how they are used in a variety of sound equipment.

The operational amplifier is the most common integrated circuit found throughout all signal routing, processing and recording equipment in the studio. There are numerous op-amp types and makes. Each has been designed or refined to meet the requirements of certain circuit applications, and each specific application will usually require different op-amp performance characteristics.

The question is, what should go where, and why? This article will outline basic op-amp performance characteristics, and discuss some specific op-amp models and where in the studio and stage environment they are best used. Such information should allow the audio engineer to realistically evaluate possible equipment updates, new equipment purchases and a variety of rumors circulating as to which is this week's hottest "super op-amp."

Circuit basics

Simply defined, an op-amp is a ready-made circuit that performs certain mathematical operations between its *two* inputs and output. Yes, there are two inputs on every op-amp: a positive and a negative input, otherwise known together as a differential input.

The math that each op-amp performs is to subtract one input from the other, then multiply the answer by some amount and deliver it to the output. The amount of multiplication, or "gain," is determined by how the inputs and output are connected up with surrounding circuitry. Let's start with the inputs and work forward.

The two inputs are labeled positive and negative, according to how they affect the output: A positive voltage on the positive input causes the output to go positive; a positive input voltage on the negative input causes the output to go negative. It stands to reason then that if the same voltage is applied to both positive and negative inputs, the result at the output is zero (it subtracted one input from the other).

A balanced or differential input on a mixing console takes advantage of this operation to amplify a signal while cancelling out any hum or radio interference induced into the connecting cables between equipment.

Here's how it works: A balanced cable contains a ground reference on the shield, and then a positive signal *and* a negative or inverted signal on two wires that are twisted around one another inside the shield. This balanced signal is connected to the differential on amp input, which subtracts the negative signal from the positive one to yield a positive signal output.

Becasue airborne interference induces the same voltage on both twisted wires, the result is that hum and RFI is subtracted at the inputs of the op-amp, and is virtually cancelled.

An op-amp's effectiveness in rejecting signals common to both inputs is called the "common mode rejection ratio," or CMRR, and is expressed in decibels. An op-amp with 60dB CMRR will reject interference much more effectively than one with 50dB CMRR. And, unless high tolerance resistors of 1% or better are used in the input stage, the CMRR can be severely degraded.

If an op-amp is to be used unbalanced or "single-ended" (positive signal and ground), then the ground reference must be connected to one of the two inputs. If the negative input is grounded and a signal applied to the positive input, then the op-amp is said to be in the non-inverting mode, because a positive input voltage yields a positive output voltage (Figure 1).

Conversely, if the positive input is grounded and a positive signal applied to the negative input, the output goes negative and the op-amp is in the inverting configuration (Figure 2.)

Each mode has its advantages in certain circuit applications; for example, summing circuits in bus networks almost always use inverting configurations to minimize gain interaction between buses.

Gain control

Now that the inputs are connected appropriately, how does one control the amount of gain or multiplication? First, you must find out how much gain the typical op-amp is capable of providing. This maximum gain is called "open loop

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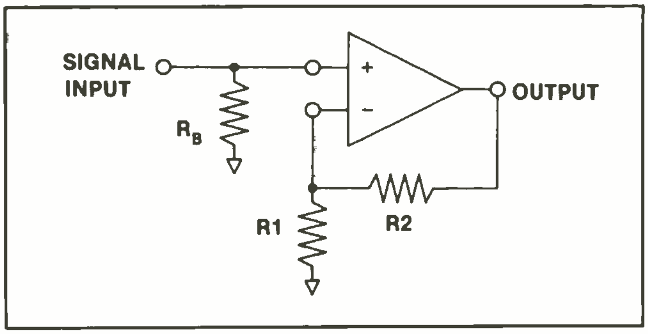

Figure 1. Basic schematic of an op-amp in a non-inverting configuration.

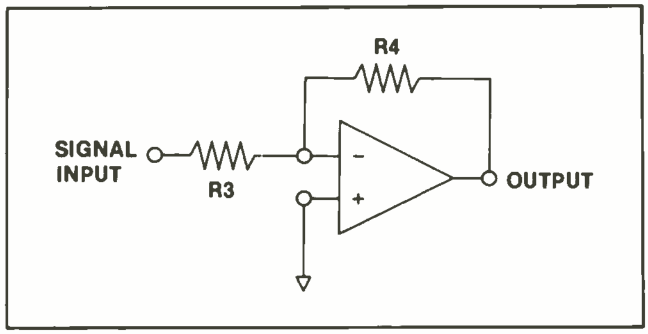

Figure 2. Basic schematic of an op-amp in an inverting configuration.

gain" in the op-amp specs, which is literally the gain of the IC without any type of feedback. Most modern op-amps have a tremendous amount of open loop gain, on the order of 100,000 or so, which is way too much for most audio applications.

The circuit gain of the op-amp is controlled by rerouting some portion of the output signal back into the differential input; the more output that is fed back to the input, the less is the actual gain. This is referred to as negative feedback, which controls gain and can also be used to reduce overall circuit distortion.

In a non-inverting mode, the amount of feedback is determined by the ratio of the two feedback resistors R1 and R2, and the total gain of the circuit is (1 + R2/R1). In the inverting mode, the gain is simply the ratio of R4/R3. As R1 or R3 is decreased, or R2 or R4 increased, the gain increases.

By adding capacitors in parallel or series with these feedback resistors, the amount of feedback changes as the frequency changes (capacitors have lower resistance at higher frequencies). This frequency-dependent type of gain is used in filters and equalizer circuits, and to reduce the high-frequency gain of the op-amp to prevent it from oscillating.

Component selection

Because most op-amps obey the above rules in the same manner, what are the differences between the various models and types available today? Almost all differences can be attributed to either noise or distortion characteristics. One op-amp may be very "fast"—able to reproduce high frequencies with very low distortion, and yet exhibit noise output that becomes a problem in audio circuits. Because there is no single "best" op-amp for all applications, one must first determine the purpose of each specific circuit, and then prioritize important specifications in order to choose the right op-amp for the job. Let's discuss the distortion category first.

Ideally, an op-amp will provide some desired gain without adding or subtracting anything from the signal—that is,

Table 1. Cross references to single op-amps. All specifications are for an 8-lead DIP package, at commercial temperature range.

Type	Manu-facturer	Absolute Maximum[1] Supplies	Noise[2] (nV/√Hz)	Slew (V/μs)	Stable[3] Gain	Current-[4] Typical/ Maximum	Minimum[5] Load	Comments
741	NSC	±18V	25.0	0.5	All	1.7/2.8	2kΩ	Reference only; do not use.
LF356	NSC	±18V	12.0	12.0	All	5.0/10	2kΩ	Drives 10.000pF. Large current drain. Generally, too noisy.
5534	SIG	±22V	3.5	6.0	22pF	4.0/8.0	600Ω	Audio reference standard. Large current drain. Poor supply ripple rejection.
OP27	PMI	±22V	3.2	2.8	All	3.3/5.7	2kΩ	Poor cost / performance ratio.
OP37	PMI	±22V	3.2	17.0	5.0	3.3/5.7	2kΩ	De-compensated OP27
MA362	AS	±26V	2.5	17.0	All	2.8/4.5	600Ω	THD=1PPM. Very expensive. Performance increases generally not worth the cost.
2134	SSM	±22V	2.8	7.0	22pF	4.0/6.5	600Ω	Best cost / performance ratio.
1028	LTC	±22V	0.9	15.0	All	7.6/10.5	600Ω	Lowest noise. Expensive. Very large current drain. May require additional compensation.

[1]Values exceeding absolute maximum power supply ratings cause all bets to be cancelled.
[2]Spot noise at 1kHz. Multiply by 141.4 (√20kHz) to calculate equivalent input noise (EIN).
[3]Fully compensated op-amps are unity-gain stable and marked "all"; de-compensated op-amps are unconditionally stable only when used in circuits with gains equal to, or greater than, that shown; un-compensated op-amps require capacitance indicated to be unity-gain stable.
[4]Total power-supply current in milliamps (mA).
[5]Minimum load resistance (ohms) for +20dBu output swing at rated slew rate and THD. Op-amp will drive all loads greater than, or equal to, this value (load capacitance <100pF). Op-amp will drive lower loads at reduced output swing, slew rate and THD. Typically, all op-amps will drive 600Ω to +19dBu at one-half slew rate, with THD <0.1%.

Authors' Note: The data presented in Tables 1, 2 and 3 was compiled by Dennis Bohn, vice president of research and development at Rane Corporation.

with no distortion. In actuality, several circumstances can cause an op-amp to become unfaithful. For one thing, speed: If the response time of the op-amp is too slow for higher frequencies, the output lags or "slews" and distortion is the result.

For high-frequency circuits, such as bias oscillators in tape machines, an op-amp slew rate (how many volts per microsecond it can produce) must be higher than required for conventional audio applications. As a result, this spec becomes important in choosing the right model.

Because there has been considerable controversy over slew rate, transient distortion and the like, the question becomes: "How fast is 'fast enough' for high-quality studio applications?" Theoretically, an op-amp need only be fast enough to reproduce 20kHz at +20dBu, which requires about 1.4V per microsecond. In practice, it turns out that two to five times this amount, or 3V to 8V per microsecond, is more than enough slew rate to guarantee inaudible distortion levels.

Slew rate tests such as transient intermodulation distortion (TIM) and slew induced distortion (SID) tests have shown high-frequency distortion in op-amps with the above slew rates. Usually, this is due to the application of frequency components far outside the audio spectrum, and it is important to realize that, although this is true, it is not necessarily relevant. Op-amps with excessively high slew rates can tend to be unstable at high frequencies, and can be triggered into high-frequency oscillations by such things as cable capacitance, temperature changes or line-voltage variations.

Although the oscillation itself cannot be heard, its effect on overall distortion, op-amp heating and component failure can be an ongoing nightmare that is difficult to trace down. Furthermore, the overall slew rate of a piece of studio equipment may be limited by internal coupling capacitance, which cannot be improved by installing faster op-amps.

The important thing is to maintain a realistic perspective and keep in mind that a studio or sound system is comprised of many pieces of gear hooked together—it will only perform as fast as the *slowest* component. (It makes little sense to spend inordinate sums of money on a super-fast mic input stage when the summing bus or tape desk pre-amps can't possibly keep up.)

Another cause of distortion is lack of juice: If the op-amp cannot supply enough drive current into the receiving circuit, then it can clip or go into a protective "current-limit" mode, resulting in quite audible distortion. Thus, such op-amp models as the NE5532 or the newer SSM2134, are highly valued as output driver chips, simply because they can supply the high currents required by driving multiple equipment, low-impedance (600Ω) loads or long cable lengths at high signal levels.

Because these high-current line drivers use up a lot of power, updating a mixing console with a number of these chips can drag down a console's power supply such that it will deliver less output than before, due to power supply sag. Furthermore, these line drivers produce much more heat and must be properly cooled to maintain reliability and long life.

When considering any equipment modifications, be sure to check with the manufacturer regarding the above considerations.

We now move into the realm of ever-present noise problems. Every op-amp has noise; even resistors have noise. This random electron activity gets amplified by the op-amp and, depending on the internal configuration, some ICs have more noise than others.

When considering the input stage of a high-quality studio console, one must take great care to use very quiet op-amps, and to design the overall gain structure to keep final output noise to a minimum. One basic rule is to take as much gain right at the mic input stage as possible; if excessive gain is taken after this stage, updates with the most expensive low-noise chips will yield only marginal results. A better approach would be to restructure the gain stages by replacing a few much-less-expensive resistors.

Similarly, console designs with true input pads are likely to yield lower noise figures than those with separate, variable-gain trim stages. The latter method requires that the unprotected input stage have lower gain, to prevent possible input overland that cannot be alleviated by reducing the trim. A true input pad network, using high-tolerance resistors, allows maximum gain right at the input stage, while preserving the necessary headroom to prevent overload.

Op-amps do have noise specifications; unfortunately they're a little tricky to interpret. Technically, the noise rating of an IC is expressed in "nanovolts per root hertz" (not to be confused with "furlongs per fortnight"). To find out how much noise the op-amp will actually produce in a circuit, one proceeds to multiply the above nanovolt figure by the amount of gain of the circuit (up to 1,000 for mic pre-amps), and then multiply the resultant product by the square root of the bandwidth which is usually 20kHz for audio applications.

The result is estimated at only a few microvolts of actual noise output and, in spite of the staggering gain and bandwidth figures, where are a number of op-amps that feature exceptional noise performance. The dual NE5532 and single 5534 have earned a well-deserved reputation for low noise.

Some newer devices have appeared that deserve close attention for superior noise performance, including National's LM833, NEC's 4570/4574 series, and SSM's 2134. Tables 1, 2 and 3 list the "latest and greatest" audio op-amp ICs, their important specs and some general comments that should be useful in any decision making process.

If you are evaluating new equipment purchases, or are considering upgrading your present equipment with improved op-amps, these tables will help you to compare performance, compatibility and

Table 2. Cross reference to dual op-amps. All specifications are for an 8-lead DIP package, at commercial temperature range.

Type	Manu-facturer	Absolute Maximum[1] Supplies	Noise[2] (nV/√Hz)	Slew (V/μs)	Stable[3] Gain	Current[4] Typical/ Maximum	Minimum[5] Load	Comments
4558	RAY	± 18V	25.0	0.8	All	3.3/5.7	2kΩ	Reference only, do not use.
LF353	NSC	± 18V	16.0	13.0	All	3.6/6.5	2kΩ	Very Stable. Generally too noisy.
TL072	TI	± 18V	18.0	13.0	All	2.8/5.0	2kΩ	Generally too noisy.
5532	SIG	± 22V	5.0	9.0	All	8.0/16	600Ω	Audio reference standard. Large current drain. Poor supply ripple rejection.
OP227	PMI	± 22V	3.2	2.8	All	3.3/5.7	2kΩ	Low current drain. Poor cost/performance.
5102	HAR	± 20V	4.3	3.0	All	3.0/5.0	2kΩ	Maximum differential input voltage only ±7V makes part unuseable in many applications.
5112	HAR	± 20V	4.3	20.0	6.0	3.0/5.0	2kΩ	De-compensated 5102.
LM833	NSC	± 18V	4.5	7.0	All	5.0/8.0	2kΩ	Excellent cost/performance ratio. Some applications may require local de-coupling.
4570	NEC	± 18V	4.5	7.0	All	5.0/8.0	2kΩ	Second source for LM833. Very Stable
2043	RAY	± 18V	3.5	6.0	All	6.0/8.0	600Ω	Medium cost/performance ratio.

See Table 1 for relevant footnotes.

Table 3. Cross reference to quad op-amps. All specifications are for a 14-lead DIP package, at commercial temperature range.

Type	Manu-facturer	Absolute Maximum[1] Supplies	Noise[2] (nV/√Hz)	Slew (V/μs)	Stable[3] Gain	Current[4] Typical/ Maximum	Minimum[5] Load	Comments
LM324	NSC	± 16V	35.0	0.4	All	1.5/3.0	2kΩ	Reference only; do not use.
TL074	TI	± 18V	18.0	13.0	All	5.6/10	2kΩ	Reference only; generally too noisy.
5104	HAR	± 20V	4.3	3.0	All	5.0/6.5	2kΩ	Maximum differential input voltage only ±7V makes part unuseable in many applications.
5114	HAR	± 20V	4.3	20.0	6	5.0/6.5	2kΩ	De-compensated 5104.
4156	RAY	± 20V	9.0	1.6	All	5.0/7.0	2kΩ	Most cost-effective noise up-grade part.
4741	NEC	± 20V	9.0	1.6	All	5.0/7.0	2kΩ	Second-source for RC4156 (also by Harris).
MA374	AS	± 24V	8.0	15.0	All	6.8/10	2kΩ	Poor cost/performance ratio.
LM837	NSC	± 18V	4.5	10.0	All	10/15	600Ω	True quad 5532. Quietest. Some applications may require local supply decoupling. Large current drain.
4574	NEC	± 18V	5.0	6.0	All	8.5/12	2kΩ	Quad 4570. Excellent cost/performance ratio. Large current drain.

See Table 1 for relevant footnotes.

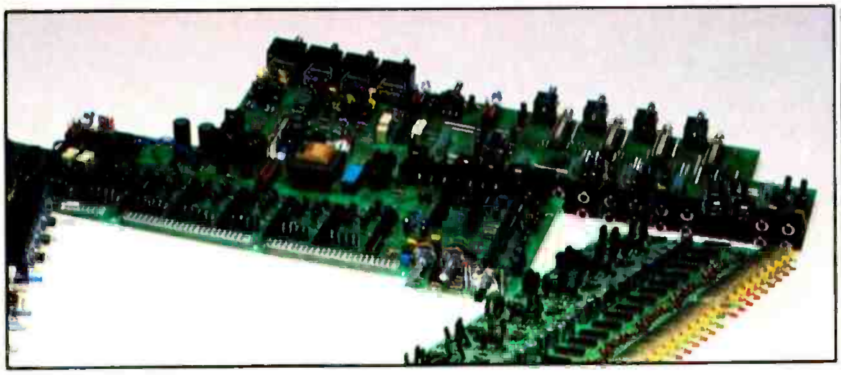

Photo concept by Larry Winter

relative cost factors of the most current designs.

The dual-FET-type op-amps (NSC's Bi-FETs, RCA's Bi-MOS, etc.) have been omitted from the tables because they are too noisy for high-performance audio applications.

If you are replacing pin-compatible op-amps, be prepared for some possible surprises. Some ICs might oscillate unless additional power supply bypass capacitors are installed, and it's nearly impossible to predict ahead of time. Also, some op-amps cannot be operated at unity gain, so heed the comments in the table carefully, and be sure you understand the circuit requirements at each IC location you plan to update.

System interface

A few general words of advice are in order here, concerning noise and equipment matching. Whether you operate a studio or a sound reinforcement system of any size, you will have several pieces of equipment hooked together in series: mixer, equalizer, compressor/limiter, crossover and the like. Each device may contain the latest op-amps and boast impressive noise specifications, but you can still wind up with excessive noise. Why?

Some people will say impedance mismatch—actually it's gain mismatch: where excessive gain is taken downstream in a system due to improper interaction between various level control settings.

Thanks to the op-amp, you don't have to worry about *impedance* matching of equipment. Most equipment can drive 600Ω loads and higher, and most input impedances are 100RΩ and up. It is not necessary to *match* the input impedance of another; one need only be concerned with *minimums*.

If a piece of gear is rated to drive 600Ω, this is a minimum; any input with greater than 600Ω will work perfectly. A safe rule is to be sure that the input impedance of the receiving unit be at least 10 times the actual output impedance of the driving unit.

Equipment loaded with too low an impedance can exhibit loss of headroom or frequency-response degradation; this is because lower impedances require more drive current from the op-amp, which can cause excessive distortion, premature clipping, or triggering of a protective current-limit mode. Noise problems, on the other hand, are usually caused by poor gain management throughout the sound system.

Gain matching

Aside from making sure that equipment is operating at the same nominal line level (−10dBv or +4dBu), a basic rule mentioned earlier applies to optimum noise management in studio and sound systems: Take as much gain as possible at the *beginning* of the signal path, keep the signal level as high as possible in each unit and avoid taking gain toward the end of the signal path. Keep mixing console levels as high as headroom will allow, from mic stages to sub-bus to output.

If you install a parametric equalizer into a channel-insert loop, keep the EQ gain at unity. Turning up the EQ gain will cause noise problems due to additional gain taken at the mixer summing nodes. If you use a compressor/limiter in concert or at mixdown, keep its output levels at maximum line level; compressing the signal way down and then bring ing it back up with equalizer, crossover or tape pre-amp gain will often result in excessive noise.

Keep equalizer, crossover and signal processor level controls at unity gain; maintain the highest possible signal level *all the way through the system*, attenuating it only at the tape machine amplifier inputs as necessary.

Check each piece of signal processing gear to find out whether it has input level control, output level control, or both. Set the control(s) to maintain the highest possible signal level *within* each unit. If you're having noise problems, chances are you've attenuated the signal level somewhere along the line, and then turned up the gain on some other unit downstream to get it back up again. This is gain mismatching.

Check every unit in the system. Avoid setting line-level controls above unity gain if at all possible. This way you will ensure the kind of performance that is expected from your state-of-the-art equipment and the many op-amps configurations that make these designs work so well.

