

APM



BROADCAST ONE

COMPACT
disc
DIGITAL AUDIO

PRODUCTION LIBRARY

APM... A world leader in licensing music for film, commercial and audio-visual production proudly introduces the *BROADCAST ONE* compact disc production library.

BROADCAST ONE will make your station sparkle. Your production will have a brilliance that your advertisers and audience will hear and love. Your station will sound better... and you'll sell more.

The music in *BROADCAST ONE* has been selected from the award-winning Bruton & KPM libraries whose credits include films (*The Color Purple*, *Wall Street*, *The Right Stuff* and *Ironweed*) and commercials (Oldsmobile, Mazda, Mercedes, and Harvey's Bristol

Creme). Some of the finest writers and studio musicians in the world contributed to *BROADCAST ONE*.

APM's commitment goes beyond the quality, variety, and convenience of *BROADCAST ONE*. APM is the first company to offer on-going service via its unique APM "Production Hotline." To solve your tough production problems the APM "Production Hotline" allows you to license additional music from the vast KPM & Bruton libraries and have it delivered overnight.

For a compact disc sampler of *BROADCAST ONE*, please call us toll free in Los Angeles 800.543.4276 (within California call 213.461.3211).

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Circle (1) on Rapid Facts Card

www.americanradiohistory.com

THE IEQ WITH Smartcurve™

Not Just Another Programmable Equalizer With MIDI™

The IEQ with Smartcurve™ is a programmable, high performance graphic equalizer that includes a video output. For those who wish to enjoy the video output of the IEQ, ART makes the IEQ Video Monitor.* Smartcurve™, proprietary software developed by ART gives you instant visual frequency response as easy as the push of a button.

The IEQ Family consists of both the 1/2 octave and the 1/3 octave graphic equalizers. Both types come in two varieties, Controllers and Satellites. A Controller is a self-contained programmable intelligent graphic equalizer capable of controlling 15 satellites at once. Satellites are exactly the same except the front panel controls are eliminated.

Model Specifications:

Controller & Satellite
 128 battery backed presets
 MIDI
 Frequency Response
 20Hz-20kHz \pm 0.5dB
 THD \leq .009% @ 1kHz, 0dBm typical
 Dynamic Range \geq 100dB typical
 Balanced inputs and outputs

Video Monitor Features

19" rack mountable
 NTSC compatible monochrome monitor
 4 Selectable inputs
 Standard RCA jacks for easy connections

It Has To Be A Work Of...

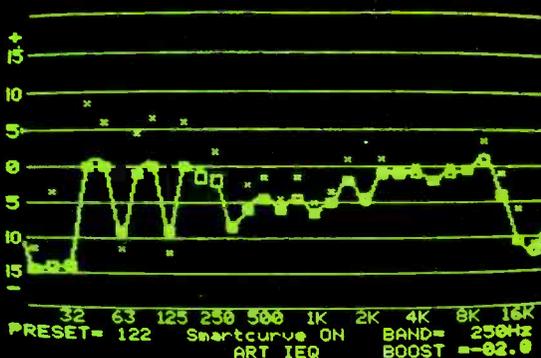
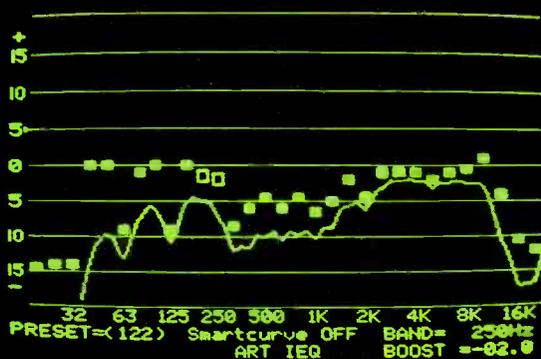
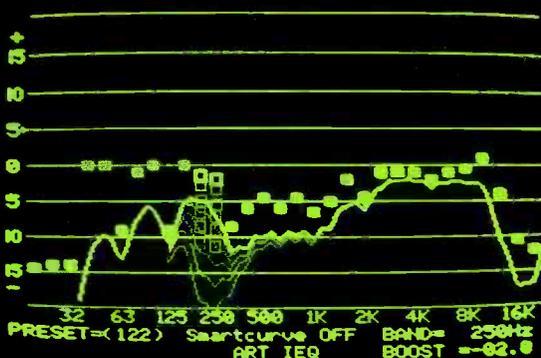


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Circle (4) on Rapid Facts Card