In subsequent installments in this regular series of articles about circuit principles, we will consider a variety of relevant topics, including how audio recording and production engineers should view a facility as a "system," and how to develop adequate interface and grounding schemes for minimum noise and interference. R·E/P

Table 4. *Manufacturer's of op-amps listed in Tables 1, 2 and 3.*

Solid State Micro Technology 2076B Walsh Ave. Santa Clara, CA 95050 408-727-0917	*(SSM)*
Harris Semiconductor 2401 Palm Bay Road Palm Bay, FL 32905	*(HAR)*
PMI 1500 Space Park Drive P.O. Box 58020 Santa Clara, CA 408-727-9222	
Analog Systems P.O. Box 35879 Tucson, AZ 85740-5879 602-792-3202	*(AS)*
NEC Electronics, Inc. 401 Ellis St. P.O. Box 7241 Mountain View, CA 94039 415-960-6000	*(NEC)*
National Semiconductor Corporation 2900 Semiconductor Drive Santa Clara, CA 95051 408-721-5000	*(NSC)*
Raytheon Company 350 Ellis St. Mountain View, CA 94039-7016 415-966-7636	*(RAY)*
Signetics Corporation 811 E. Arques Ave. P.O. Box 3409 Sunnyvale, CA 94088-3409 408-739-7700	*(SIG)*
Texas Instruments P.O. Box 225012 Dallas, TX 75265 214-995-6611	*(TI)*
Linear Technology Corporation 1630 McCarthy Blvd. Milpitas, CA 95035 408-942-0810	*(LTC)*

Production Viewpoint:

Roy Thomas Baker

By Adrian Zarin

Although, by his own admission, RTB is a producer that often pushes technology to its ultimate limits, he has enjoyed an enviable production and engineering career working with some of the most innovative talents to emerge during the past two decades, including Queen, The Cars, Journey, Foreigner, Slade and, most recently, T'Pau.

The name Roy Thomas Baker has become synonymous with a peculiarly adventurous style of record production. More than anything else, his studio and recording techniques appear to be based on an unflagging desire to push audio equipment beyond its normally recognized limits. It's an approach that has helped Baker shape the sound of more than a few landmark groups, including Queen, The Cars, Journey and Foreigner.

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RTB's earliest experiments with techniques such as tape-based flanging can be traced back to the Sixties, a period when he was working as an engineer on sessions for The Who, The Rolling Stones, T-Rex, Frank Zappa and others.

It therefore came as a surprise to many when Baker took a sabbatical from the studio in 1983, and accepted an A&R position at Elektra Asylum. But since leaving that post in 1985, he has again taken up his old position behind the mixing console. He began by producing some tracks with Duran Duran guitarist Andy Taylor for the *American Anthem* film soundtrack. Recently, Baker has completed tracks with Slade for their forthcoming new album and a debut album release for the group T'Pau.

The T'Pau project brought Baker and the group to Royal Recorders, Lake

Geneva, WI. There, tracks were cut on a synchronized 3-machine format: two Studer A-800 Mark IIIs analog 24-tracks locked to a Mitsubishi X-850 PD-Format digital 32-track, with the studio's 80-input Solid State Logic SL4000-E console as main command center. By Roy Thomas Baker standards, it was a fairly modest project: only 80 tracks.

"Doesn't everybody use that many tracks?" he asks with mock incredulity. "When I did the Foreigner album, which was several years ago now, I had nine 24-track machines linked up. I assumed most people use that many."

Don't be fooled though. This sort of nonchalance is just one of several evasive tactics Baker is wont to try when pressed for technical details. Perhaps it's all part of his Charles Laughtonesque charm—jovial, avuncular, and terribly British.

But the hapless interviewer's first impulse is to be irritated. Ultimately though, you realize that a certain irreverence toward details is an integral element in Baker's creative approach. Irreverence and experimentation, you'll recall, are the twin pillars of RTB's chosen musical genre: something that was once referred to as "rock and roll."

Individuality in the studio

In actual point of fact, Baker has always taken a certain self-conscious delight in his "over-the-top" style of record production. And the use of what he likes to call "multi, multi, multitrack" has long been a key feature of that style. During the Seventies, he was one of the first customers of John Stevens' custom 40-track tape machines.

"In those days, you understand, linking [synchronizing] multitrack machines wasn't done. And when they started to do it, it was incredibly unreliable. So I began using the 40-track Stevens. It's a great-sounding machine, and I still use it.

"I don't use it as much as I used to, unfortunately, because the only person who can maintain it properly is John Stevens. And he's based out here in Burbank [CA]. If something goes wrong in the south of France where I might be working, it doesn't make much sense to fly someone all the way out from Burbank just to change a knob!"

Principal reason for the large number of tape tracks that Baker uses is to provide creative options. He frequently compares his craft to that of a painter. He has little use for pre-production, preferring to rely on spontaneity and instinct in the studio itself. To capture a good guitar sound, for example, he might start with 20 microphone channels, each representing a different option in terms of either ambient micing or close micing on the amp cabinet.

"I just put piles of mics in the studio," he confesses. "I don't always know what mics they are. For guitar, I use a lot of tube mics; a few FETS, but mainly tubes. Each mic is plugged into its own channel; and I've got a list in front of me. I just go through and pick the ones I think sound good."

Paintings and snapshots: analog vs. digital

On the recent T'Pau project, RTB's "multi, multitrack" methodology became a vehicle for exploring the relative merits of digital and analog recording

Roy Thomas Baker with a "tastefully" decorated Mitsubishi X-850 PD-format digital 32-track during sessions at Image Recording, Hollywood.

technology. Every track was recorded on X-850 digital, and also on one of the A-800 analog machines.

"We worked on analog and digital simultaneously," he says, "and we picked what was the best sound for any given track. For instance, I could get the best snare sound from analog—something that's nice and 'squelchy.' And then the best cymbal sounds from the digital multitrack. We always had the analog and digital machines linked up, right from the start of the session."

For the most part, however, Baker says that he found digital less than conducive to his unique style of working: "In most cases, there was a huge change in the sound with digital. It seemed to me that a lot of the 600Hz to 1kHz area got lost. Another big problem, I found, was bouncing tracks on digital. Say you were bouncing down your drum ambience tracks, and you had maybe four or six tracks that you wanted to mixdown to a stereo pair, adding whatever effects you wanted to add in the process.

"What I found was that the tracks came back out of phase—not the full 180° out of phase, but somewhere in-between. It was fully out of phase, that would be fine; you could just hit the phase reverse buttons on the desk and it would come out OK. But this was somewhere in between.

"Here we have a case of something we can do very easily on analog, but can't seem to do very well on digital.

"We worked on analog and digital simultaneously and we picked the best sound for any given track."

"And the worst thing," the producer continues, "is not being able to edit the way you can on analog. You see, I don't just edit to delete parts or move them around; I edit for *effect*. If you want a guitar chord to really bang in, for example, you edit out a little bit of tape right before it. Or if you want the chord to slide in, you add a little bit of tape. That's all part of what I mean when I say I'm 'painting a picture.'

"But digital machines, as I see it, are just for taking 'photographs.' If you've got a jazz band or a classical orchestra, and all the parts are already set and laid out, then you can use digital to take an accurate photo of what's going on. But if you're making a rock record, it's a different process entirely.

"What if you suddenly want backward echo or a backward snare drum? You can't just turn a digital tape over. Which means that digital is totally unusable for anyone who's got any artistic ideas."

Given these feelings, it's no surprise that Baker opted for analog multitracks (specifically, Studer A-800 Mark IIIs) to record basic tracks and many of the overdubs for the Slade album project. In every other respect, however, the producer's overt intention was to place the seminal proto-metal band in a "state-of-the-art" environment.

"They've normally recorded in small studios and worked 24-track," the producer says. "I think this is the first time the band has actually done a record using multi, multi, multitrack, large SSL boards and that sort of equipment. We're doing the record at Wessex Studios [London]. Lots of us, producers, friends and such, have always loved Slade. After all, where would Quiet Riot be without them? So a lot of us who were producers got together, and we all decided to do a couple of tracks each with Slade."

To capture the band's strident edge, Baker opted for a live-in-the studio approach to cutting basics.

"We went back and overdubbed a few things afterward, but everyone was playing live as it went down, including a rough vocal. To start with, we had the bass and guitar amps located off in the drum room, and the drums in the main studio. Then, for the overdubs, we brought the amps out into the studio again."

Still in the midst of recording the project at the time of our interview, Baker stretched out a basic plan for completing the project: "First we have to do some editing, which we'll do on analog. And if we do any extra tracks, which we probably will, we'll use the Mitsubishi X-850."

As part of Baker's program for "painting" a record, the photographic realism of digital is mainly useful for adding a few highlights, rather than building a solid base of colors.

"Digital doesn't take the level, which is another problem. The only way to get a decent sound out of an SSL desk, in my opinion, is to really *drive* it. And when you're doing that you have to know that your machine can take it, and also that the tape can take it.

"On the T'Pau project, the problem we were having with the digital recorder is that the tracks kept cutting out. If things go into overload, the machine just says [affecting a boorish accent]: 'Well, it shouldn't be that loud.' and just mutes."

Analog tape techniques

A great deal of popular music history has arisen out of a few, very special relationships between a given individual and a particular technological medium. Along with such elective affinities as Jimi Hendrix and the Marshall Stack, or Giorgio Moroder and the CV/gate sequencer, one could safely list Roy Thomas Baker and analog multitrack recording. Many of Baker's techniques stem directly from his work on the first four Queen albums, work that netted his first significant production credits and became the cornerstone of his subsequent career.

"I love to really *saturate* tape," he confesses. "On the Queen sessions, the only time the VU needle ever came off the end stop during the mix was the beginning and end of the song. As soon as the song began, the needle would just hit the black and stay there. I always work tape very, *very* hard. The big problem is that recording tape keeps getting better and better, so you have to work it harder and harder!"

And although the epithet "Mr. Compression" has often been applied to Baker, he insists that his characteristic sound is based more on his use of tape itself than on outboard signal-processing gear.

"I hardly ever use compression," he

"I love to really saturate tape. On the Queen sessions, the only time the VU needle ever came off the end stop during the mix was the beginning and end of the song."

*Pictured during a recent tracking session with the band T'Pau at Image Recording studios, Hollywood (*left-to-right*): second engineer Steve Krause, producer Roy Thomas Baker and facility co-owner Jon Van Nest.*

laughs. "That's the real joke. I'll compress individual instruments, and I use a bit of compression when we do the mastering; but I *never* compress the overall mix.

"All the supposed compression for which everyone calls me 'Mr. Compression' is basically just tape overload."

Track bouncing is no less important as a manipulative tool, Baker says. Nowadays, he will bounce tracks mainly as a means of condensing and narrowing down the options offered by his multi, multitrack approach. As the project nears mixdown, he will bounce tracks to reduce the need for auxiliary mixing boards to handle the 46+ tape tracks.

"If we bounce down from, let's say nine 24-track machines to three for the mix, then we're generally OK," he says. "But, if we have to bring in a fourth machine, then we have to bring in an extra mixer."

Back in the days of those early Queen sessions—before the Stevens 40-track machine came along, and 16-track was still the prevailing technology—track bouncing had even more of a creative role, the producer recalls.

"The backing vocals, for example, might have been a 3-part harmony, triple-tracked, which meant we had 12 tracks to bounce just for that one section. The biggest problem we had was wearing the tape out with so many passes. We were using the old 3M machines. Although we got a lovely sound from them, they would really stretch that tape out. In fact, tape was always up to full tension on the heads that there were just huge amounts of oxide coming off all the time. So we spent a *lot* of time making copies onto copies.

"All the supposed compression for which everyone calls me 'Mr. Compression' is basically just tape overlaod."

"By the time we were mixing, we were generally on to our third roll of tape; our third generation. The drums, by that time, sounded distorted and 'flabby.' But that signature became part of the sound. The 'Phil-Spector sound' also came from generation loss. It was funny to hear the drums getting flabbier and flabbier as we went on. It sounded great on the radio, however, and everyone said, 'Wow, how did you get that sound?' Basically it's just distortion, that's all."

Recording console technology

As noted already, Baker is no less hard on mixing consoles than he is on analog tape. Having had a role in developing the original Trident A-Range console—during his late-Sixties/early-Seventies tenure as an independent engineer based at London's Trident Studios—perhaps he's earned the right to push consoles to the limit.

"To me, mixing desks are like guitar amps. If you take a tube guitar amp and turn the level up to one, it sounds boring; you've got to turn it up to 10 to get the third and fifth harmonics from the tube. It's the same thing with desks—they just don't perform unles they're nice and 'squelchy.'

"And, just like guitar amps, every desk sounds different and each has a different use. I like the SSL for some things, but I'd also like to have a Trident and a Neve as well—all in the same room!

"It would be great to have 20 channels of Neve, 20 channels of Trident and all the other functions. That way, you can vary your sound. Even if I got a console manufacturer to copy the Neve EQ sound for me, it still wouldn't sound like a Neve.

"What I tend to do now is use the mixing board as one of the tools that enable me to achieve different sounds. Sometimes I bypass the console EQ entirely, and go straight from the mic pre-amp outputs into an Orban parametric equalizer, which really lets me screw up and destroy the sound! Then I feed the processed signal back into the board.

"For example, I prefer the acoustic guitar sound I get from an Orban to what I get with the console EQ. Or, if I'm micing an acoustic guitar that also has a pickup, I might use the console EQ for the electric [DI and/or amp cabinet] side of it, and parametric EQ for the acoustic.

"I always try to go everywhere with stacks of Orban equalizers, and stacks of Drawmer noise gates. The Drawmers are a very good musical tool, although now and again I like to use the old Valley People Kepex gates for certain things. There's something nice about an old Kepex [dropping his voice to a conspiratorial whisper]; I think it's the way it distorts!"

Baker's work on the four Andy Taylor songs destined for release on the *American Anthem* soundtrack illustrates his "mix-and-match" approach to recording technology. The project began at a venerable Hollywood studio, Crystal Sound, where basic tracks for the four songs were recorded.

"The ambience sound at Crystal there was very good, and the board [a vintage Crystalab model 2424] was good too. For the mix we went over to Larabee Sound, where they have an SSL [4000-E] and all the modern equipment. That way we had a combination of modern technology and the old acoustic sound. We took the best of both worlds, literally."

Future plans

Right now Baker is in the process of pooling the business experience he gleaned at Elektra with his years of studio expertise to start a company of his own, RTB Audio Visual. He's hoping to include a record label and video production company under his corporate umbrella. Originally, the producer planned to build his own recording studio as well, but the plan has since been scaled back.

"What I tend to do now is use the mixing board as one of the tools that enables me to achieve different sounds."

"What I'd like to do instead is to find a studio that's upgrading, and then get involved in the upgrade process. I have a lot of new ideas for studios that, I feel, are quite revolutionary. I was designing my own studio in England before I left—I was going to put one in my castle—and those ideas are still on paper. They have never been utilized, but they're all scientifically sound. It's just a matter of putting them in the right place." R·E/P

All photography by Elizabeth Annas/Photosensations.

Rear view of author's personal-use studio, showing control-room door (left) and equipment racks. Note the inward flare of the side walls.

Designing a Conical Bass-Horn Control Room

By Clifford A. Henricksen

Reproducing clean, detailed low-frequency information in a control room often involves several compromises. This personal-use design uses the room's side walls to form a conical bass horn, with some interesting results.

This is the story of the conception, design and construction of a personal-use recording studio control room and monitor system.

In fact, the monitor system forms an integral part of yielding an unusual sonic capability. Because my profession is acoustical (loudspeaker) engineering and my true passion is musical composing and recording, the system's design will be discussed as 2-headed philosophy.

In writing this article, I hope to stimulate new ideas for studios, present what should be a pretty interesting experience, and generate correspondence on this subject, both for my own education as well as a mutual trade out of ideas.

Clifford A. Henricksen is a group leader in loudspeaker engineering at Electro-Voice, where he is responsible for management and engineering direction of the Components Group developing low-frequency drivers, horns and compression drives. He was chairman of an AES working group for drafting the society's first-ever recommended practice for specifications of professional loudspeaker components and has been elected a fellow of the AES. He is also an active musician involved with live performance, studio recording and composition.

System concepts

Why did I ever take *on* this project, I pondered, as I first turned on the monitor system in my new, bare drywall, conical bass-horn control room. It sounded just awful, but at that point I had no choice but to bite the bullet and press on. I took on the project because I remember a sound, a sound you can't get in a normal room.

The origins go back to 1980, when I was working as an Altec speaker engineer by day, and a musician at other times. I was playing at a rhythm session at Westlake's newly opened "Beverly" studio in Los Angeles, and Allen Sides was the engineer hired for the date. When Allen found out where I worked we got into what has amounted to a semi-endless conversation about all this.

Allen, now owner of Ocean Way Studios, Hollywood, was at the time operating out of a garage/studio in Santa Monica and he invited me over to have a listen. Just above the console was a pair of double 15-inch RCA bass horns and other components to complete the system. When he turned on the tape machine, I was both delighted and amazed with the overall sound, but especially with the bass transient and detail. This was wide-open, clear, "water-pik" bass and I finally came to understand it.

Allen's room was pretty narrow, say nine feet across, which was pretty close to the combined width of the bass horns. At the back of the room, behind the tape machines, was what appeared to be a very effective bass trap; the LF wave went through you and didn't come back.

I said to myself, "This has got to be the best way to produce solid bass. The limit has to be actually sitting in the horn itself. That's it: make the room a horn and absorb all the bass at the rear end." Since then I've dreamed of doing this for myself.

My opportunity came when I relocated to Michigan, and moved into a spacious English cottage-style mansion with a giant basement. Many weeks were spent gutting the inside of a 12'x8' concrete-walled basement room, and I was ready to design my "dream" control room.

High quality monitoring

First of all, I think that any studio needs a high-output, low-noise, low-distortion monitor system for "detailing" and scrutinizing a mix in the same way that a diamond cutter uses a microscope.

Second, you need a very pleasant (but revealing) system that doesn't annoy you after hours of working on a mix. It's also a wonderful listening pleasure to be able to play clean masters on a monitor system that has effortless horsepower behind it. (I guess that such a system can also attract and impress clients. Because I'm not running a commercial facility, that aspect doesn't matter to me, except to treat friends to an occasional "audio sauna."

The monitor system should be actively amplified, *period*. Because of the addition of series capacitors, inductors and resistors between the drivers and the amplifier, passive systems provide less control over the dynamics of frequency components. I decided early on that a 3-way system would sound best, *if* I could make it work. For an ultimate tracking monitor, a 2-way puts too much demand on the high-end of a woofer and the low-end of the tweeter.

Low-frequency response

I thought that if the horn-room gave me a lot of gain, there would be no need to use a pair of 15-inch speakers per side for normal tracking levels. Although a single 15-inch driver functions as a pure bass source, it will also produce more excursion and related nonlinear distortions than a pair operating at the same SPL. On the other hand, a pair of 15-inch drivers will load a modern amp a bit better and produce less Doppler distortion, but will produce a nasty directionality spike (read "quack") at 1kHz or so.

One way around this problem is to cross-fire the two drives and cross them over at a lower frequency, which I definitely wanted to do anyway. (A single "good sounding 18-inch" would provide a good compromise, but my boxes were cut out for 15s. There is, seemingly, a very difficult to violate relationship between deep bass and "punch"; the more you have of one, the less you have of the other. By nature, low-efficiency woofers (1.5% efficient or so) produce great sounding deep bass, but tend to lack "attack." High-efficiency woofers of more than 5% have great attack, but generally no deep bass. Woofers in the 2% to 3% efficiency range seem to work best for deep bass of less than 30Hz and goodly amounts of "smack-o."

Last, if you listen to *comparisons* be-

Table 1. Parameters for "Henricksen-determined good LF and punch balance" woofers.

	EV Sentry III	EV DL18W	Special Altec	JBL 2234	TAD 1601
f_s	38	25	33.1	23	28
No	3.0	2.9	24	2.0	2.4
$V_{AS}(ft^3)$	8.0	19.20	10.56	16.2	12.63
Q_t	0.38	0.27	0.377	0.22	0.29

Table 2. Adjusted values of Table 1, with Q set to 0.38 for all units.