1 See the Sound

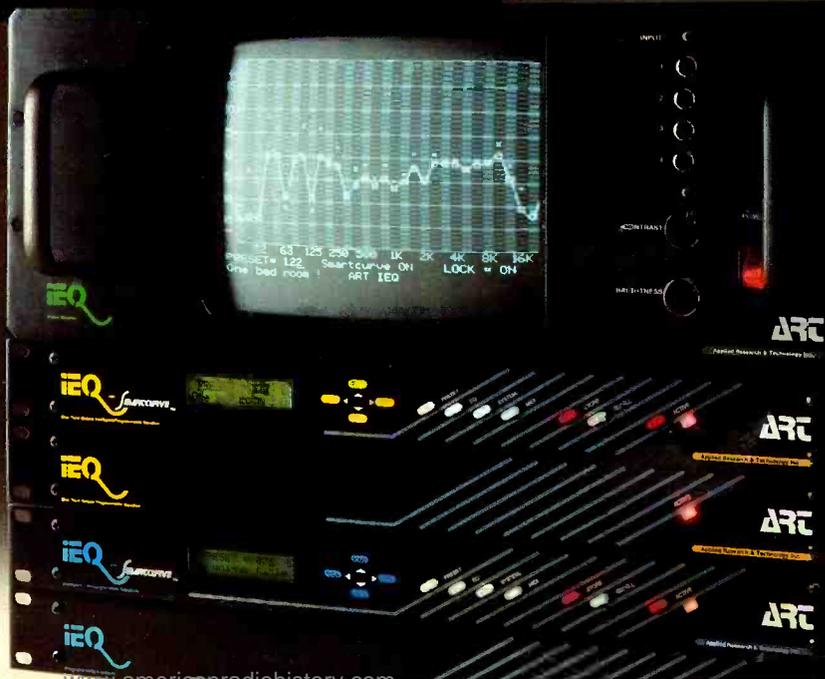
This is a video output of the IEQ as the unit is being adjusted. The sliders can be moved \pm 15dB in 1/2dB steps to get the exact response you need. With the simple push of a button, complex equalization can be done in seconds with incredible accuracy.

2 Hear the Sound

The power of the IEQ readily becomes apparent as the video display plots the frequency response due to the slider settings. The IEQ offers high quality constant "Q" equalization. The video graphic display shows the correlation between the sliders and the frequency response.

3 Perfect Sound

Turn Smartcurve™ on and perfect equalization is at your fingertips. Note the difference between the second frame. (Smartcurve™ OFF) and this frame. The position of the sliders represent the actual frequency response of the EQ. Interaction between bands is virtually eliminated. Incredible! Just think, now when you adjust the EQ you get exactly what you need. The "perfect" EQ? Let your eyes and ears decide.



Commercial Production

The Engineering "Hot Seat:" Commercial Music Production

Every aspect of a high-speed, high-pressure commercial production session has to flow through the "media" engineer.
By Michael Fay 20

Audio Post Production

An insider's look at the end of the line: contrasting the "film mix" and "sweetening session" production techniques.
By Brian Lee 30

The Speed of Sound

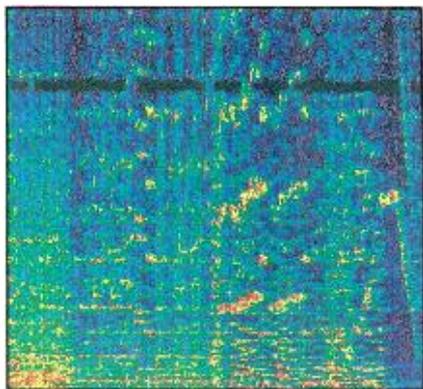
New analysis shows that tight control of temperature and humidity must accompany the popular trend of splitting microseconds when time correcting sound systems.
By Dennis Bohn 38



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Equipment Rental: Fantasy and Reality

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By Tom Behrens 52



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A Bits Eye View of Digital Signal Processing

Signal processing is an inexact science that requires the use of various techniques that often achieve an only adequate response.
By Richard C. Cabot, PhD., P.E. 56

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Winter NAMM Replay

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By Laurel Cash 68

The Basics of Music Libraries

The hows and whys of music libraries, from the owner/composer of one such company.
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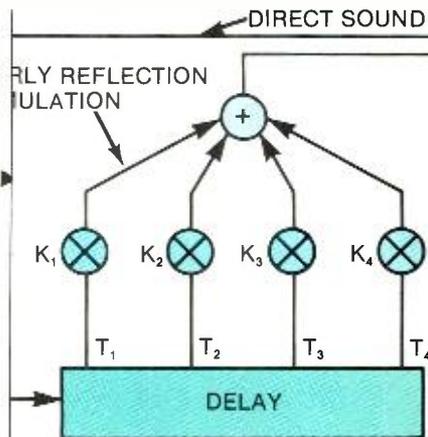
The composer/producer of IDs for the Wave format created not jingles, but signature songs.
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HLC
A self-described "utilitarian" facility, HLC's commercial production success is borne out by its client list.
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RE/P Special Report: NBS Rejects Copycode