	EV Sentry III	EV DL18W	Special Altec	JBL 2234	TAD 1601
f_s^1	38	35.1	33	39.72	36.7
No	3.0	2.7	24	2.0	2.4
V_{AS}^1	8.0	9.69	10.56	5.43	7.36
Q_t^1	0.38	0.38	0.38	0.38	0.38

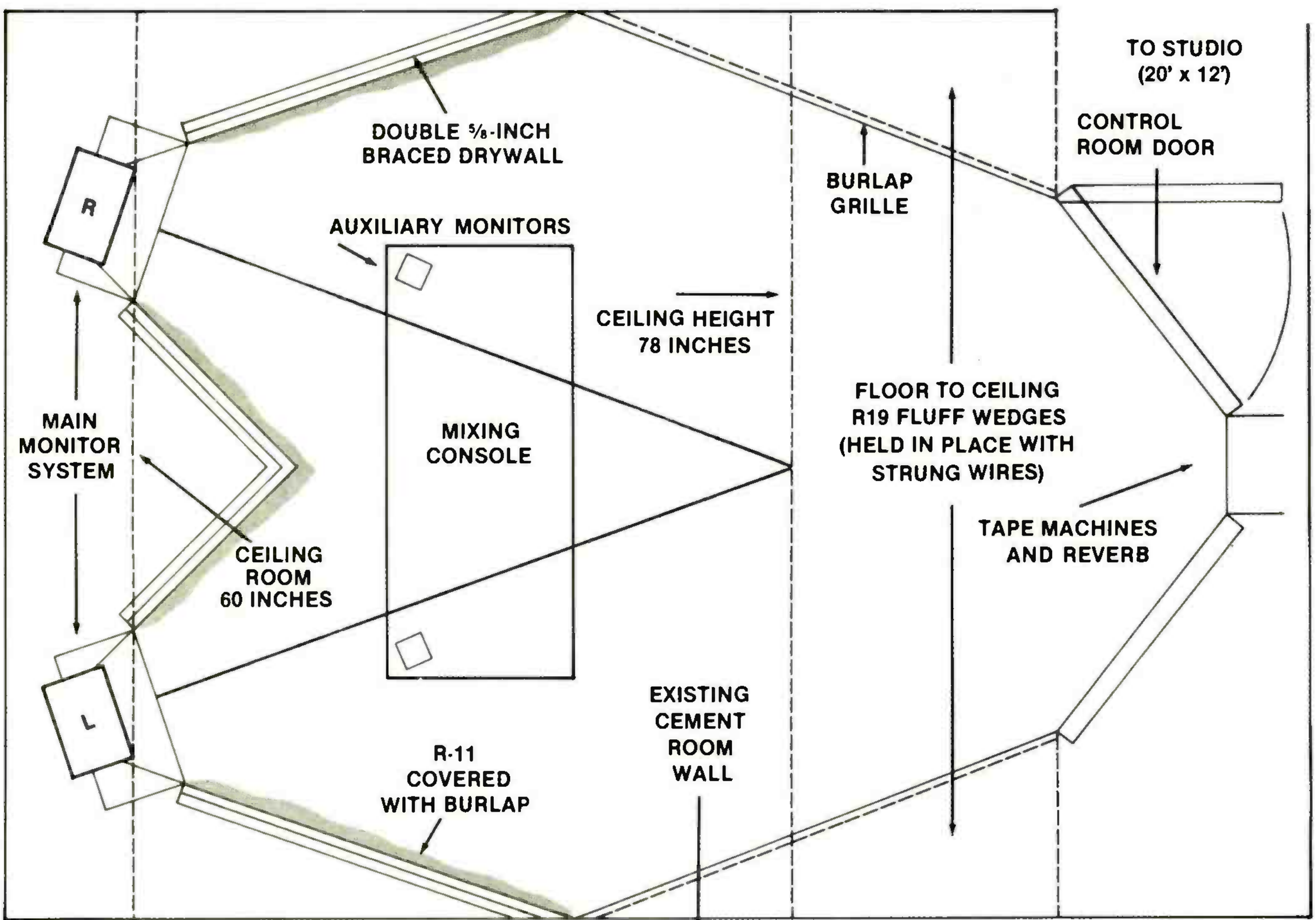
tween woofers in a similar acoustical environment, you'll always hear differences, independent of the test situation (big box, small box, vented box, closed box, no box, etc.).

Table 1 contains relevant Thiele-Small-type parameters for woofers which, from personal listening tests, I consider will work well in a large box, and fall into the good punch-and-deep bass category.

There is an interesting way to compare these woofers: Set all the Q values to 0.38 thereby turning them into (or fooling them into being) Butterworth low-end drivers. The low-end corner frequency essentially would be directly proportional to f_s and the box size would be V_{AS} if you raised their resonance to make Q_t equal to 0.38. Table 2 lists the adjusted values with Q set to 0.38 units.

Note from Table 2 that all five drivers end up measuring about the same; f_s is in the 30Hz to 40Hz range, and V_{AS} is between five and 10 cubic feet. Interesting? I can't make sense out of it. The Sentry IIIs get down the best, and the "special Altecs," built up from old Alnico 15-inch guitar speakers, have the best overall sound, especially below 250Hz.

Floor plan and acoustic details of author's personal-use conical bass horn control room.



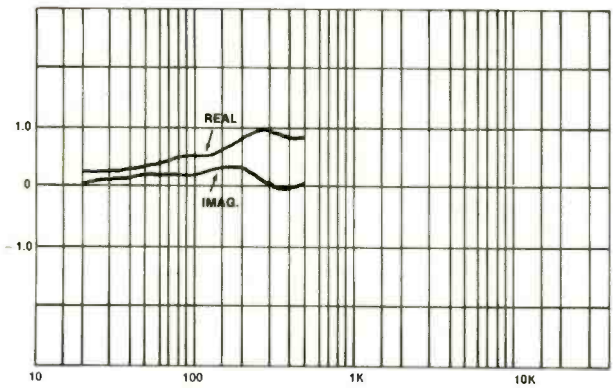


Figure 1. Normalized acoustic impedance of the control room, with Pc termination.

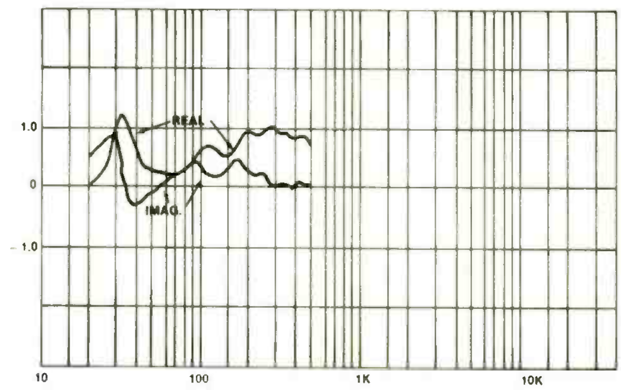


Figure 2. Normalized acoustic impedance of the control room, with free-air termination.

I included the 18-inch DL18W as an interesting comparison because it has a nice balance of get-down and punch. If you want to do other comparisons, use the Q ratio (0.38/Q) to scale f_s and use this ratio squared to scale V_{AS}. Look for woofers with an efficiency between 2% and 4%. (Note: This is my *personal* opinion. Trust your own ears.)

Midrange considerations

I worked at Community Light & Sound for three years, mainly on the development of the M4 midrange driver. All things considered, I don't think there's a better diaphragm for the 200Hz to 2kHz range. Basically, the M4 is an extremely strong motor (5 grams of aluminum coil in a 1.7 Tesla field) coupled to an advanced-composite diaphragm (aluminum skins and rigid foam core) that has its first significant distructive resonance up around 3.8kHz.

I removed the phase plug and used the M4 with a 1:1 compression ratio, 150Hz cutoff hypex horn; it sounded even better than stock. The horn mouth is a bit small (28"x11"), but the room walls help give it a solid 160Hz response.

High-frequency drivers

I opted for a compression driver high-end. For high fidelity, I don't like the idea of high-compression, and driving a diaphragm through a complicated labyrinth of acoustical paths (phase plug), which basically gets in the way of the diaphragm's natural sound. In their favor, compression-driven loudspeakers have the best acoustic "jump" when it comes down to producing high monitor levels during tracking, and they can direct sound into a waveguide (horn) that can put sound only where you want it.

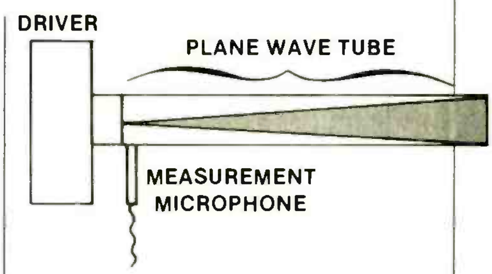
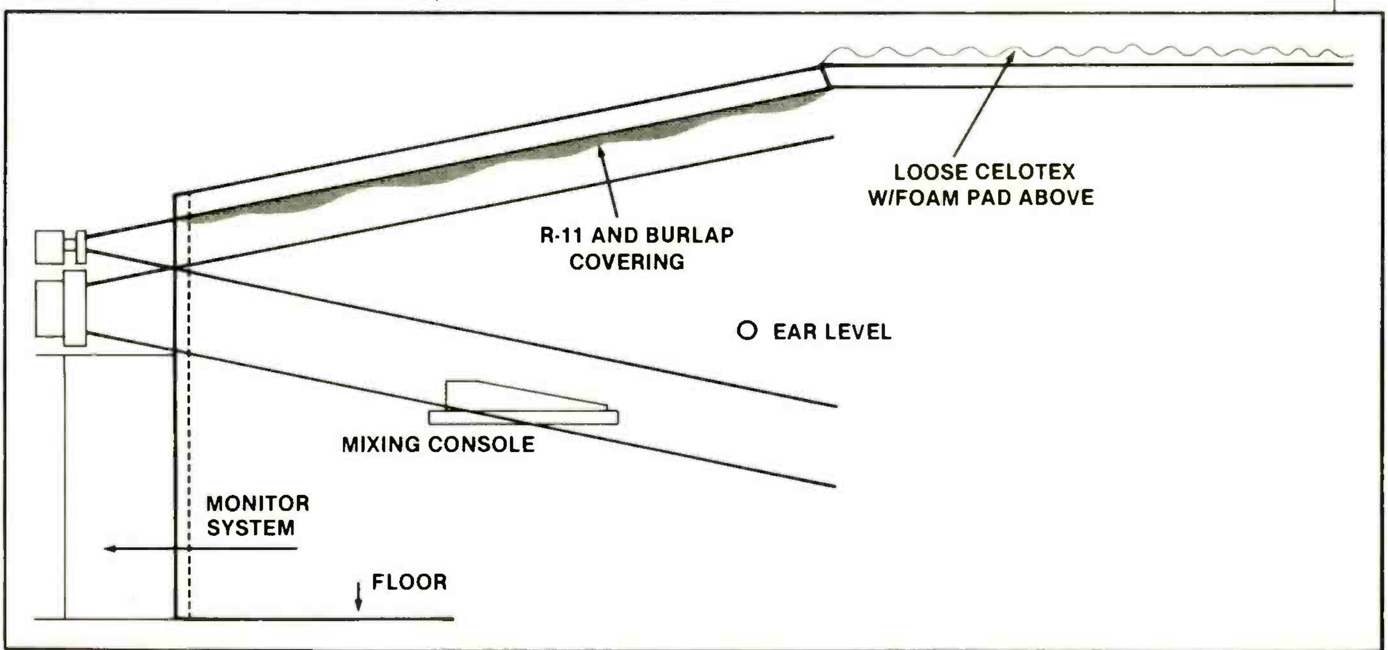


Figure 3. Elements of a plane wave tube.

It's yet another compromise; high compression results in a complicated design with wide bandwidth and high output for a small diaphragm. Use the same diaphragm as a direct radiator, and it'll have much lower efficiency with its intended magnet, much less total power output capability and, to my ears, it will not sound as good.

Sideview and monitor orientation for author's personal-use control room.



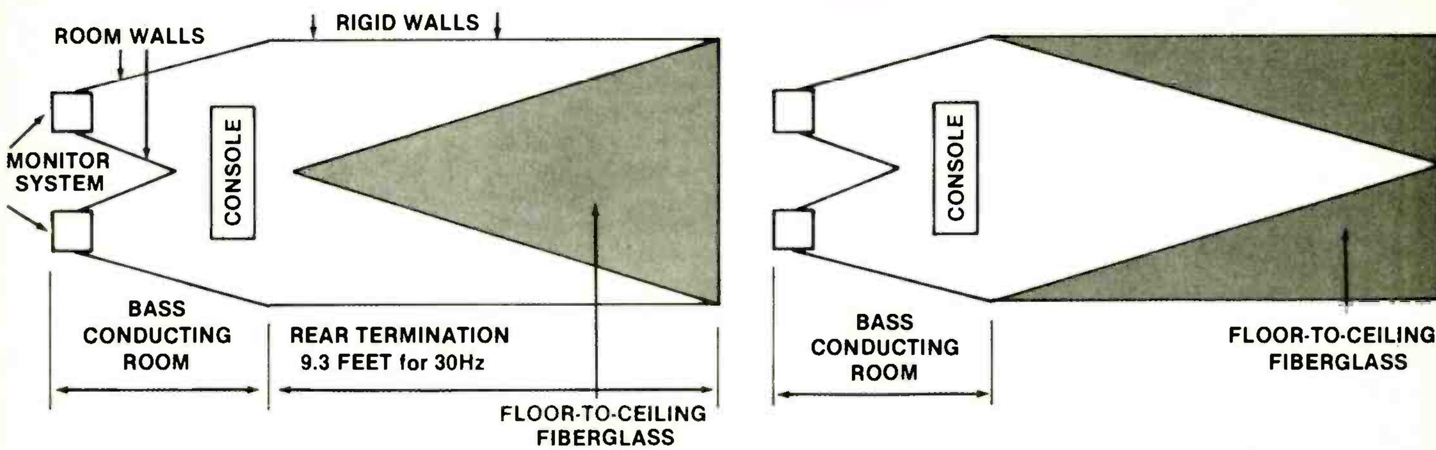


Figure 4. Two alternative control room layouts using the principles of plane wave termination.

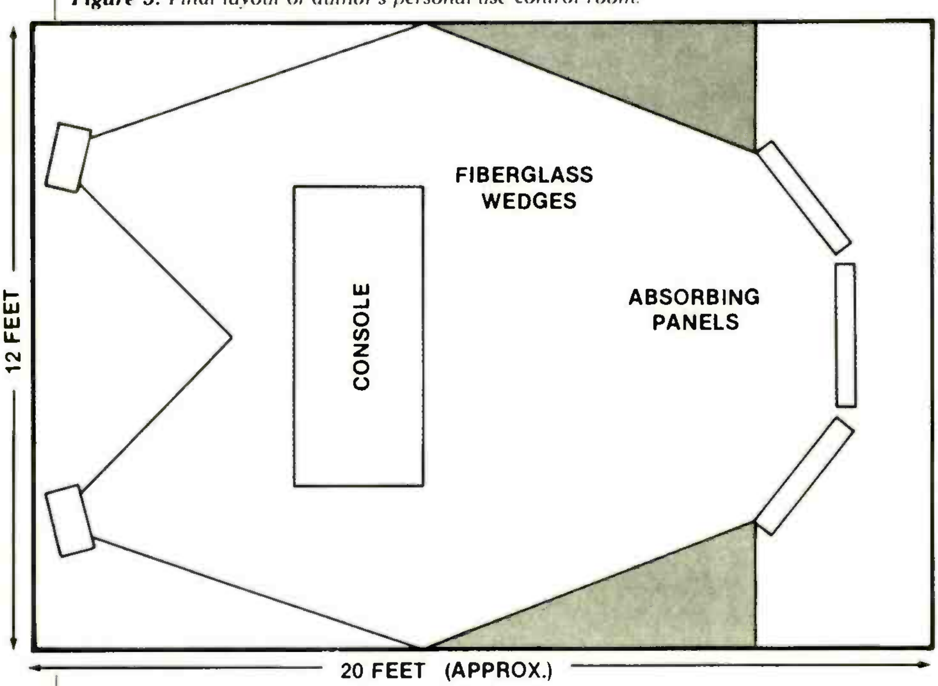
Having listened to several different compression driver diaphragms as direct radiators, I found that they generally sounded pretty unfamiliar. Normal direct radiators simply are too inefficient and don't swing enough pressure effortlessly enough to be worth pursuing. Plus, they have no directivity control.

The best compromise I've found is a well-designed compression driver with as large a diaphragm as possible. But, there are limits. The larger the diaphragm, the better the overall sound, but the lower the breakup frequencies. Smaller diaphragms tend to have a "pinched" vocal sound and complain more when you hit them with vocal, snare, electric keyboards, etc.

Very large diaphragms, however, have upper resonances that invade the passband so much that they can do an audible job of transient smear: confusion on snare, clean guitar, hi-hat, etc.

I tried several drivers; the choice is pretty limited. I ended up with the TAD 4001, a pair of original Alnico Altec 288 prototypes, which I own, and the new EV DH1A. The beryllium diaphragm TAD 4001 sounds great, and I admire its design and construction. It is a bit "dry" sounding, has somewhat of a mild "quack" around 4kHz, and lacks a bit in vocal warmth. The 288s have a nicer vocal and trombone sound, but the response zips up around 15kHz (the 288 "zing," which its proponents love).

Figure 5. Final layout of author's personal-use control room.



The DH1A seems to get around the "titanium-sounds-bad" myth, and is the driver I'm now using. For your system, check out at least these three models. Above all, don't take my word for it; once again, trust your own ears.

As you might guess, the main problem with every large format compression drive is the fact that, somewhere above 9kHz, they produce nasty diaphragm and acoustical resonances. These three drivers pose the least problems, but it's still there. The alternative is to crossover into a super tweeter and quad-amp the system. Try this: Obtain a variable active crossover, set it to 7kHz or 8kHz high-pass, drive a variety of "supertweeters" with known good material and see if you can find any that sound familiar *and* have any output capability before failure. I think the best approach to this is to stay 3-way.

I designed a special conical horn for the system, strictly for keeping high frequencies off the console and into your face. It's set up nominally with an 80°x20° coverage pattern, and will handle the 20° vertical from about 5kHz on up. Most people will not go to the trouble of designing and building their own horns; I did so because I think I know what I'm doing (which is not always true). Any high-frequency horn will have to be listened to carefully, and the specific horn/driver combination may or may not work.

In general, if your friend speaks to you through the horn, with his mouth up to the horn throat, and it sounds "good," or like your friend's speaking voice, you'll probably be able to use it. Horns that sound funny during such a test should be discarded immediately, independent of any data you might have to tell you the reverse. Horns that honk and quack are real problems; you can clean some of it up with EQ, but why add *that*? Use system EQ as little as possible.

Another way to review components is to listen to them by themselves. Hook up the horn/driver (or any component) to an actively crossed-over amp and play it in the intended range. Yet another way is to *visually inspect* the horn; units that look pure and well detailed inside will perform better. Shiny surfaces are better than rough ones for high frequencies. Exponential radial horns beam up vertically, raising or acoustically equalizing the horn's high end, and tend to keep 4kHz and above off the console. Tough choices, all.

tube. Inside the tube is usually a wedge made of an absorbing material, such as fiber glass, with the small end at the entrance and full width at the enc.

The room can be built just like a plane wave tube, as shown in Figure 4. The longer the room, the lower in frequency it will go, as a ¼-wavelength device. In other words, if you want the rear end of the room to absorb to 30Hz, the room has to be about 9⅓ feet in length.

Given my available space, I could only achieve a semiwedge. Then, to the rear of the mix position, I augmented this with a ½-inch celotex ceiling, loosely fastened with nails and washers with a rubber carpet pad above it. The rear walls are open wood-framed and hung with R-11 and rubber carpet-pad composite, as shown in Figure 5.

I built the console on an open-air platform, so that the bass will pass through the listener at the mix position. I treated all of the control-room hardware as if they were located in an "airflow" of bass; in this way, low frequencies will pass

Constructing the room

First, I gutted the existing room. Then I positioned the speaker systems in the bare room so they looked "right;" I can't describe the process any differently. I sat at the approximate listening position, looked at the system, thought about it, changed the position of the system, looked and thought some more, and continued this way until everything was OK from a "feel" point of view.

Then I sketched the room around the speaker systems, with the centers of the system focused a bit behind the listening position. I decided that the horizontal layout was very important in that it should be coincidental with the mid- and high-frequency's horizontal wall layout, without interface reflections.

The ceiling boundary started at the top of the HF horn and went up at about 10°, reaching full height (about seven feet) just behind the mixing position. I decided against a sloping lower floor at the front, because the vent in the box had to load into the room. I also thought that the floor, being concrete, would do more for promoting bass than any structure I could construct.

The result is shown in the accompanying diagrams: a pair of conical bass horns with about a 28"x60" throat at each, expanding to an 84"x60" mouth, just behind the listening position. Figure 1 shows the theoretical response into a ρc termination, while Figure 2 shows the theoretical response into free air.

The difference between the ρc and free-air mouth terminations bears some discussion. The latter would be possible if you opened the room into the side of the building and, in essence, exhausted the bass into the neighborhood. Besides the problem of angering the neighbors, you get some nasty reflections and ripples. The ρc case is much better, because there are no reflections.

In the lab we make these kinds of devices to load compression drivers; they're called plane wave tubes, as shown in Figure 3. A PWT acts as a standard acoustical load whose value is the numerical product of the density of air (ρc), the speed of sound (c) in air (thus ρc), and the inverse of the area of the

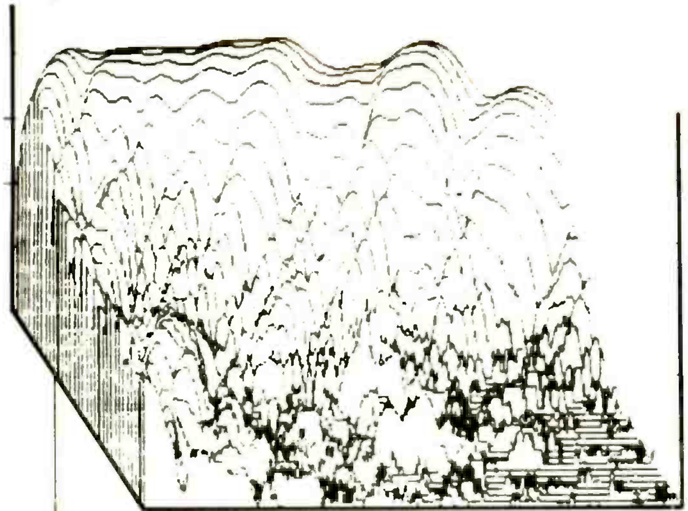


Figure 6. Time Delay Spectrometry (TDS) low-frequency response of control room at console position, 20Hz to 200Hz (100Hz at center) with mid and HF components turned off. Vertical scale: 12dB per division with base of display at −61dB. Horizontal scale: 20.35Hz to 201.31Hz (49.48Hz per inch).

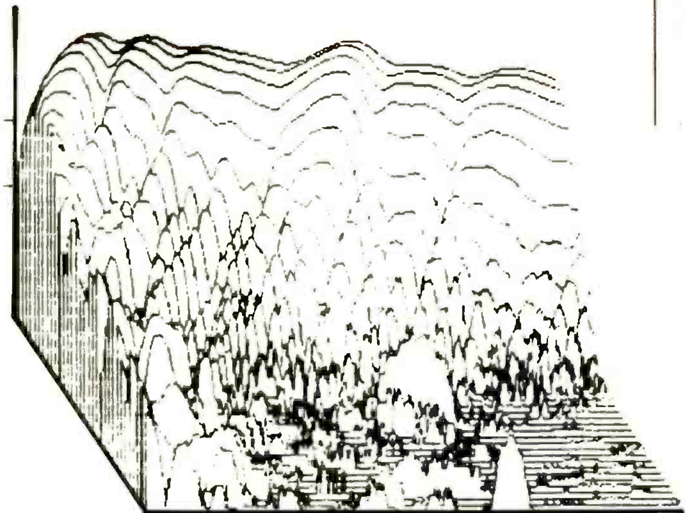


Figure 7. TDS low-frequency response of control room in nearfield. Measurement parameters identical to Figure 6.

through or over them most effectively. If you have a console that is closed off at the base, you wouldn't get as much of a total body feel at the low-end.

Mixing in free air is much better than being located behind some large obstruction; I definitely wanted the bass to be a *directed flow* rather than just an SPL level. I placed all the power amps directly below and to the rear of the console near the speakers. I use an Altec 9440 for bass and an AB Systems model 710 stereo to biamplify the mids and highs. A Crown VFX2A handles the low/mid crossover.

Author Cliff Henricksen beside the 3-way custom monitoring system used in his conical bass-horn control room design.

The first time that I turned the system on, it sounded awful. Don Davis of Syn-Aud-Con suggested that I quiet down the front end with fiberglass and make the rear more reflective, LEDE-style. That makes sense for mid-highs, but not for bass.

Glen Phoenix of Westlake Audio suggested using some wedge-shaped absorbers toward the rear sides to quiet a 100Hz cross-room mode. I finally realized that I was only half done. The trick was LEDE, only invert the matrix for bass.

Actually, I not only needed a rear absorber, but a rear wall to prevent the sound of the house's oil burner from getting into the way. (And not only when it's operating; kick-drum excites it in the most hideous way, like a big, steel resonator!)

First, I stapled R-11 sheets to the walls and ceiling, and added carpet and pad to the floor; much better. I made the rear-end glossy, with loose ceiling panels of celotex, open rear walls with hanging carpet pads and R-19 behind the openings. I also installed a pair of floor-to-ceiling wedges of solid R-19, starting about the mixing position and expanding out. I cut out 6'x2' triangles and simply stacked them up until they hit the ceiling.

Covering the fiberglass is an interesting study in the use of local available materials. Rapidly discovering that burlap is good, but expensive, I hit on the idea of using grain bags from the

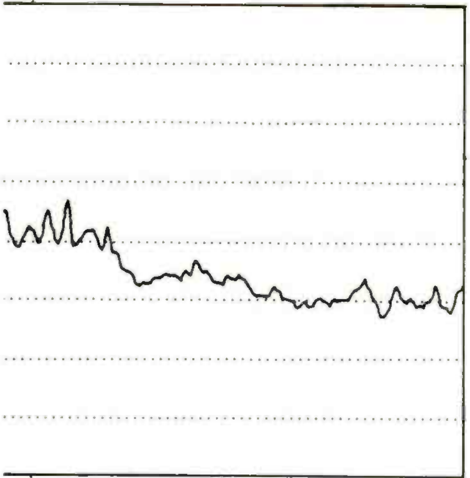


Figure 8. EFC (level vs. frequency) of control room with all components powered. Vertical scale: 12dB per division with base of display at −73dB. Horizontal scale: 39.62Hz to 19.998kHz (5.456kHz per inch).

local Buchanan Co-op. They fold out to about 36"x60" when you undo the stitching. For $10 I eliminated unsightly fiber glass, disguised by control room as a grain warehouse and saved a bunch of money. I also had some left over for gobos in the studio.

I ran response curves on the burlap covering and found it to be completely transparent up to about 18kHz, above which I got a little (1.5dB) loss.

I located all the reflective hardware, such as tape machines, outboard gear, etc. toward the rear of the control room. I tried voicing the system with a pair of third-octave equalizers and pink noise, but achieved better results by using a sinewave generator while viewing the output of a flat microphone mounted at mix position. My biggest problem was a hot spot in the 50Hz and 100Hz bands, and some strangeness between 400Hz and 600Hz caused by console reflections.

By the time I had sorted out these minor funnies, differences between well and poorly recorded material became very clear. Material with great bass provides a lovely audio message; the system went down clear into the mid-20Hz region without complaining. The 400Hz-600Hz range still sounds odd, but I'm told this frequency range always presents a problem in control rooms. My next project is to install some absorbing material around the rear of the console, and more Haas-Effect reflectors behind the mix position on the rear walls.

I played with the lateral placement of the superbly aligned mid-high section for time coherence. There was a vaguely different "halo" effect when woofers and mid/highs were in exact alignment, but it was pretty vague; my choice of a 200Hz crossover frequency probably accounts for this. It's probably quite different at the mid-high crossover of 1.2kHz, but I had that covered.

The room's measured response is flat and smooth; ETCs are sharp. Also, mid and high frequencies are right on the money with regard to first arrivals, as shown in Figures 6, 7 and 8.

All in all, the monitor system is a pleasure to work with in my basement studio, and it really tells me what I have on tape. What more could you ask?

R·E/P

Photos by Michael Leader

Acknowledgements

My thanks to Dave Gunness and Don Keele for helping run and collect TDS curves; Dave Carlson for generating the theoretical bass curves; Alan Shirley for help and suggestions in voicing the system; Bill Putnam and Allen Sides for inspiration; Glenn Phoenix and Don Davis for very helpful advice; Greg Hockman for encouragement and bass-massage appreciation; and Mel Lambert for cruel and merciless editing suggestions, making this a much more useful article.

NAMM Winter Market Replay

By Paul D. Lehrman

A report of MIDI-capable sequencers, effects units and musical software shown at the recent Anaheim NAMM exhibition.

The NAMM Winter Market exhibition, held Jan. 16-19 in Anaheim, CA, had few significant suprises. In many ways, it was and incremental show, in which manufacturers showed working models of previously announced products, or upgrades to existing lines. Despite the lack of major innovations, the NAMM Winter Market was a satisfying show, if only because it demonstrated how the electronic music industry is willing to let technology mature, before plunging off into new worlds and new civilizations.

Mixing automation

Certainly one major direction that several manufacturers are heading, according to evidence on show at Anaheim, is MIDI-based studio automation. Several companies are getting their feet wet with the simple approach, in which MIDI program changes handle channel mutes and effects routings. **Soundtracs,** whose products are now being distributed in the United States by AKG Acoustics, is using that approach for its PC MIDI consoles, as is Allen & Heath Brenell in its CMC, KMR and Sigma lines.

Other companies, however, are using MIDI on a more sophisticated level. **Simmons,** the electronic drum manufacturer, surprised a lot of NAMM attendees with a deceptively simple looking unit. The SMP 8:2 is a completely programmable 8-channel mixer occupying a single unit of 19-inch rack space. A total of 64 storage memories each contain level, pan position, two effects sends an 3-band EQ data (with frequency-adjustable midrange) for each channel. Each program in memory also contains a crossfade time, adjustable up to 10 seconds. In addition, each channel can have its own LFO effects, which include autopanning and frequency-sweep filtering, the rates of which can be either preprogrammed or determined by the input signal level.

Paul D. Lehrman is a Boston-based free-lance writer, electronic musician, synthesist, producer and regular contributor to RE/P.

A-kia Electronics' X-Z 100 system, which was first shown at last summer's NAMM show, has since been considerably upgraded. Based on a Commodore 64 computer, the system comprises various interfaces and cables and one or more "Recording Studio Computers," which essentially are 16-channel VCAs. The units communicate with one another via MIDI commands, but the designer has modified the protocol in such a way that the system is *not* compatible with standard MIDI sequencers. This is of little consequence, however, because the system software is quite complete, with both snapshot and real-time control on up to 128 channels. The system can sync up to any MIDI or time code timing sources, or read and write its own tape-sync signal.

J.L. Cooper Electronics showed a MIDI- and time code-based automatation system including a new device, the Mixer Automation Gain Interface. Available in various configurations of up to 64 channels, MAGI consists of a VCA unit, a master controller and a remote with eight "soft" faders. The various sections communicate via MIDI, using either note and velocity or controller information. They can be set up to work with a standard sequencer, or "invisibly" in conjunction with the company's SAM (SMPTE Automation Manager), a time code and MIDI Time Code addressable event controller and sequencer.

The MAGI system is already available as an option on several Amek/TAC consoles, and the company is in discussions with several other high-end console makers. External software is reportedly being developed by Digidesign.

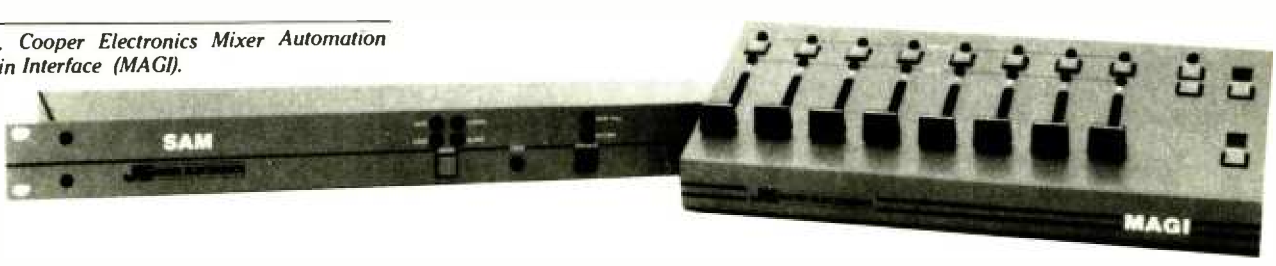
TOA showed a prototype of an automated mixing system based around several 4-channel modules being controlled by an NEC computer. The modules can be configured through software to act as mixers, equalizers or processors. Staff from TOA, both American and Japanese, were actively soliciting comments on the system from attendees. "We're listening to people," one spokesman told me, "as we design the control surface." The goal, he added, is "to develop a 32-input mixer in an attache case."

Processing automation

No one argues that MIDI-controlled signal processing is an idea whose time has come, and yet there were few new products that use real-time MIDI controllers or other performance data to modify settings. There were, however, plenty that change their identities when sent a MIDI program change.

Roland, for example, showed the DEP-3 effects processor, which contains

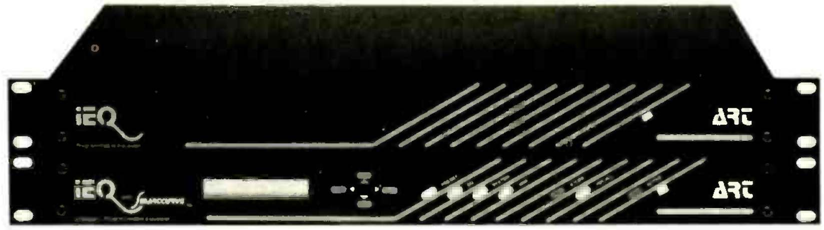
J.L. Cooper Electronics Mixer Automation Gain Interface (MAGI).



eight reverb programs, a straight delay and a 3-band digital equalizer. A total of 99 storage memories are accessible via MIDI. **Ibanez** also showed an inexpensive effects unit, the SDR 1000+, which features 10 reverb/delay programs—three of which act as "dual" programs with independent control over left and right signal paths—and a 4-band EQ section. The unit has 200 storage memories, 128 of which can be accessed through MIDI, and a MIDI controller can be assigned to the unit's effect level.

AKG Acoustics showed revision 2.0 software for its ADR 68K MIDI-controllable reverb and special effects processor. The new software release has some rather spectacular effects, including reverse reverb, inverted gates and "splits" that create two independent stereo reverbs. In most cases, there are no "holes" or glitches when programs are changed, allowing for smooth sounding transitions.

Applied Research & Technology showed its line of processors with real-time Performance MIDI, including two new Intelligent Equalizers or IEQs. The digital graphic equalizers are available in two models, both of which offer 15-band, two-third octave frequency centers. The Controller has an operational front panel and a composite video output, while the Satellite can only be operated by a Controller (up to 15 Satellites per Controller) or through MIDI. With either model, MIDI system-exclusive data can be used to adjust and store the 15 "slider" positions in any of 120 memories, and these can then be called up later with program changes.



Applied Research & Technology Intelligent Equalizers, (IEQ).

The company claims that the equalizers can be set up to automatically compensate for adjacent-band interaction and resonance effects of conventional analog graphic equalizers, with the "true" frequency response of the device displayed on a monitor connected to the Controller's video output.

Also on display was editing software for controlling ART's DR1 digital reverb from IBM-compatible computers.

MIDI interfacing

Synchronizing MIDI-equipped sequencers and other equipment is one of the hottest topics of discussion these days. As time code hardware finds its way into smaller studios, some interesting ways of linking the two synchronizing schemes are developing.

Allen & Heath Brenell, for example, is linking its MIDI-automated mixers to time code through the new CMPTE system, which was developed for the company by Synchronous Technologies. Using a Commodore 64 or 128 computer as a controller, the CMPTE system can read and write all four standard time codes, store 1,024 console setups and trigger up to eight external devices.

Hybrid Arts demonstrated a time code interface for its Atari ST520/1040-based MIDI sequencers. The accompany-

Adams-Smith Zeta Three.

ing software has an internal "tempo map" for executing tempo changes in the sequencer at designated time code locations. A similar system is used by **Southworth Music Systems**, which showed its JamBox/4 in full operation for the first time. (Interestingly enough, Southworth had the device controlling Mark of the Unicorn's Performer software, not its own Total Music.)

Fostex showed its new model 460 combination 4-track cassette deck and mixer. Both tach and servo functions, as well as time code synchronization, can be controlled by the company's model 4050 autolocator and controller in a multimachine hookup.

Adams-Smith showed the Zeta Three, a multiformat time code reader/generator and MIDI converter. The unit can be controlled externally through both RS-232 and RS-422 serial ports, operate slave transports and external processors, and even provide compensation for the ballistics of individual tape machines.

Just before the NAMM show, it was announced that the MIDI Time Code specification, in development for nearly a year, had won final approval by the Japanese MIDI Standards Committee. Few software or hardware manufacturers, however, were prepared to say exactly what they planned to do with it, and there were some voices raised that advances in the economy of directly converting time code to MIDI and *vice versa* might just make the format moot.

Sequential, for example, has implemented MIDI Time Code in its new Studio 440 but, as one demonstrator at the show said, "There's nothing out there for it to talk to yet."

Described as a combination drum machine, sampler and sequencer, the Studio 440 contains an 8-channel, 12-bit/41kHz sampler with analog envelope and filter control, a highly programmable drum machine, plus a 50,000-note sequencer with 32 channels, 99 sequences, and 12 songs. The 440 reads and writes all four types of time code.

Some synchronizers have among their features the ability to allow incoming "taps" or analog pulses to set the tempo for whatever devices it is controlling. An interesting gadget that does *only* this function—converting incoming pulses to MIDI clocks—was shown by **Kahler**. The Human Clock can reportedly be used for live performance, replacing lost sync tracks, or to give sequenced material a more "human" feel.

Music software

Although the number of companies releasing music software seems to be getting completely out of hand, there were surprisingly few new players. Instead, established firms were seen as perfecting and widening their lines. Many companies are accomplishing this by becoming publishers of software developed by individuals or smaller companies that do not have sufficient marketing experience.

Typical of these is Music Software from **Dr. T**. Known originally for its Commodore 64-based programs, the Doctor has expanded into the Atari ST world with the KCS-ST sequencer and several patch editing/librarian programs licensed from Caged Artist Productions. In addition, the company is taking on the Macintosh market with a graphics-oriented sequencer program called Views, which is an update of an older program, Megatrack, whose previous distributor reportedly ran into some difficulties.

Another such company is **Opcode Systems**. Known for its Macintosh software, including graphics-based synthesizer patch editors and sequencers, the company has made the latest version of its sequencer file-compatible with Deluxe Music Construction Set, a music-notation program from Electronic Arts, and is distributing the latter program as well.

Sequential Studio 440.

In addition, the company is publishing a program for film composers, called Score!, which generates cue sheets, calibrated and timed score paper and production summaries; and Music Mouse, a real-time "intelligent instrument" by New York composer Laurie Spiegel. In this latter program, the movement of the mouse controls melody and harmony in an interactive fashion, while the computer keyboard changes such parameters as velocity, program number, harmonic structure, articulation and tempo.

Opcode also announced editing programs to be delivered soon for several synthesizers, including Yamaha's multitimbral FB-01 (the program includes a feature for converting DX-7 patches to FB-01 voices), the Oberheim Matrix-6, and the Kawai K-3, as well as the Akai MPX820 mixer.

Also showing an FB-01 editing program for the Macintosh was **Digital Music Services**. Known as FB Pro, the program includes some voice comparison features and extra "temporary" banks for maximum flexibility.

Another IBM program for the FB-01's "big brother," Yamaha's TX-81Z (also introduced at the show), was announced by **Bacchus Software**, scheduled for delivery in late February.