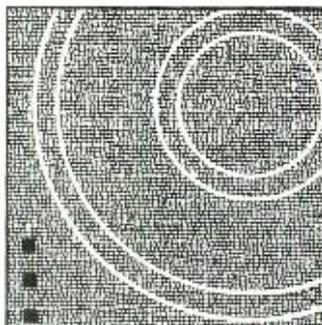
Copycode fails to achieve its purpose.
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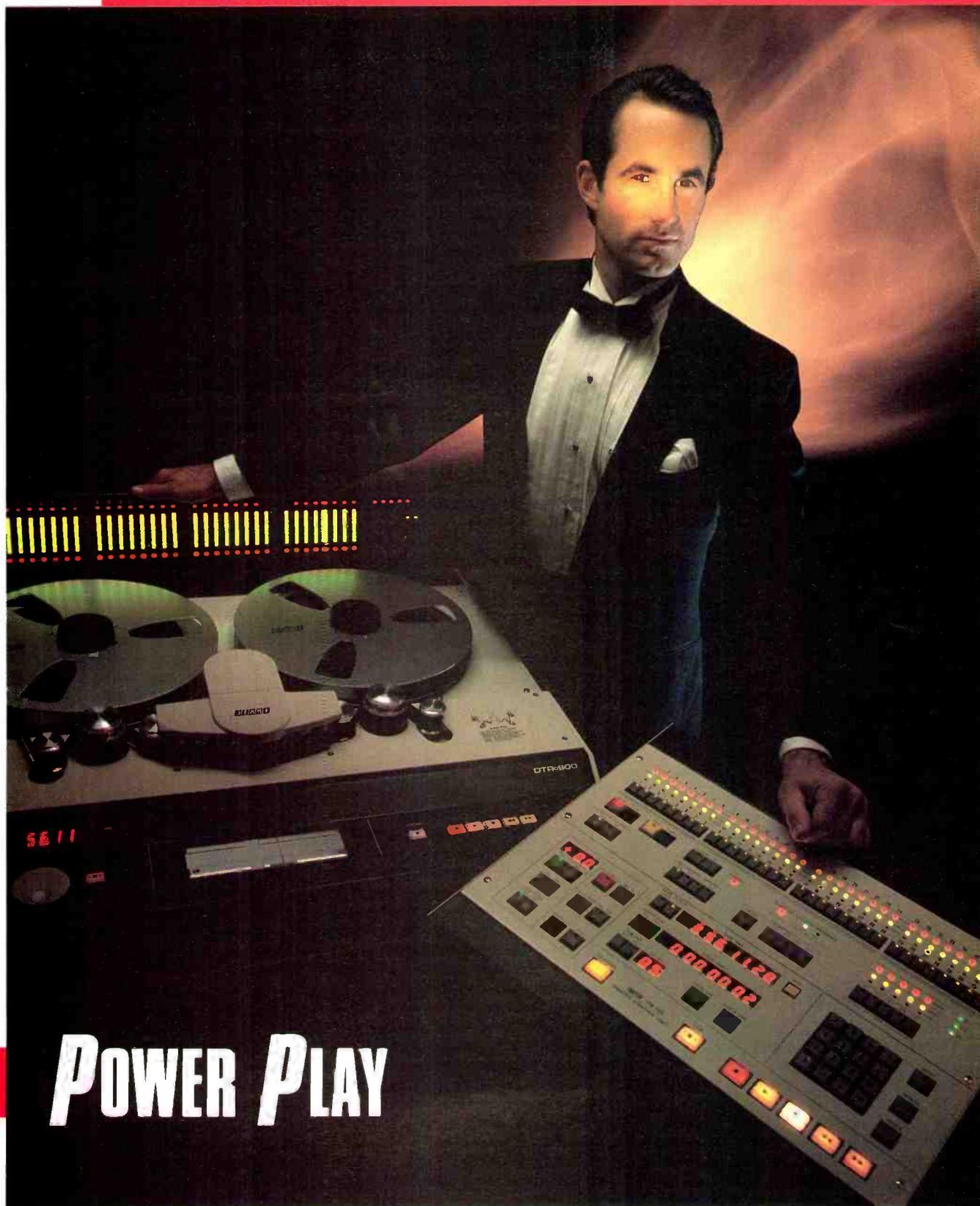
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Circle (5) on Rapid Facts Card

Think... Audio

Rarely does a month go by that I don't get a letter from an aspiring engineer with a love of music. The most common questions they ask are: how can they get started in the business or, can we recommend a school that teaches *recording*?

If having a love for music does not necessarily make someone a good engineer, then what does?

Nearly every owner and studio manager is faced with the same problem—a constant wave of phone calls, resumes, demo tapes, and requests for interviews from novice and not-so-novice engineers. Sometimes these applicants come in with decent qualifications; other times they have little or no audio experience.