Passport Designs showed a brand-new (as in "not quite ready") sequencer for the Macintosh, called Master Tracks Pro. It seems to have incorporated many of the best features of other Mac-based products, and is the first sequencer for any computer with graphic controller editing, which will be a must for the next generation of totally MIDI-controlled studios. Also shown was Score (not to be confused with Opcode's Score!), a notation program for the IBM PC; and a sequencer for the Atari ST, called MIDIsoft Studio.

The Atari ST was also represented at many other booths, indicating that the computer, already very popular among European musicians, is making inroads into domestic studios. **Sonus** showed The MasterPiece, a full-featured sequencer for the Atari (and the Macintosh) and a compatible notation program called The Super Score. **Beam Team** showed several software modules for the ST: TRANSfORM-Xtrack is a graphics-oriented sequencer; TRANSfORM-Xnotes a notation program; and TRANSfORM-Xsyn a set of editors for Yamaha, Casio and Roland synths.

Mimetics has been working on a modular approach to implementing music on the Commodore Amiga, with different programs that utilize the computer's internal sampling/sound generating hardware, MIDI input and output, external sync, and even video image manipulation and generation. The user can set the modules up in a patchbay-like configuration, using various modules to control others, and taking advantage of the computer's multitasking capabilities to run several programs simultaneously.

Digidesign showed a subtle but significant addition to its Softsynth program for the Macintosh and most sampling synths: an FM synthesis page. In this new mode, the separate harmonics that the program normally uses in an additive-synthesis mode can instead be configured to frequency-modulate each other. Although this can be used to emulate a DX-7, for example, the possibilities are much farther-reaching: up to 32 operators can be programmed in almost any conceivable combination or algorithm, with envelopes 32 segments long.

Finally, **Intelligent Music** showed two products for the Macintosh. M and Jam Factory are not sequencers *per se*, but instead take live or pre-recorded input and manipulate it in various ways. Jam Factory, for example, can work on four musical ideas simultaneously and independently, changing scale, accent, tempo, rhythm and MIDI program for each, in real time, using the Mac mouse. A wide variety of other manipulations can be handled from a MIDI keyboard. Files created with Opcode's new Sequencer can be used as raw material in the program. R·E/P

Rapid Facts Card Numbers

Need more information?

For more details of products mentioned in this NAMM Winter Market report, circle the following numbers on the Rapid Facts Card located at the back of this month's issue:

Product	Number
Soundtracs PC MIDI consoles	125
Simmons SMP 8:2	126
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J.L. Cooper Electronics MAGI	128
TOA automated mixing system	129
Roland DEP-3 effects unit	130
Ibanez SDR 1000+ effects unit	131
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ART IEQ digital equalizers	133
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The author in his personal-use studio during recording of The Celtic Macintosh *project, surrounded by the various MIDI-equipped sequencers, keyboards and signal processors. From left-to-right: a Panasonic PV-1231V VHS videocassette recorder used during the PCM-encoded mastering stage; Auratone 5C Sound Cube close-field monitors; a Sony PCM-701ES EIAJ-format digital processor; Taxan video monitor (for use with the 701); Kurzweil K250 digital synthesizer; Yamaha DX-7 FM synthesizer; TOA model D-4 console and D-4E expander module; Lexicon PCM-70 MIDI-capable digital delay and reverb processor; Southworth Music System's MIDI interfaces and companion Apple Macintosh PC.*

An Electronic Music Case Study

By Paul D. Lehrman

MIDI-based sequencers, synthesizers and outboards now enable a producer/musician to create music entirely in the surroundings of a personal-use facility. This article provides an in-depth overview of a recent direct-to-digital album project based on traditional Irish folk songs recorded with Eighties technology.

For many years now, we've been hearing how the personal computer revolution will forever change the working habits of professionals in many fields. Musicians make up just one of the groups that has long waited in line to benefit from computerization of their art, especially now that MIDI applications have begun to mature.

Like many promises of the computer sages, this one hasn't quite materialized as fast as predicted, or in exactly the manner foreseen. There are artists working in the all-computer medium, but most of them are connected with well-funded research groups, like Stanford University's CCRMA, or Pierre Boulez IRCAM in Paris, or larger film and post-production houses that can afford a full-blown digital synthesizer system.

For more modestly endowed producers and musicians, however, computers have so far served merely as aids to production, to be used in conjunction with conventional recording tools and techniques. Multitrack tape is still alive and well (although for some it may now be digital), as is discrete hardware for mixing and processing.

There are still many steps and many people involved in a composer's realization of musical ideas: The traditional way of recording music is still very much with us: hiring a spacious studio and populating it with a group of musicians whose job it is to make some sense out of charts carefully inscribed by the arranger, who in turn has done his best to meet the needs of the composer.

But the next step, completely automated music by an artist working alone, is ready to be taken. With the right software, computers can replace almost everything standing between artists and the realization of their music.

This article describes how producers and electronic musicians can take the next step. It chronicles how this writer recorded an album of music alone, using equipment that almost any serious musician can afford to own. There were no other musicians; there were no instruments other than MIDI-equipped synthesizers controlled and sequenced by a computer, which also handled most of the balancing and processing chores; and there was no intermediary tape stage, the mix being done live to a 2-track digital master. Requirements of the project were that it sound basically like a live band; an acoustic/electric Irish dance band, to be precise.

Most important, the project was done on a budget of practically nothing. After an initial equipment investment of about $20,000, the cost of producing the finished tape was less than $100: half a dozen floppy disks, two high-grade videocassettes and a few days' rental of a PCM converter.

From start to finish, the entire project was done in my home studio, on my own schedule. From concept to completed master, it took only two weeks.

There were two general principles that allowed me to record the project in this way: the availability of truly convincing sounding electronic musical instruments

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"Chiff" voice made less strident to blend better with the K250's strings and horns. Several stock organ patches were made velocity-sensitive.

The DX-7's stock "Harmonica BC" patch was turned into an accordian by slowing down the attack (and moving the expression control from the breath controller, which I do not have, to keyboard aftertouch); and into a concertina by detuning and increasing the level of the higher-pitched operators. A stock 12-string guitar voice was turned into a Hammer Dulcimer by lengthening the decay and changing the keyboard scaling to allow more operators to sound at the lower end.

To duplicate the sound of a solo violin, the vibrato on the "Agitato" patch was made more coherent by "un-detuning" the operators; it was also made slightly brighter and given a heavier velocity-based attack.

Correcting MIDI data tracks

After constructing the songs, and more or less finalizing the orchestrations, I then needed to clean up the MIDI sequencer tracks. There were often subtle mistakes in some parts to be removed, which proved to be the most time consuming part of the project. Certainly if I had been more careful in fixing tracks immediately after I had played them, this step could have been eliminated, or at least made significantly simpler. Correcting the tracks as I went would have interfered with the flow of the composition process, however, and so I didn't mind taking the extra time at this later step.

Some mistakes were deliberately left in. For example, a chord on an electric piano may have been played raggedly but, rather than time-correct it so that all the notes played simultaneously, I left it ragged, which gave the track a much more live feel. Drum fills were sometimes slightly off, and I tended to leave them that way because it helped create rhythmic tension.

Generally speaking, MIDI sequencer tracks were recorded without aftertouch or modulation-wheel information. Cutting and splicing tracks with huge volumes of controller information can lead to disaster. For example, if you have a segment of a track in which the mod wheel has been set to full, moving it into the middle of a section during which no modulation is wanted can have very disconcerting results.

In addition, copying a MIDI channel with controller information and pasting it onto another channel doubles the amount of information the MIDI data stream has to deal with, and exponentially increases the possibility of MIDI timing delays when the sequence is played.

Therefore, aftertouch and modulation wheel were added after the fact, as overdubs on the appropriate MIDI channels. Aftertouch was sometimes added to a track by playing a note on the DX-7, far removed in pitch from the other notes on the track, as softly as possible, and holding that note through the entire pass, varying the aftertouch as required. After the pass was recorded, I could simply eliminate the one note I had played (a single motion of the Macintosh mouse), leaving the aftertouch data intact.

System connections

It was at this point that I had to deal with the idiosyncrasies and limitations of the various machines I was using. The K250, for example, although its MIDI implementation is extensive, has some odd features. For example, when the instrument is set to Multi Mode (receiving polyphonic information on more than one MIDI channel), only one channel will children and adult audiences. She and my sister produced a series of four concerts for adults during the 1985-86 season at a small theater in Brookline, MA. For the first concert, just before Halloween, they asked me both to perform live, and to come up with some taped background music to be played before and after the concert and during intermission.

For her St. Patrick's Day show, however, Sharon wanted something a little more structured: genuine Irish tunes that would put the audience in the mood of being on the Emerald Isle. The music had to be much more acoustic sounding, and could contain few elements unfamiliar to the audience.

About three weeks before the performance, I started arranging a few tunes. At first I wrote relatively simple arrangements and, after about four days, four tunes had been more or less completed. I was so pleased that the work was going so fast that I decided to keep working right up until the show, to see if I could finish an album's worth of material by then.

Soon the music began to take on a life of its own and, on top of arrangements that sounded like traditional dance bands, I found myself applying Vaughan Williams-style symphonic orchestrations, Phil Spector-ish sound-washed backbeats, Springsteen-ian bass lines, Fairport Convention-ish electric sizzle, and Windham Hill-like ambiences.

Song constructions

The tune, a single melody line, often without a fixed harmonization, forms the starting point of just about all Irish music. I started building most of the compositions for the album by choosing a lead instrument and playing the tune through once or twice. Because the tunes were being both arranged and simultaneously recorded as MIDI data into the Macintosh sequencer, there was no need to start with a rhythm section; in fact, this was never done. Instead, an internal metronome produced by the Total Music software via the Mac's audio generator was used as a click track most of the way through the production, often even after drums (if used) were laid down.

Because the instrumentation could easily be changed at any time after a MIDI sequencer track had been laid down, I was free to use just about any sound for the initial track. The usual choice was either a solo DX-7 violin patch, the CZ-101's flute preset, a clarinet/pennywhistle patch I created for the CZ-101, or piano or guitar from the K250.

Because the rhythm could always be corrected after the fact, there was also no need to use step-time entry into the sequencer. I also wanted to maintain, as much as possible, a "human" feel to the tracks. As it happened, step-time note entry was used only once on the entire album, to handle a counter melody for one of the faster jigs.

After the initial tune was laid down, obvious mistakes were cleaned up immediately, and then an accompaniment part added, using bass of some kind and a rhythm instrument such as piano, electric piano, accordian or guitar. To keep rhythmic errors at a minimum, these parts would be recorded at a slow tempo. Again, to maintain the human feel, the tracks were usually not rhythm-corrected. After recording the accompaniment, I would play with the lead line's orchestration, and spend some time looking for (or designing) an instrument that would fit.

With the right software, computers can replace almost everything standing between artists and the realization of their music.

Irish music, particularly dance music, relies largely on repetition, and therefore much use was made of the sequencer software's ability to generate automatic repeats, known as the "Double" function. If transposition were needed, I would use the "Copy" function, and then paste the copied section in after the initial section, with the computer set to transpose the section at the desired interval.

Continuously repeating sections, with the pitch raised by a half-step at each iteration, is an effective means of creating musical tension and movement. If more tension was needed, small accelerandos could be programmed in from time to time.

All through the process of doubling and transposing, I could experiment with the orchestration, and change flutes to horns (or double them at the octave), guitars to harpsichords, bass to tuba, etc. It was very fast work—once the initial musical elements were recorded, the compositional process simply became one of modifying and assembling those elements in a pleasing and coherent way. Occasionally I found that after changing an instrument, the playing style didn't match the new sound—(chords played on a piano sound very different from chords played on a guitar, for example)—and I would have to re-record the segment in question.

Drum tracks were handled in a somewhat different manner. At no time was the TR-707's internal sequencer used. The Total Music sequencer software allowed rhythmic patterns to be sequenced in the computer much faster, and with a far greater degree of control over both timing and dynamics.

First, Total Music was set up to loop a 4-bar segment, starting at the drum entrance. A basic pattern was built, using the K250's keyboard to "play" the individual drums. Then the pattern was rhythm-corrected and extended out to the desired length.

Drum fills were recorded onto a separate sequence. The software allows up to eight sequences to be played simultaneously and, by putting them on a different sequence from the basic pattern, they could be rhythm-corrected and edited separately without affecting the basic track. (For example, the basic track might by corrected to one-eighth notes, but I would want to rhythm correct a fill to triplet 16th notes.)

For some of the more symphonic sections, the drum tracks were recorded as individual notes, instead of repeating patterns. These were usually not rhythm corrected. If they needed to be, conformed to a beat, that was done by hand, one note at a time, using Total Music's graphic-editing mode, which allows placement of a note with a resolution of 1/192nd of a beat (a triplet-512th note). Many of these orchestral sections used drum sounds from the K250 instead of, or in addition to, the TR-707.

Sound textures

Because of the music's acoustic nature, several of the factor presets provided on the K250, including grand piano, strings, horns, woodwinds, vibes, guitar, harp, acoustic and electric basses, were entirely appropriate and needed no editing. I already had a good library of original patches for the CZ-101, including clarinets, woodwind choirs, basses, Minimoog-type lead sounds, and nice, thick "growls," and made few changes in those sounds. I also made extensive use of the CZ-101's flute and whistle factory preset.

When it came to defining and modifying sounds, the DX-7 received the most attention. Many patches I already had on hand were useful, but a few needed tweaking. To achieve a "singing foghorn" sound, I modified an acoustic bass patch by adding infinite sustain and detuning the operators fairly drastically. For an Oberheim-like vocal sound, I used a stock DX-7 voice patch and sped up the attack, at the same time smoothing out the pitch envelope and (again) detuning the operators.

Electric piano patches were modified to give more bite, and an interesting using a variety of synthesis methods (in this case FM, sampling and phase modulation), and the recent development of sophisticated MIDI control sequencers.

The instruments included a Yamaha DX-7, a Casio CZ-101, a Roland TR-707 Rhythm Composer drum machine and a Kurzweil K250 digital keyboard. Mixing was handled by a TOA D-4 keyboard mixer with a D-4E expander, allowing 10 inputs, stereo effects and stereo outputs. The only signal processing in the chain was provided by a Lexicon PCM-70 digital delay and reverb.

Monitoring was via a variety of speakers, including TOA 280MEs, Auratone 5C Sound Cubes for close-field monitoring, and a pair of Rectilinear XIs. The final 2-track digital master was produced on a domestic Panasonic ½-inch VHS videocassette deck and a Sony PCM-701ES EIAJ-format digital processor.

Controlling everything was a standard Apple Macintosh equipped with an external floppy disk drive, 512Kbytes of memory and Total Music MIDI sequencing software from Southworth Music Systems (Figure 1).

For various reasons, I did not fully utilize the PCM-70's Dynamic or real-time MIDI capability. Instead, I used it merely as a programmable reverb, something that it does very well. The K250 could have been replaced with a lower cost unit, although at least eight other synthesizers would have been necessary to duplicate the many tasks I required it to perform simultaneously. (It would have taken some time to arrange all the necessary patches or instrument sounds in those eight synthesizers.)

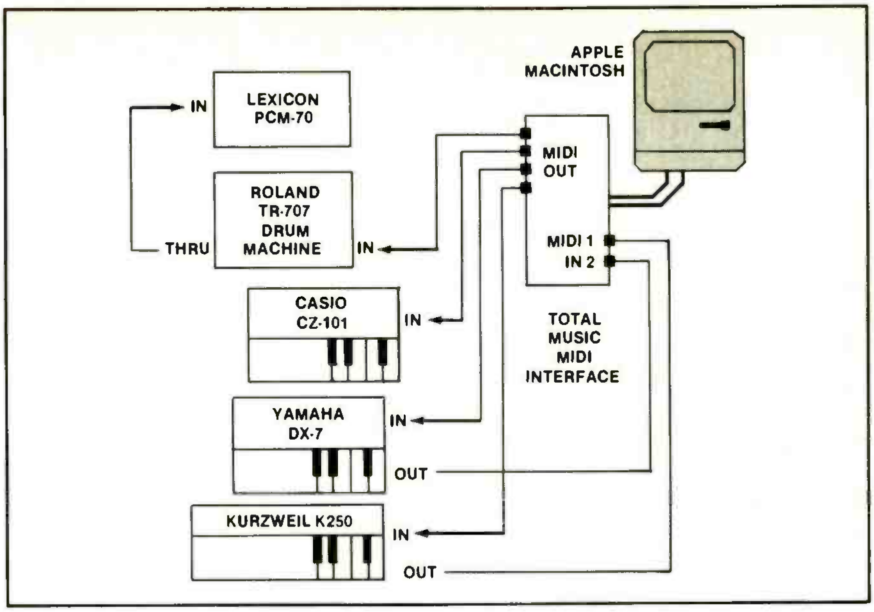

Figure 1. MIDI interconnect scheme for the computer-based sequencer, three keyboards, drum machine and effects processor.

The Celtic Macintosh, as the project ended up being called, resulted from a friend's request. Sharon Kennedy is a Boston-based storyteller who specializes in Irish stories and songs for both

MIDI Channel Number	Instrument	Keyboard Source
1	Piano	Kurzweil K250
2	Flute	Kurzweil K250
3	Woodwinds	Kurzweil K250
4	Baritone Horn	Kurzweil K250
5	Strings (fast)	Kurzweil K250
6	"Active Channel"	Kurzweil K250
7	Acoustic Bass	Kurzweil K250
8	Drums and Percussion plus Effects Programs	Roland TR-707 Lexicon PCM-70
9	Electric Bass	Kurzweil K250
10	Vibes	Kurzweil K250
11	Strings (low)	Kurzweil K250
12	Various	Yamaha DX-7
13	Various	Casio CZ-101
14	Various	Casio CZ-101
15	Various	Casio CZ-101
16	Various	Casio CZ-101

Figure 2. *MIDI channel assignments used during production of* The Celtic Macintosh.

respond to controllers (pedals, modulation wheel, etc.) or to program changes. Therefore, most of the instrument voices I wanted to extract from the K250 had to be pre-assigned to existing MIDI channels. If I wanted to use any other instruments, they would have to be assigned to the K250's "basic" channel, which I set up to be MIDI channel 6.

Because there were 10 channels available for assignment to the K250 (16 MIDI channels total, minus one for the DX-7, one for the TR-707 drums, and four for the CZ-101), this was not a serious drawback. As can be seen from Figure 2, the instruments being used the most were pre-assigned, and channel 6 was reserved for "specials."

However, if MIDI channel 1 was pre-assigned to a grand piano voice, I could not use the sustain pedal on it. When a piano track was recorded on 6, and then moved to 1, the pedal information was lost. On those occasions the track had to be edited so that the notes I wanted to sustain were made longer. I didn't use the K250's pitchbend at all—if a pitchbend command is sent to the instrument's basic channel, it bends *all* of the channels at the same time.

On the plus side, the K250 handled MIDI patch changes in a way that I believe is unique. If you send it a patch change while a note is sustaining (either because the key is held down or the sustain pedal is on), that note will continue to stand, without changing, until the key or pedal is released, and the patch change will only affect the *next* note played. Because note-ons and patch changes can be sent extremely quickly to the K250 without ill effect, I was able to use one MIDI channel for several different, virtually simultaneous voices.

For example, I wanted a timpani roll to end on a downbeat with a chime note. The roll was started on beat four, and a patch change for the chimes was inserted on clock 95, the last division of the beat. Then the chime note was drawn on the downbeat of the next measure, followed by a patch change back to the timpani on clock 1, and the final timpani note on clock 2. Any delay between the notes was inaudible.

The only other obstacle I came across while using the K250 was its limit of 12 notes sounding simultaneously. However, a "channel-stealing" algorithm built into the instrument means that, although the 12-note limit was exceeded on many occasions, I only found it necessary to eliminate extra voices on a couple of passages when things started to get really thick.

Pitchbend was also an issue with the CZ-101. The synth can receive homophonic (single-note; sometimes called, confusingly, "monophonic") information on four different (adjacent) MIDI channels, with independent patch changes on each channel, but will only respond to pitchbend on *one* channel. This may not necessarily be the "basic" channel (which in CZ-101's case is the lowest of the four). Therefore, I had to be careful assigning voices in which I wanted to use pitchbend, and also make sure the Casio was always set up correctly.

Another interesting trait (which is shared by only a few other synthesizers) is that if portamento is applied to a note, it will only be heard if there is no interruption between that note and the following one. In other words, if two notes are played legato on the CZ-101 while portamento is on, there will be a glide between them, and the second note will have no separate attack; but, if there is any separation at all, there will be no glide and there will be a fresh attack on the second note. This meant that in those sections in which portamento was used on the CZ-101, note durations had to be very carefully edited.

Another area of concern resulted from the way in which the DX-7 handles patch changes. If a change occurs while a note is sounding, the note does not finish, or even politely shut off. Instead, the instrument immediately executes the final decay stage of the *new* patch, which most often results in a metallic glitch. Even if the note is no longer being played by the sequencer, should a patch change occur any time before the synthesizer envelope is completely finished (and many sounds I was using had a long ring to them), the glitch will occur.

Under certain circumstances, the DX-7 will not respond to a patch change right away, but will actually delay the onset of a note following a patch change by a few milliseconds. Therefore, patch changes had to be placed extremely carefully, and sometimes notes cut off (or pedals turned off) earlier in order to make room for the patch changes.

The Lexicon PCM-70 digital delay and reverb unit responds to program changes quickly, but at a cost. When a MIDI patch change is sent, the input momentarily clears and the contents of the delay registers are dumped. Although this action doesn't cause a glitch, it introduces a hole into the sound that can be disconcerting. Because I was using fairly straight-ahead reverb programs, this problem could have been avoided by using the unit's Dynamic MIDI feature. Instead of actually changing the program, I could have used a keyboard controller to adjust the decay time and/or the wet/dry mix, which results in completely smooth transitions.

I chose not to do so for several reasons. One was the sequencer software's limited controller-editing capabilities, which meant that if I made a mistake it would be difficult to correct. Another was that the PCM-70 only has a mono input; if the wet/dry mix were placed under MIDI control, the stereo image would collapse toward the center as the reverb was diminished.

As it worked out, limiting program changes to the pauses between selections produced no audible effects. Because I was addressing the PCM-70 only with program-change information, and no notes or controllers, I could use one MIDI channel to control both the TR-707 drum machine (which, of course, ignores patch changes) and the digital reverb.

Live to digital 2-track

Because EIAJ-format digital processors, especially when used with a con-

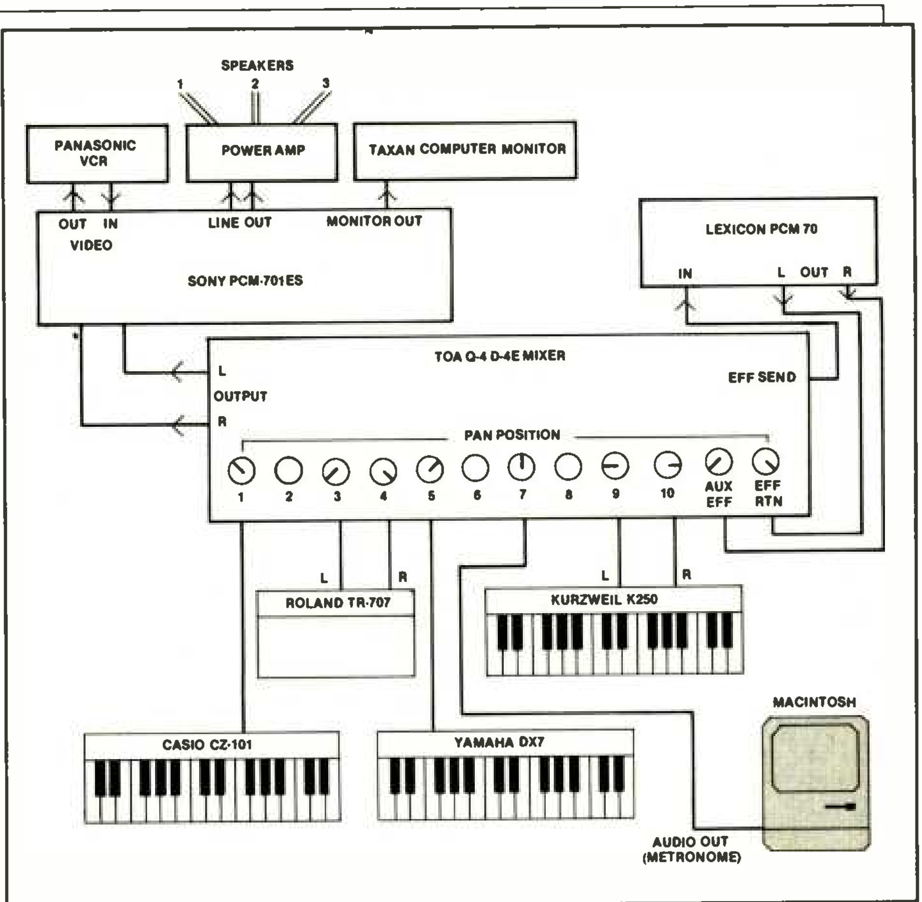

Figure 3. *Audio signal routing scheme during live-to-digital remix of MIDI sequences being replayed from the Apple Macintosh.*

sumer-grade ½-inch VCR, normally do not allow editing, I knew I'd have to record *The Celtic Macintosh* in just two passes, one for each side of the final cassette album release.

I assembled the sides in much the same way as I assembled the individual selections, by chaining the selections together with sequence calls. I could adjust the amount of space between selections by noting the tempo and inserting the call to the subsequent sequence a specific number of beats after the current sequence ended. For example, if the tempo of sequence 2 was 120bpm, and I wanted to leave four seconds between it and sequence 3, I placed the call to sequence 3 eight one-quarter notes after the end of sequence 2. Also taken into consideration when determining the length of the pauses was the time necessary to execute smooth-sounding program changes on the PCM-70.

Because each side of the tape was more than 20 minutes long, I knew I would have to keep fader moves to a minimum. If I were to make a serious mistake any time during the pass, without console automation I would have to start over. Fortunately, I had left all of the mixer levels pretty much alone throughout the recording process, which meant that the balances were more a function of how I played the individual tracks than how the mixer was set up (Figure 3).

When it became apparent that some level-adjustment was necessary, I could change the velocity of the MIDI track in question with Total Music (on a long track, or one with many notes, this could be very time consuming), or I could design a new patch identical to the one I had used to record it, but with a softer or louder overall level. Then, instead of doing a fader move live, as I was mastering, I could automate the process by inserting a program change into the MIDI track.

By adjusting velocity levels and patches, I managed to get to the point where there were fewer than a dozen fader moves on each side that had to be performed during the mix. Some of these moves were done on the D-4 mixer, while others were made on the synths' output-level sliders, which are much easier to calibrate.

Input level to the PCM-701 digital processor was adjusted by looking for the loudest spot on the album, and then setting the input gain so that it peaked at −3dB (the unit's display had been modified to read 3dB low, to provide better headroom). After that, mastering became simply a matter of loading the videocassette, calling up various sequences into the Mac, and pressing start.

I deliberately took my time doing the mixes, with long breaks between passes to keep my ears fresh. The first side took four passes, and was completed in an afternoon, while the second side, which took three tries before I was satisfied, was done the next morning.

During the recording of the second side, I decided to try something a little risky. On a lengthy violin solo on the final cut, I removed all of the modulation-wheel data, and instead "played" the mod wheel while the master tape was running, being careful not to make any gross mistakes so that I wouldn't have to start over. The risk paid off; the result was a more spontaneous, live-sounding track.

Cassette duplication

The digital master was then taken to a local real-time duplication facility. I had noted during the mix that the peak levels were 12dB above the pre-recorded reference tones (a safe 6dB below clipping). I explained this to the engineers, telling them I wanted the cassettes recorded at as high a level as their machines were capable to keep the noise floor as low as possible.

We decided that a touch of compression would improve the noise level. A limiter was used with a slope of 2:1 at +6dB; it kicked in briefly at two points on the first side and once on the second. We did a partial run of 30 tapes, which would be enough for the show.

The bottom line

Besides Sharon's immediate need for the music, there was one major reason for recording *The Celtic Macintosh*: to prove that it could be done. Like all musicians with delusions of grandeur, I have always wanted to have my own orchestra. But my delusions went beyond that; I wanted to *be* my own orchestra and ensure that all the parts were played just the way I heard them in my mind.

From this point, the technology can only improve and get cheaper. There are now inexpensive FM synthesizers and samplers on the market, as well as MIDI-controlled processors and even MIDI-controlled mixers. Software is getting faster and more comprehensive, and database-style music-editing systems that can automatically and rapidly perform sophisticated composing and arranging tasks will soon be commonplace.

MIDI-delay problems are being overcome by systems that use multiple MIDI data streams and synchronization between MIDI and the rest of the world is becoming more accurate and easier to use. R·E/P

Copies of *The Celtic Macintosh* are available from the author at 31 Maple Ave., Suite 1, Cambridge, MA 02139, for $10, including postage and handling.

emergency repair

preventive maintenance

routine maintenance

Service and Maintenance Contracts

By Richard J. Moore, Ph. D.

What sort of features should a facility operator look for when establishing an in-house or independent service and maintenance program?

Downtime is an expression that sends a shudder of terror into the hearts of all studio operators and their clients. One key element that every successful studio has in common is the reliable operation of the facility's equipment. Lack of good maintenance can make millions of dollar's worth of equipment sound like string with tin cans, not to mention a real pain to work with.

The creative process has many obstacles to overcome to capture a great performance; the recording and production equipment cannot hinder this process at all. Many of the really fine studios in operation today are successful because of reliable day-to-day operation, and not because of the bells and whistles.

Studio maintenance is a very important consideration. It can be also quite expensive to maintain a quality maintenance staff at a facility. No studio can be without some type of maintenance staff, but the degree of service coverage can vary. Should a studio maintain a full-time staff? Are outside service personnel available and competent?

In this article I'll attempt to analyze what is necessary to keep an audio recording or production facility up and running.

Studio maintenance can best be classified into three categories:

First, we have emergency repair. This is: Fix it now!

Second, there is preventive maintenance, a process by which equipment is checked and analyzed on a routine basis to maintain a consistent level of operation, and to spot potential problems before they become an emergency.

Third, there is routine maintenance, a day-to-day method that involves such aspects as tape machine alignment and console normaling. Routine service provides a good indication of how well the preventative program is going, and helps identify potential serious problems. All three programs are necessary for any studio to be truly covered.

Large facilities require a full-time maintenance staff. When there are three or more production rooms in use 24 hours a day, someone must be available to respond to any technical situation that may arise. In larger operations, the maintenance staff is well versed in all areas of the facility's equipment, and usually have attended factory seminars covering care and repair of major equipment such as consoles and tape machines. The only area that may require outside service would be specialized digital equipment. For the most part, however, very little outside services are necessary.

Medium-size facilities may find that a full-time maintenance staff is not necessary, because routine service usually can be performed by staff assistant engineers. In a medium-size studio, the real need is to find outside personnel to handle preventive and emergency repair services.

Small studios have basically the same needs as a medium-size facility, the main

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difference being in the frequency of service calls. They must rely, however, on outside services for their maintenance needs.

There are many factors to be considered when a studio is assessing their outside service needs. Geographic location of the studio is of primary importance. If your studio is located in a major recording center, such as New York, Los Angeles or Nashville, you are reasonably assured of a large pool of technical assistance. Local expertise can be in the form of individual technicians, large audio dealerships and manufacturer's technical representatives. If your facility is located outside one of these major markets, you'll need to find someone who is trained to repair your equipment and well versed in parts availability.

The type of facility you are operating can affect your service requirements. If you are set up for just audio recording and production, it's much easier to find a good electronics technician who can repair tape machines and consoles with a manual, and a good set of road maps. However, if your studio is equipped, for example, to handle audio-for-video production, then you need personnel versed in TV and video repair, time code synchronization and video connections. If you are a digital facility or a MIDI-based studio, then you'll need to find technical support in these specialist areas.

What is your client base? If your facility hosts major talent and first-rate engineers, regardless of its size, you should have a first-rate staff working on a full-time basis. Outside services are only required to repair the special and sophisticated equipment that you cannot handle. If your client base indicates that you run 24 hours a day, then you will need quick access to specialists, materials and test equipment to ensure no time is lost.

There are two final factors to be considered: the age and source of the equipment used in the studio. An older piece of equipment will require more attention than a unit straight out of the packing case. Also, it may be important to find technicians who know the gear and its particular quirks and know how to track down scarce parts.

The source of a piece of equipment is also very important. If the unit is new, a certain degree of dealer and manufacture backup will be available. If it is used equipment, you should try to find out as much information from the previous owners as possible. This will enable you to tell your service people what they might expect based on previous experience.

Finding outside service personnel

After you understand what kinds of outside service needs have to be covered, then find the people to execute your maintenance requirements.

We live in an era of specialization. Twenty years ago, when equipment components consisted primarily of vacuum tube circuits, finding a competent technician was not that difficult. Today, equipment is based primarily on solid-state circuitry using sophisticated analog and digital technology (with just a few odd pieces of valve gear floating around to keep the techs on their toes).

The outside service technician or organization with a high degree of competency in all the service areas is somewhat of a rare commodity; and they are usually concentrated in the major recording centers. The process of becoming competent at service and maintenance, by attending factory training and certification courses, is a very time consuming and expensive process. For this reason alone, excellence commands top dollar fees and salaries.

For a large, multiroom facility the annual maintenance budget can exceed $50,000 a year; a medium studio can expect to spend $25,000 a year. A small studio's expenditure can be almost negligible in comparison. Today, a reputable and competent technician will cost between $35 and $65 an hour, and sometimes more. (Parts not included.)

As mentioned earlier, a large facility *must* keep a qualified service technician on hand whenever sessions are in progress. Medium- and small-size facilities must count on service personnel that can be reached at a moment's notice. In a recording or production studio there is no such thing as "normal hours." Because of this, it is reasonable to expect the cost of service to rise according to the amount of time needed to repair a faulty piece of equipment. A late-night service call is going to cost more than a day call, and an emergency call sometimes more.

In looking for good service personnel, several parameters should be analyzed. First, what is their reputation like? The best way to determine this is to call around to other facilities that they service, and talk to the owners and staff to find out how well their maintenance operations are working.

Second, what kind of equipment (devices and brands) is the service person qualified to work on? Each studio has their own specific needs and, so that you are properly covered, you have a right to know what can and cannot be done.

Third, does the service person (or organization) work well with the staff of the studio? This is an extremely important consideration, because a good rapport ensures that problems are effectively communicated and solutions are made.

Last, how reliable is the service? No matter how great a technician is, if you can't count on them arriving when they are supposed to, it's a grave situation.

Remeber that maintenance is not just a *fix-it* game. Maintenance provides an interface between the engineers and musicians; handled properly it can make the engineer look good to the client, and the musician look good to everyone. It is a real low-key team effort, and more people-oriented than many realize.

In some areas of the United States, larger service organizations are able to provide contract services to their clients. Contracts are usually based on the facility size, the type of equipment, and the amount of use the equipment receives. Regular programs can be set up to cover the preventative needs of the various types of equipment. Such needs are usually calculated on the basis of how often equipment requires a thorough review of its operation and what measures have proven to avoid major system failures.

Many independent service personnel tend to shy away from any kind of service contract. If a contract exists on a specific type of equipment, such as a tape machine, and the unit goes down in the middle of a session, the studio may hold the technician liable for any lost session revenue involved. Because most independents are not financially capable of coping with this situation, only rarely will they enter into service contracts. The best thing to do, if you feel that a service contract is what you need, is to shop around.

Setting up a service program

One of the most important aspects of setting up both an in-house and outside maintenance program is to establish adequate lines of communication. It is extremely important that all maintenance tasks and schedules be arranged and adhered to. One of the most common complaints you hear in a good studio is that there is insufficient time to take care of maintenance. For this reason, service time usually has to be booked in the same way as regular session time and, with few exceptions, on a non-bumpable basis. One person at the studio should serve as the contact in arranging all maintenance time and should communicate the situation to all concerned.

Record keeping is one of the most cost-efficient ways of keeping a studio on-line. When setting up the maintenance program, the service technicians should sit down with the entire staff of the studio and explain the record keeping requirement. Trouble reports need to be designed and conveniently distributed about the studio for easy access. A trouble report that is written on a matchbook is not a lot of help when trying to figure out what to do.

The report needs to mention information such as where the malfunction occurred, the time at which it occurred, the specific piece of gear (for example, console mic/line module 22), and a decent description of the problem including what was going through the channel at the time and what pieces of outboard gear were attached. It is always best if the *exact* circumstances of a problem can be recreated as closely as possible.

Maintenance logs are extremely helpful in establishing an operational profile of studio equipment. For example, it can be extremely useful to know when the heads were last changed, when the capstan was last replaced, when the in/out switches on the equalizers were last changed, etc.

Many studios keep their maintenance log in their computer as a very large database. Such information is handy for quickly retrieving data on a specific piece of gear. The database does not have to be so grand, however; something as simple as a diary with accurate note keeping on a day-to-day basis would work just as well. The important thing here to remember is that the information should be coherent and available to the service technician when they come in to work on the equipment.

Routine maintenance can be performed by the assistant engineers at a studio. The service people you hire can set up a training program by which common maintenance can be handled by the in-house staff. Some of the regular duties should include routine tape machine alignment, console cleaning and normaling, cleaning patch-cords on a routine basis, lubricating mic stands and making up cables.

Outside of these routine duties, unless someone is very competent, equipment repairs should not be undertaken by the assistant engineers. There is nothing worse than an assistant saying that such and such a device is now repaired, only to have a more catastrophic failure in the near future. Routine maintenance combined with the preventative can create a technologically harmonious studio without a lot of added expense.

There are a couple of other temporary solutions to providing service and maintenance. Most manufacturers provide a period of after-purchase support which, depending on the type of gear, is usually from 90 days to one year parts and labor. With most new microprocessor-based devices, software updates are included for the life of the device simply for the cost of the PROM replacements. Outside of these, few if any manufacturers offer any kind of extended warranty or service contracts on their equipment.

For the most part, manufacturers have found that, with a good program of cleaning and preventative maintenance, a piece of equipment can last a long time. But with fewer and fewer studios, because of the cost, having staff technicians, things can go wrong. Manufacturers cannot afford to be liable for loss of business due to improper upkeep. They do offer factory help to analyze your problems, however, and help you come up with solutions plus reasonable exchange of parts that may fail.

A good pro-audio dealer is also an excellent source of initial technical support. Outside of purchasing proven equipment with a great track record, there is really nothing else to do about additional support.

As we have seen, maintenance and service provide the very backbone of a modern recording or production studio. Maintenance is not a glamour position with recording studios and, therefore, does not attract large numbers to its cause. The cost of such service personnel is very high, especially if you can find a responsible and competent one. It need not be an outrageous expense, however.

With some homework, a maintenance program can be set up to satisfy the needs of any size studio. There are many questions to ask, and many answers to be discovered. When you finally arrive at personnel and a program that work, hold on for dear life!

Most important of all, you should strive to make the whole process a team effort between the studio and the technicians. In our business, a happy client and engineer make for a busy studio, and that makes for a *happy* studio.

Remember that if everything works well to make more business for the facility, then maybe the maintenance program won't seem so costly or bothersome. R·E/P

Close-field Monitor Loudspeakers

By Neal Avron

■ Yamaha NS10M
■ Auratone T6
■ Tannoy NFM-8
■ Calibration Standard Instruments MDM-4
■ Westlake Audio BBSM-6

Originally, recording and production engineers began to use small, supplemental monitor loudspeakers because most people who listened to their recording would be doing so on domestic systems with a limited frequency and dynamics response. In general, control room monitoring comprised large speakers flush-mounted in the wall near the ceiling, and powered by multiple amplifiers—a situation quite unlike the home user's environment. As a result, a growing number of studios began to carry alternate sets of monitoring sources, which ranged from bookshelf and car-radio speakers to television speakers, to enable engineers to listen to a mix using equipment similar to a consumer's, and let them maximize the sound quality for that medium.

In the past, the use of large wall-mounted speakers meant that a distance of nine to 15 feet from the center point between the monitors represented a comfortable listening distance. Close-field monitoring utilizes smaller speakers and a shorter distance between the engineer and the loudspeakers. Although there is no exact specification being used, the majority of engineers opt for a distance between three and six feet as representing a useful starting point. Which works out well, because the average distance from the meter bridge of most consoles to the engineer is within that distance.

Working close to monitors provides a substantial decrease in sonic colorations caused by control room acoustics. In this way an engineer can achieve a consistent mix in even the worst listening environment, or reduce the perceived sonic differences when changing rooms. Two factors are at work: first, the amplitude ratio between direct to reflected sound is increased, thereby diminishing room effects; and second, the time difference between the direct and reflected sound is increased, such that less time-delay coloration occurs.

Also, close-field monitoring delivers a more efficient way of listening to a mix. Because of the inverse-square law effect, the reduced distance from the monitor to the engineer produces an increased sound pressure level proportional to the distance squared. The bottom line is a proportional decrease in the required amplifier power. Conversely, the same amplifier power would deliver a much higher SPL from the monitors at the engineer's position.

When taking advantage of this increased output level, one should also be aware of the Fletcher-Munson effect. Basically, these standardized response curves plot the way in which the human ear is sensitive to varying frequencies at different listening levels. Therefore, a mix that is monitored at a loud listening level will sound quite different at a lower level. (And, because most consumers listen at lower levels, it's an effect worth keeping in mind.)

Another reason for bringing the monitor up close would be to improve source localization. Because the speakers are now near the engineer, practically all of the early reflections that hinder good source localization become of little consequence.

Evaluation sessions

All of the conclusions from this review were drawn from two monitoring sessions at the University of Miami's Gusman Hall control room. In the first session, engineers used the speakers for monitoring mixes they were currently working on in the studio. After the session they were asked to comment on aspects of the speakers' performance. Although there were many variables present in this first session, it did typify the primary function of a studio monitor in

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Yamaha NS10M MkII

its everyday use.

The second session took the form of a more controlled type of test. To begin with, the listening sources used were from Compact Disc, which provided consistent, repeatable material for all listeners. All engineers were blindfolded and placed in the center between the two speakers located approximately four feet away. Each engineer heard the same song over each monitor pair, and was asked to report on each speaker's characteristics.

Both of these test sessions proved consistent with each other and provided the basis for the conclusions set forth throughout the rest of this article.

It must be realized that by no means were all five of the representative monitors selected for evaluation directly comparable with one another. Some of the units were 2-way systems, while others were 3-way and contained up to four separate and active drivers. Three of the units contained a single low-frequency driver, while the remainder contained a dual-woofer configuration. Other considerations, such as portability, power handling capacity, and design purpose, also precluded a straight comparison. Therefore, unless directly compatible, results for each pair are given separately and not with respect to the other monitors.

We will discuss the speakers in order of size, no other reason is to be implied by their ordering.

Yamaha NS10M

The Yamaha NS10M is unarguably the most prevelant close-field monitor being

STOP PRESS:
Yamaha updates NS10M monitor

The MkII version of the 2-way monitor, unveiled at the recent NAMM show in Ahaheim, retains the white cone 7-inch LF driver of the original design, and adds a newly developed 1.4-inch dome tweeter. The unit's overall HF response has been tailored specifically for studio applications, according to Yamaha.

The original vertical design has also been replaced with a horizontal orientation for easy console placement and construction is said to be more rugged to withstand constant pro-audio use.

Auratone T-6

used today; nearly every studio in the United States has at least one pair. Measuring 15" x 8½" x7⅜, the NS10M was the smallest speaker involved in these tests.

Each monitor features two drivers: a 7-inch lightweight woofer, and a 1⅜-inch soft-dome tweeter crossed over at 2kHz with a 12dB per octave filter slope. The manufacturer's specified frequency response is 60Hz to 20kHz, with a power handling capacity of 25W continuous, and 50W maximum. In addition, the 8Ω nominal impedence presents few problems with amplifier-to-speaker interfacing.

While the NS10M cabinet construction is very solid indeed, it is the only sealed enclosure included in these test sessions. The company claims that a sealed enclosure produces the best transient bass response. Also, a sealed enclosure better controls cone excursions, making it harder to damage with high-level bass signals. As far as wiring is concerned, rear panel color-coded clips provide for quick and nonconfusing hook-up.

The results from both the in-use and blindfold tests were in complete agreement with one another. With respect to frequency response, the NS10M is, as most users have found, a "bright" speaker. In this we found an increased level of upper-mid and high frequencies with respect to the rest of the audio band. Also, many listeners perceived two resonances around 3kHz and 10kHz. This proved to be more problematic at high monitor levels, because the first sign of distortion occured around these frequencies. Fatigue due to the enhanced high frequencies could become a problem during long listening sessions, but did not seem to occur during the period of our evalutions which, admittedly, were short.

One other frequency-response anomaly involved the roll-off of bass energy. Instead of having a steep roll-off at low frequencies, the NS10M appears to exhibit a gradually decreasing slope that starts around the low-mid area. This presented some low-bass response problems down in the 60Hz-200Hz range even at loud volumes where, as the Fletcher-Munson curves show, the ear is more sensitive to such frequencies.

Stereo imaging was quite good, with no major weak spots anywhere in the stereo field. In our studio, best results were not achieved with the speakers upright but on their side. Laying down the speakers with the tweeter closer to the plane of the engineer's ears seemed to improve stereo imaging.

The unit's 12dB per octave crossover network works very well; no audible problems, such as crossover distortion or phase cancellations, were discovered around the 2kHz crossover point.

Practically speaking, the NS10M is a very rugged speaker which, combined with its small size and weight, makes them easy to carry from studio to studio.

Auratone T6

The Auratone T6 is a newer speaker that the NS10M. It obviously starts with the same basic design concepts, including compact size, two drivers, left-right symmetry and solid cabinet construction. It differs from the NS10M, however, in featuring a ported-cabinet design.

Each cabinet contains a 6½-inch woofer and a 1-inch soft-dome tweeter with ferrofluid voice-coil coolant, which helps prevent burnout due to high-level, high-frequency information. The T6 features a 12dB-18dB per octave crossover at 2.8kHz, and is designed to handle a maximum program level of 80W while presenting an 8Ω load to the amplifier.

The compact enclosure measures 14½" x 9½" x 10" and weighs 18lbs. The T6 can be wired in two different ways: either by bare wire, or using dual banana plugs.

The low-end response of this ported system is very precise and clear, and extends below 60Hz. The quoted frequency response extends from 60Hz to 20kHz, ±3dB. All of the upper and lower mid-range areas, including the crossover region, are crystal clear; no weak spots around the 2.8kHz crossover point were detected at all.

The T6 also has an impressive high-end response. Unlike the somewhat "peaky" response of the NS10M's tweeter, the T6's high-frequency driver is very even sounding.

As far as stereo imaging is concerned, the T6 performed very well. The image appeared to extend beyond the speaker's physical position, while monitor placement did not seem as critical as with the other monitors. Source localization was solid throughout the stereo field, and was presented with realistic, 3-dimensional depth. The phantom-center image was very stable.

As one might guess, the T6 left a good impression on all of the people involved in the tests. It performed well in every catergory, leaving no doubts as to its high quality. Although small in size, the T6 pack a hefty punch and can be played at surprisingly loud listening levels. Overall the T6 is an impressive speaker.

Tannoy NFM-8

The NFM-8 is a moderately sized speaker, with dimensions of 18" x 12" x 8" and an overall weight of 24.3lbs. It is the only close-field monitor reviewed here that utilizes a dual-concentric driver design. By placing the high-frequency driver in the center of the low-frequency speaker distortions created by the interactions of multiple wavefronts can be reduced, and thus more closely approximate the "ideal" sound point source.

The 8-inch ported design has a quoted frequency response of 57Hz to 20kHz, ±3dB. The crossover is a passive, 8-element, time-compensated design centered at 1.8kHz. The unit presents a nominal 8Ω (minimum 6Ω) input impedence, and has a quoted peak-power handling capacity of 120W. Rear-panel binding posts accept either bare wire or dual-banana plugs. A red terminal is labeled "high" and the black terminal

Tannoy NFM-8

"low;" a yellow terminal, when connected to the black terminal via a bridging strap, reduces the level to the high-frequency driver by 3dB.

While listening to the NFM-8, we noticed that low-bass frequencies were tight and exact, causing very little excursion of the 8-inch woofer. Yet somewhere in the lower mid-frequency range, a slight peak, which did cause quite a lot of woofer movement, was perceived by some listeners. This was attributed to mechanical resonances of the woofers and cabinetry.

The upper mid-frequency content (around 2kHz) seems to be much lower in amplitude than the rest of the speaker's output. One possible reason for this could be that the lower mid-range "resonance" causes those frequencies to sound louder in comparison, with the upper mid-range. Another possible explanation is that the NFM-8 exhibits some problems around its 1.8kHz crossover point. In addition, at louder monitoring levels, the tweeters tend to overload first, causing objectionable HF response.

In terms of source localization and stereo imaging, the NFM-8s proved to be the most accurate of all the units tested. Apparently, Tannoy's dual-concentric design does indeed act very much like a single point source; accurately defined stereo field produced by the NFM-8 allows for pinpoint localization. Also, the center image presented by a pair of speakers is extremely solid. The only downfall is that the stereo sound field does not seem to extend any further than the speakers themselves. Due to their highly directional design, placement of NFM-8s is more critical than with the other monitors; once placed correctly, however, they provide greatly detailed stereo information.

Calibration Standard Instruments MDM-4

The MDM-4 is the only 2-way, dual-woofer system to be included in our review. The speaker measures 19"x13"x9¾" and weighs 25lbs. Each unit features two, 6½-inch ported bass drivers connected in parallel, and a single 2¾-inch mid-high driver located between the woofers.

The manufacturer's quoted frequency response is 60Hz to 17kHz, ±3dB, and is matched between speaker pairs to within ±0.5dB. Power handling is rated as 40W continuous, and 160W instantaneous peak below 1kHz, and 15W continuous, 60W instantaneous peak above 1kHz. Distortion characteristics are quoted at less than 5% total harmonic or intermodulation distortion from 60Hz to

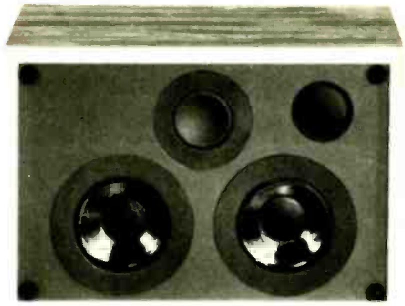
Calibration MDM-4

20kHz, and less than 1% THD or IM from 100Hz to 20kHz, measured at 94dB SPL at 1m. (Which seems to indicate that most of the distortion problems are occuring, as is usually the case, in the low-end between 60Hz and 100Hz.)

The crossover is an equalizer/filter-type centered at 1.5kHz. The MDM-4 presents an 8Ω nominal and a 5Ω minimum input impedance. Connections are made via bare wires or dual-banana sockets.

In the mid-range frequencies, the MDM-4s presented themselves quite well and were never overbearing. The bass response was also good. Although punchy and never muddy, even at higher volumes, it did seem to roll off a little early, leaving the lowest few octaves behind.

The MDM-4 has good stereo imaging, which probably results from the less than 100ms time offset between the LF and HF drivers. Source localization, while good at times, did tend to be slightly undefined, however.



One is left with the feeling that these monitors are well constructed both sonically and structurally.

Westlake Audio BBSM-6

The largest close-field monitor reviewed here, the BBSM-6 measures 22"x13"x10¾" and weighs 43lbs. Because of its weight, you need to be careful when monitoring these monitors on the console meterbridge.

The BBSM-6 is a 3-way monitor that features two 6-inch active woofers wired in parallel in a ported enclosure, a 3¼-inch midrange unit in a separate sealed enclosure, and 1-inch dome tweeter. The midrange driver and the tweeter are located symmetrically between the woofers. All three drivers are mounted in a solid and well built cabinet.

The two crossover points are 600Hz and 6kHz, with 24dB per octave minimum slopes, and the speaker features a 4Ω nominal and a 2Ω minimum impedance. Quoted power ratings are 60W below 600Hz, 50W from 600Hz to 6kHz and 30W above 6kHz. Frequency response is a quoted 70Hz to 18kHz, ±3dB measured on axis and suspended.

Connections are made via a 4-terminal barrier strip located on the back panel. Of the four positions on the terminal strip, one has a red marking, another a green marking and the other two are left blank. The red terminal is "high" and the green one "low."

As we began our listening tests, distortion problems at high monitor levels became immediately apparent. To our relief, the problem was not with the speaker itself but with the power amplifier, which was incapable of accurately delivering the large currents required by a 2Ω load. In order to rectify the problem, a more suitable power amp was used.

Frequency responses of the BBSM-6 is very good through its range. High frequencies are smooth all the way up, and exhibit no major peaks. Similarly, middle frequencies are unobtrusive and bridge the gap between the woofers and the tweeter with no problem. In comparison to all of the monitors reviewed, the BBSM-6s excel in their low-frequency response. Although the quoted ±3dB range extends to 70Hz, the monitors appear to reproduce frequencies well below this without any problem.

Stereo imaging and source localization at first seemed to be a little undefined but, once we moved the speakers back another two feet, they seemed to lock in. A possible explanation for this is that, because the BBSM-6s each have four separate drivers, it is impossible for the speakers to sound like a point source at close distances; increasing the distance helps reduce the apparent size of the source at the listener's position.

The two 24dB per octave crossovers are seemingly transparent around their 600Hz and 6kHz center frequencies, and produce no audible problems.

The BBSM-6 was capable of producing the highest monitor levels among the units tested, and is more than adequate for even the most demanding of listening purposes. (During some of our tests, the unit drew much more than 3A, easily blowing our in-line protection fuses. Because of this high current demand, Westlake has provided a printed label on the rear panel warning of possible amplifier damage.)

Westlake BBSM-6

Technical Specifications

Yamaha NS10M

Frequency response: *60Hz to 20kHz.*
Power handling; *25W continuous; 50W maximum*
Impedance: *8Ω (nominal).*
Drivers: *7-inch LF and 1 3/8-inch tweeter*
Crossover frequency: *2kHz; 12dB per octave*
Dimensions: *15" x 8½" x 3/8"*
Circle 120 on Rapid Facts Card

Auratone T6

Frequency response: *60Hz to 20kHz, ±3dB.*
Power handling: *80W, maximum*
Impedance: *8Ω*
Drivers: *6½-inch LF and 1-inch tweeter*
Crossover frequency: *2.8kHz; 12dB-18dB per octave*
Dimensions: *14½" x 9½" x 10"*
Weight: *18lbs*
Circle 121 on Rapid Facts Card

Tannoy NFM-8

Frequency response: *57Hz to 20Hz*
Power handling: *120W*
Impedance: *8Ω nominal, 6Ω minimum*
Drivers: *8-inch dual-concentric*
Crossover frequency; *1.8kHz*
Dimensions: *18" x 12" x 8"*
Weight *24.3lbs.*
Circle 122 on Rapid Facts Card

Calibration MDM-4

Frequency response: *60Hz to 17kHz, ±3dB*
Power handling: *40W continuous, 160W peak (below 1kHz); 5W continuous, 60W peak (above 1kHz)*
Impedance: *8Ω nominal, 5Ω minimum*
Drivers: *two, 6½-inch LF, 2¾-inch mid/high*
Crossover frequency: *1.5kHz*
Dimensions: *19" x 13" x 9¾"*
Weight: *25lbs*
Circle 123 on Rapid Facts Card

Westlake BBSM-6

Frequency response: *70Hz to 18kHz, ±3dB*
Power handling: *60W below 600Hz; 50W 600Hz to 6kHz; 30W above 6kHz*
Impedance: *4Ω nominal, 2Ω minimum*
Drivers: *two 6-inch LF, 3¾-inch and 1-inch tweeter*
Crossover frequency: *600Hz and 6kHz, 24dB per octave*
Dimensions: *22" x 13" x 10¾"*
Weight: *43lbs*
Circle 124 on Rapid Facts Card.

The BBSM-6 is certainly one of the best built close-field monitors made today. And, without a doubt, this attention to detail has paid off in that it is one of the cleanest sounding monitors around.

Conclusion

These evaluation sessions reviewed five popular close-field monitors that are in use today. In fairness, however, and for the sake of completeness, several additional comments must be included. The results of these tests are not direct comparisons; instead, they represent separate reviews covering such topics as size and weight, durability, power handling, output level, stereo imaging and, of course, how the speaker sounds.

Also, the speakers reviewed here represent ony one particular model from each manufacturer. Each company provides several other different designs of close-field monitors; a review of the same type of system design from each company (for exampe, 2-way or 3-way) would be particularly interesting.

I would like to thank all the companies involved with this assessment, not only for sending the speakers so promptly, but for their willingness to help with any questions or problems throughout the tests. Any one of the units would serve as an excellent close-field monitor. R-E/P

Circle (46) on Rapid Facts Card

Studio Update

Northeast

Shelton Leigh Palmer & Co. (New York) has expanded as a music production house to include a new recording studio located in midtown Manhattan.

Shelly Palmer, president, said, "The idea for the facility started out with a plan for a large control room and a tiny studio. As we were building it, I said 'Let's make the studio big enough for more players,' We just kept stretching what we wanted the room to do, and ended up building a world class production facility."

Studio A features a **New England Digital Synclavier, E-mu Emulator II** digital sampling keyboard and a **MCI JH-538/38C** production console with 38 inputs. Other hardware includes an **MCI JH-24** analog, 24-track, **Otari** ½-inch 4-track and an **Adams-Smith** 2600 system with edit controller. Monitors include **Yamaha NS-10M** and **Auratones** powered by **Hafler** amps.

Studio B is designed for low-budget voiceovers, video post-production, radio, industrial and multimedia audio production projects.

The studios are designed for the production of TV and film soundtracks, radio IDs and network promos, sports programs, video game sounds, industrial films and multimedia shows. *19 W. 36th St., New York, NY 10018; 212-714-1710.*

Airwaves Audio Productions (Manchester, NH) has expanded and moved to a new location. Previously headquartered with Group Five Communications, the facility will now be based at the former Kevin Tracy Productions studio.

Joel Schwelling, manager and copartner of the company along with technical director, **Gerry Putnam,** says that the New Hampshire advertising community is accustomed to having quality radio and commercial music produced locally. "We very much want to see that trend continue," Schwelling confided, "but we're extending our capabilities to include a full range of expanded audio services for advertising, business and the media."

Radio commercials will continue to be produced at the studio's B room, equipped with a new **Foxtex B-16** 16-track and an automated **Allen & Heath CMC24** console. Plans are underway to improve the room's sound and sync capabilities for video and film post-production.

The A room, a 24-track facility, will be used for jingles and other music production. Hardware includes a **Sound Workshop series 30** console, **Otari MTR-90** 24-track and a **Sony JH-110 2-track.** *342 Lincoln St., Manchester, NH 03103; 603-627-2774.*

Highland Studios (Delmont, PA) has added **J. Michael Jacques** as account executive. Jacques attended the University of Tennessee and is a native of Nashville. He will coordinate the sales and marketing efforts for the studio. *5 W. Pittsburgh St., Delmont, PA 15626; 412-468-6661.*

Studio Update

GRC Studios (Baltimore, MD) has upgraded to a 24-track facility with the addition of a **3M M-79** recorder, **SECK** mixing board, **Yamaha SPX-90** and a computer MIDI sequencer.

Robert Friedman, president, said, "Locking the sequencer to the 24-track gives us a total of 23 analog and 16 MIDI tracks." *1137 Fillmore St., Baltimore, MD 21218; 301-889-4228.*

NFL Films Video (Mt. Laurel, NJ) has added **Jerry Mahler** as chief audio engineer. Mahler was formerly with Caribou Ranch Recording Studios in Nederland, CO, and worked with Dan Fogelberg, U2, Amy Grant and Al DiMeola. At NFL Films Video, he will supervise the audio staff. *300 Fellowship Rd., Mt. Laurel, NJ 08504; 609-778-1600.*

Power Station (New York) has added two **Otari DTR-900 PD-format** digtal 32 tracks. *441 W. 53rd St., New York, NY 10019; 212-246-2900.*

Southeast

Protolog (St. Petersburg, FL) has opened a new facility in St. Petersburg. The recording studio is geared toward time code MIDI controlled synthesis, digital sampling and special effects. Hardware includes a **Trident 65** console, **Tascan 580 B** with noise reduction, **E-mu** keyboard and a **Macintosh** with **Digidesign** software. *P.O. Box 41438, St. Petersburg, FL; 813-345-8836.*

Midilab (Atlanta, GA) has upgraded its facilities to include a **Version 4.0 Kurzweil 250** digital sampling system and an **Apple Macintosh Plus** with **Digidesign Soundesigner** software. The K250 allows on-line access to more than 96 instrument voices as well as 341 keyboard setups. It also features 40 seconds of sound sampling at 50kHz. The K250 can also be synchronized to time code subframe accuracy.

Bill Anderson, Midilab producer said, "In addition to creating virtually any sound, the artist can hear that sound mixed and processed with other parts of the composition before committing it to a master recording." *106 N. Avondale Rd., Avondale Estates, GA 30002; 292-210-3404.*

Midwest

White Room Productions (Indianapolis, IN) has opened a new facility that includes the following equipment: **Tascam 80-8** 8-track, **Otari MX-5050 BII** 2-track and a **Tascam model 32** 2-track.

The studio features a **Panasonic Ramsa WR-8816** console. Processing equipment includes a **UREI 1176 LN** limiter, **Yamaha SPX-90** digital processor, **Orban 622B** stereo parametric EQ, **Aphex type B Aural Exciter** and **Omni-Craft GT-4** noise gates. Monitors include **JBL 4412** and **Auratone 5-C** units. *7168 N. Graham Rd., Suite 140, Indianapolis, IN 46250; 316-841-0002.*

Studio Update

Royal Recorder's overhead video monitors in background display stored settings and the 80-input Solid State Logic console with 64 mono channels, eight stereo channels and primary/total recall.

Designed by Lakeside Associates, **Royal Recorders** (Lake Geneva, WI) has recently upgraded with an 80-input **Solid State Logic SL 4072E** console, equipped with 64 mono channels, 8 stereo channels and **Primary Computer** plus **Total Recall** automation.

Recording duties are handled by twin **Studer A800** 24-tracks, a **Mitsubishi X-850** PD-format digital 32-track and X-80 2-track, plus two **Studer A-80** ½-¼-inch 2-tracks. To provide access to up to 80 independent tracks, the two A-800s and X-850 can be locked together with an **Adams-Smith model 2600** synchronizer.

Outboard gear includes echo, reverb and delay systems, a well as limiters, noise gates, parametric EQs, choruses, and flangers. A **Hammond B-3** organ with **Leslie Cabinet, Yamaha C7** grand piano, **Roland** and **Yamaha** keyboards, **Sonor** drum kit, **Emulator II** and a **Memory Moog** are also available.

Situated among the hills of southern Wisconsin's Americana Lake Geneva Resort, the new facility is 90 minutes away from downtown Chicago. *Americana Resort, Highway 50, Lake Geneva, WI 53147; 414-248-9100.*

Southern California

Lion Share (Los Angeles) has been purchased by **Terry Williams**, former studio director, **Jay Antista,** director of engineering and **Donn Chickering.** The facility was previously owned by Kenny Rogers.

"We have a great deal of exciting new plans for 1987," Antista says. "We have long-range plans that include a new studio, new automation systems and plans that will enable Lion Share to grow with technology and inevitably to the realization of a complete digital recording facility." *8255 Beverly Blvd., Los Angeles, CA 90048; 213-658-5990.*

Le Mobile, (North Hollywood) has taken delivery of a **Necam 96** console automation and 24 tracks of Dolby SR noise reduction.

Compared to the existing **NECAM II** system, which it now replaces on the existing **Neve 8058** console, **NECAM 96** is said to provide full color status displays on a video monitor and faster operation. The new system, like **NECAM II**, features servo-activated faders and touch-sensitive grouping. *11131 Weddington St., North Hollywood, CA 91601; 818-506-8481.*

Studio Update

Record Plant's new studio "L" super-synth room.

Record Plant Scoring (Los Angeles) has opened a new facility dedicated to synthesizer recording.

Studio L features a **Trident Series 65** console, **New England Digital Synclavier** and **Fairlight CMI Series III** digital synthesizers.

The new room is MIDI-controlled through a **StratoMac IV**, described as an enhanced **Apple Macintosh** developed by Julian Music Systems in conjunction with Apple Computers. Further upgrade plans for the room include the addition of Julian Music system's Rackintosh rack-mountable version of the StratoMac designed for studio applications and traveling musicians.

Joel Moss, executive director of Record Plant Scoring, says, "In one room we have all the tools for high-end synthesizer work. Many composers have retreated to their home studios and there has been a sacrifice in quality. Film producers have found that getting five top synth players and their equipment into a studio can end up costing more than hiring a full orchestra. We are providing a cost-effective alternative that fills a basic need in the industry today." *1032 N. Sycamore, Los Angeles, CA 90038; 653-0240.* R·E/P

Send Studio Update announcements to: Sarah Stephenson, Recording Engineer/Producer, Intertec Publishing, 9221 Quivira Road, Overland Park, KS 66215.

New Products

Yamaha DX-7 MkII FM synthesizers

The DX7IID (the "D" standing for dual-FM tone generators) and DX7IIFD (Dual-FM and 3½-inch floppy disk drive) feature dual six-operator, 32-algorithm FM tone generators, as utilized in the DX1 and DX5.

Dual-FM allows each keyboard to function as a 16-note (single mode), two independent eight-note (split), or two layered eight-note voices (dual) polyphonic synthesizer.

A random pitch shift feature allows the pitch of each note to be slightly and randomly detuned, to simulate the "warm" sound of several acoustic instruments playing the same note at slightly different pitches.

Although the DX7II models use a larger RAM cartridge with twice the capacity, cartridges for the DX7 will also work with the new units. The ADP1 cartridge adapter allows a RAM1 or other DX7 cartridge to be plugged into the enhanced model.

Voice memory is doubled to 64 internal and 64 external (RAM). The DX7IIFD also has a built-in 3½-inch Mbyte floppy disc drive to store both synthesizer data and any MIDI data from sequencers, drum machines and other synthesizers.

The DX7II series also allows the tuning of each key to be programmed. The Micro Tune memory has 11 presets, including Mean Tone, Werckmeister, and Pythagorean and two user programmable memories.

Also featured: left and right outputs with panning to determine the position of voices according to key touch, LFO, key velocity, or which part of the keyboard is being played; and a 40-character, 2-line backlit LCD and two alpha-numeric LEDs; inputs for breath controller and headphones.

Circle (165) on Rapid Facts Card

New Products

Jan-Al Pro-Rack rack system

The new EIA rack system can be assembled with a screwdriver and expanded by replacing the side panels.

The rack is available with 7, 13 and 20 spaces, and constructed of ¾-inch multi-ply hardwood with a scratch resistant, water resistant finish.

Standard features include steel rack rail with a 18 inch rackable depth.

Accessories include casters, rear rack rail, mounting hardware and a Rhino (ATA) flight case.

Circle (177) on Rapid Facts Card

Studio Magnetics SML1216 recorder

The ½-inch 16-track recorder manufactured by Studio Magnetics Ltd. of England is now available from their U.S. distributor, Rock Studio supply.

The recorder may be wired for either 15 or 30 ips and features a 50% vari-speed built-in noise reduction with individual bypass, solid-state switching, electronic braking, edit and cue mode, +4 operating levels.

The company claims a frequency response of 40Hz-18kHz, ±3dB, less than 0.7% distortion and signal to noise ratio of −80dB with proprietary noise reduction system.

Circle (181) on Rapid Facts Card

Kurzweil 250 rack-mount module

The rack-mount module contains all the sounds and control features of the K250 synthesizer, including user sampling. The K250 RMX is operated via any controller equipped with MIDI.

The RMX is designed to appeal to studios already equipped with MIDI controller or synthesizer.

Like the K250, the module features a total of 36 resident instrument voices as well as 98 keyboard setups stored in the ROM.

Four sound blocks that store additional instrument voices are also available, including woodwinds, vibes, electric bass, percussion, Minimoog sounds, electric piano and electric guitar.

Circle (168) on Rapid Facts Card

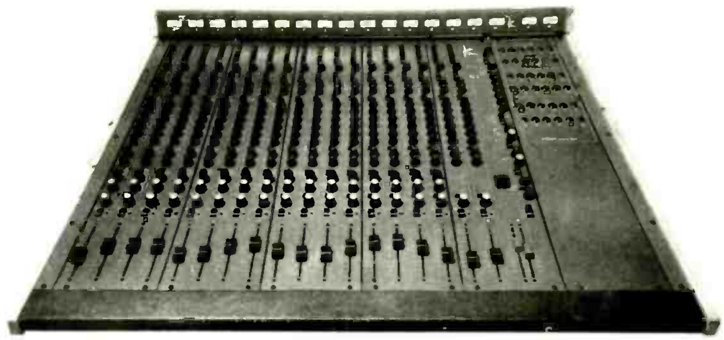
ACES B1816 production console

The B1816 in-line console, manufactured in the United Kingdom by ACES, is now available from the newly appointed U.S. distributor, Rock Studio Supply. The new mixer features 18 inputs (expandable to 30), 16 subgroup outputs, 16-track monitoring and stereo master.

Inputs feature phase reverse, 48V phantom power, six auxiliary sends, 4-band EQ with sweepable mids, solo and mute.

The master module includes a 3-frequency oscillator, headphone monitor, auxiliary send masters, two auxiliary returns with level, pan and solo, talkback function, 2-track return, monitor mute and mono sum and a 36-point expandable patchbay.

Circle (166) on Rapid Facts Card



New Products

Korg DRV-3000 MIDI-capable effects processor

The new unit features a wireless remote control processing and 16 effects programs including five reverbs, two early reflections, two echoes, auto pan, flange, chorus, ensemble, two pitch shift and a parametric EQ.

The factory presets may be edited by the user with the option to recall them from internal ROM at any time and can be set up as a dual effect, using a combination of two effects in series or parallel connection.

DRV-3000 programs may be selected and edited from the wireless remote control and accessed and controlled via MIDI.

Signal frequency response of effects output is a quoted 20Hz-20kHz with a 94dB dynamic range.

Circle (176) on Rapid Facts Card

Audio Kinetics Eclipse Off-Line

Described as an extension to the existing Eclipse audio editor and machine controller, the new accessory uses an IBM PC (or compatible) computer with an AK-IBM RS232 interface. The accessory provides the following features:

- Off-line preparation: 100 new loops can be loaded or existing loops edited onto floppy disk, before entering the edit suite. Additionally, events and Q. keys may also be pre-programmed.
- On-line download: whereby at the end of the session all generated loops and events may be down-loaded to disk, allowing rapid recall at any subsequent session, minimizing on-line time.
- Q. key storage, whereby personal sequences of key data can be used in other studios.
- Hard copy, whereby the loop information comprising an edit decision list, together with events and Q. key data, may be printed out on any IBM-compatible printer.

Circle (167) on Rapid Facts Card

Yamaha RX-5 digital rhythm programmer

Each of the unit's 64 internal sounds may be edited for tuning, envelope, gate time, voice level, pitch bend, loop, or reverse.

Each individual drum stroke can also be edited for pitch, level attack, decay and reverse.

The 64 voices initially available include standard drum and cymbal sounds, standard and Latin percussion sounds, DX voices and three human voices. Voices may be accessed from internal RAM and ROM, or an external Waveform Data Cartridge.

All voices may be played from a velocity sensitive MIDI keyboard or sequencer, with each voice assigned to a different key and/or MIDI channel.

The expanded internal memory holds three percussion sets of 24 rhythm voices, each with 100 patterns, 20 songs, and three song chains. This data can also be stored on RAM4 cartridges, cassette tape, or to an outboard disk drive. The unit also has the ability to title songs and song chains in memory.

Other features include tempo and volume change commands, autolocate function within songs, and sync to MIDI or non-MIDI products and tape with FSK sync.

Circle (179) on Rapid Facts Card

New Products

Audio Technica AT4462 portable mixer

The AT4462 features two mono inputs that can be panned left or right and two conventional stereo inputs. All inputs and the two 600Ω outputs are transformer coupled at microphone or line level.

Pre-fader cue is provided for each input. An interface unit allows program audio or other sources to be delivered as a monitor feed to the talent over conventional microphone cables connected to input channels 1 and 2.

Other features include 12V phantom power with provision for A-B power modification, VU meters, stereo/mono output switching, slate tone and internal slate mic, a 3-frequency oscillator, 2-color LEDs to indicate potential overload or limiter operation and a stereo limiter that may be switched to provide individual channel limiting.

Circle (184) on Rapid Facts Card

Roland MKS-50 synthesizer module

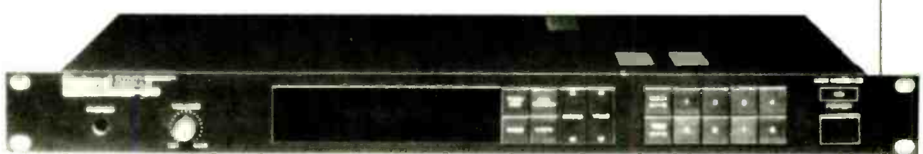
The new unit provides MIDI users with 128 patch memories and 16 chord memories. Each patch memory consists of one of the 128 user programmable tones along with a wide range of function including upper and low range limits, modulation sensitivity and key shift.

The MKS-50 offers six DCOs, six LFOs, six VCF/VCAs, a high pass filter, chorus and six 4-stage envelopes with time and level controls for each stage.

The unit provides level and rate controls for attack, sustain, decay and release functions to simulate sounds such as plucked strings, xylophone, jazz guitar and synth drums to achieve a new realism, the company says.

The MKS-50 can transmit and receive patches and tones using MIDI system exclusive messages. This allows single patches and tones to be sent and received as well as bulk dumping and loading into an extended sequencer or synthesizers.

Circle (182) on Rapid Facts Card



New Products

Studer A812 recorder

The new transport features three servo-controlled motors and handles reel sizes up to 12½ inches. Four tape speeds are standard and are front-panel selectable.

The software library of over 40 functions includes tape dump, zero locate, rollback and start locate.

Fast wind speeds are also programmable and varispeed operation may be specified in ips or percent deviation from nominal tape speed.

Other features include thumbwheel shuttle control, aluminum splicing block, microprocessor control of all audio alignment parameters, phase compensated audio electronics and choice of transformer or transformerless inputs and outputs.

Circle (162) on Rapid Facts Card

Paso series of mic accessories

Floor stand models include the B-10B light duty tripod stand and the B44B heavy duty stand with collapsable die cast base. The B44B uses synthetic rubber "feet" to isolate the mic stand from vibrations and uses a nylon clutch to adjust the height.

The A-13B boom uses a "quick set" clamp that allows spontaneous adjustments. It can be used with either the B-10B or B44B floor stand.

Another desk and floor stand, the B-16B has an adjustable height between 16 and 24 inches. A 51b die cast wide stance base prevents tipping and a felt cover prevents scratching on tables when used as a desk stand.

The GNK15B noiseless black gooseneck uses non-reflective black annodized finish that eliminates glare.

Circle (186) on Rapid Facts Card

New Products

Yamaha PM1800 production console

Available in four configurations of 16, 24, 32 or 40 input channels, and derived from the PM3000 design, the new console is said to feature improved input circuitry; four stereo auxiliary returns; eight group mixing buses; master stereo bus; an 8x4 mix matrix configuration; eight master mute groups; and cue and solo capabilities.

All auxiliary and group buses may be operated independently, resulting in a total of 14 discrete buses. Additionally, a total of 20 mixes are effectively available when using the stereo bus, and by re-setting the mix matrix internal jumpers.

Inputs are differentially balanced, equipped with a 3-position attenuation pad and a continuously variable gain trim control.

Each input channel includes a 4-band sweep EQ, and a 12dB/octave high pass filter with a −3dB cutoff adjustable from 20Hz to 400Hz.

Also new is a master mute function: each channel has eight mute assign switches that permit the on/off function to be remotely controlled by the eight master mute group switches.

Cue and solo capability includes switches on every input channel, and a cue switch on every master auxiliary send, group outputs, stereo master outputs and the auxiliary returns.

Circle (170) on Rapid Facts Card

Ross CN3201 electronic crossover

The unit features two separate 3-way channels for stereo tri-amping and, by patching the two channels, allows use as a 5-way mono electronic crossover.

Each channel features separate low, high and mid output jacks with phase inverters and a level control for each section. Signal to noise ratio is a quoted −95dB with a total harmonic distortion of 0.005%.

Circle (161) on Rapid Facts Card

Community Light & Sound M4 driver accessories

The SH2064M is a hyper-exponential flare horn providing full driver loading down to 270Hz and with an M4 driver, and can be packaged in a 22½-inch deep enclosure. The SH2064M horn is constructed of balsa-reinforced, hand-laminated fiberglass.

Designed to occupy a minimum amount of height in stacked columnar arrays, the SH1894M horn features the same construction as the SH2064M, and is designed for touring sound or fixed installations.

It is said to be capable of properly loading an M4 for full power operation down to 300Hz, while a vertically stacked pair will go as low as 250Hz.

Circle (172) on Rapid Facts Card

Carver PM-350/PM-175 power amplifiers

Using the company's magnetic field power amplifier technology, the new

For additional advertiser information, use the Rapid Facts Card in the back of this issue.

New Products

unit features slow startup, input muting during turn-on and protection mode, the ability to drive a 70V line and the capacity to provide onboard signal processing via inboard with plug-in modules.

The PM-350 is 350W per channel into 8Ω and the PM-175 175W into 8Ω. The amps are bridgeable and feature level controls, phono jacks plus XLR input connectors and clipping indicators.

Both amps can be mounted in a 19-inch rack.

Circle (174) on Rapid Facts Card

Panasonic S-Series miniature microphones

The RAMSA WM-S1 features a quoted frequency response of 50Hz to 18kHz and can handle a maximum sound pressure level of 148dB, 1% THD. The WM-S2 covers frequencies between 120Hz and 15kHz and can handle a maximum SPL of 138dB.

The WM-S5 boasts a maximum SPL capability of 158dB and frequency range of 70Hz to 16kHz. The WM-S10 headset microphone has a frequency response of 120Hz-15kHz with a maximum SPL capacity of 138dB, the company claims.

All four microphones can be operated on phantom power and the WM-S2 and WM-S10 can be run from internal batteries.

Circle (178) on Rapid Facts Card

Tannoy SGM Super Gold studio monitors

The new dual concentric monitor series uses a form of crossover design that improves the distribution of high peak currents, the company claims.

The result is a range of loudspeakers that bear little external change from the existing series, however, printed circuit boards and layouts and contacts between potential dissimilar metals have been eliminated.

Circle (175) on Rapid Facts Card

Roland DEP-5 digital effects processor

Combining reverb, chorus, stereo panning, parametric equalizer and digital delay, the new processor has a quoted frequency range of 30Hz to 12kHz and a dynamic range of over 80dB. The unit also features a quoted harmonic distortion of 0.03% or less.

The reverb function produces an RT_{60} of up to 90 seconds in both reverb and on-linear modes. Eleven rooms, seven halls, two plates and two special rooms are available. Pre-delay time can be set from zero to 500ms in reverb or non-linear mode. In non-linear mode, reverse reverb is possible with RT_{60} of −0.9 seconds to 99 seconds.

The EQ section offers three bands centered at 100Hz, 300Hz to 12kHz (sweep) and 10kHz. A variable Q (bandwidth) range is also available for the mid range from 0.2 to 9.

Circle (163) on Rapid Facts Card R·E/P

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PRODUCT SPECIALIST—Otari Corporation has an immediate opening for an energetic self starter with a technical electronics background, excellent communication and training skills, and audio with video or broadcast production experience, who would enjoy working in a dynamic, directed "small company" environment. The job will require continued immersion in the hardware and software product offerings of Otari Corporation and other manufacturers in the professional production products marketplace, with the goal of providing up-to-date technical sales support for Otari dealers, independent representatives and staff. The job will include attendance at industry trade shows, the ongoing production of Otari technical sales memos and reference manuals, and daily contact with current and potential users of Otari and related products. Direct resume to: David Roudebush, National Sales Manager, Otari Corporation, 2 Davis Drive, Belmont, California 94002. 3-87-1t

MISCELLANEOUS

FREE CATALOG: Institute of Audio-Video Engineering, 1831 Hyperion, (RP), Hollywood, CA 90027. Recording School. (800) 551-8877 or (213) 666-2380. Approved for International Students. Employers: Call for Employees/Interns. 3-87-6t

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