Let's assume for the moment that *your studio* has an opening for a trainee engineer—how do you make the right choice from all the available applicants? Will you look for a recording school graduate, or take a chance on the persistent musician, who's been musically involved since grade school?

Many know how much time and energy it takes to become a qualified "first engineer." The consensus is usually about 10 years. The effort necessary to train a new engineer mandates an equitable return on the time and effort you will invest.

Having faced this situation a few times, I offer a few observations.

Although the number of "quickie" recording schools continues to grow each year, I have serious doubts as to their usefulness and credibility. Before going any further, let's note that there are many good schools out there—mostly four year programs and often at universities that offer degrees. The majority of 12-week schools do little more than take the students' money in exchange for a chance to hang out and "play" with the equipment, providing minimal or unqualified instruction. Unfortunately for owners and/or managers of professional studios, the graduates

are seldom screened, either before or after the *course*, for long-term potential.

Learning by rote, a list of microphones, EQ settings, patch configurations, and signal processing setups is certainly a part of an engineer's education, but it is often the easiest to teach and simplest to learn. Most short-program schools fail in their ability to teach students how to *think audio*.

There are a number of factors that make an engineer good...not the least of which is that ever-popular subjective quality, recognizing good sound. Sure it's important that engineers are honest, trustworthy, and have good personalities, but what about their ability to "think and hear?" Maybe this is taken for granted...**it shouldn't be.**

Experienced engineers don't plug in a microphone, listen to what they get, and try to adjust or process the sound to their needs. They evaluate the situation by identifying the desired sound, sound source, acoustics, available equipment, available time, end product and the final application. All this happens before any microphone, EQ, or other signal processing is selected. This is possible because the engineer is mentally able to "hear" the needed results in advance, and then apply their abilities to create and capture those sounds.

Several years ago I concluded the single most valuable attribute a recording/mixing engineer could have was the ability to "hear" sounds mentally before hearing them aurally.

With this in mind I developed a simple little test to tell more about the long term potential of that aspiring engineer (who's a graduate "cum laude" from the "Stardom" school of rock-and-roll recording, and willing to sweep his way to the big time) than all the demo tapes and resumes combined.

Here's how it works. When interviewing

the aspiring engineer it is necessary to establish their experiences and abilities with sound. To do so, ask the person if they can hear, in their head, the sound of the following instruments. Start with something easy like a guitar, trumpet or snare drum. These are very common and should stimulate an immediate response. Now, move on to something a little more unusual such as a viola, oboe or French horn. Here again these instruments are not so unusual that the person should have trouble mentally hearing them, but you may be surprised.

Continuing on, suggest they hear a contra-bassoon, an English horn, a bagpipe, a mandolin or any number of other uncommon instruments that may one day cross their path as a "first" engineer. Note not only their response, but their body language. Are they responding "yes, I can hear those instruments," but struggling with the fact these instruments exist. This short quiz can offer significant insights to the engineers' future potential.

The most promising applicants will appreciate all forms of music as well as sound in general. It will be fairly obvious if they have given the qualities of different sounds any serious thought. It is this conscious thought over time, which allows the journeyman engineer to recall the instrument's sound mentally.

Studio and control room hardware exist to help the engineer achieve the sound that is heard mentally. Once that sound is "heard," the engineer can use tools, tricks and experience to direct the sound through the system and out the monitors—so it matches the original mental picture.

The potential for greatness resides in the *minds* not the fingertips of young engineers.

REP

Michael Fay
Editor

This is the mixing console that will cause a revolution in 24 track studios.

The world's first Dual Mode mixing console.

With the advanced features and performance of a desk you'd expect to cost a great deal more than it actually does, it's a remarkable achievement. Take Dual Mode.

When you record with the TS12, its in-line monitoring means you create a control room mix on the main faders.

In mixdown this becomes the basis for your master mix. Saving time and helping you keep a train of thought as to how the final mix will sound.

Unlike other in-line desks, though, the TS12 also allows true stereo subgrouping on long-throw group faders.

In this mode the routing matrix offers either six stereo groups or four extra auxiliary sends – totalling seven sends – plus four stereo groups.

No other console in the world provides such versatility.

The TS12 is an open-ended investment, with optional disc-based SMPTE automation for faders, meters, EQ in/out and auxiliary on/off. Again, at the price it's unique.

And fittingly, audio performance is superb.

Recent Soundcraft advances in summing amp technology, and in grounding and decoupling systems, make the TS12 one of the cleanest and quietest consoles ever.

The mic amplifiers, a critical factor in console quality, create less than 0.01% distortion at 10kHz at 20dB of gain. (Easily exceeding 16-bit digital specifications.)

Standard features are impressive, to say the least. Six auxiliary sends, seven stereo line inputs or effects returns, a 'musician friendly' headphone mix, an extensive 19" metal frame patchbay – and the option to create a massive total of 102 inputs.

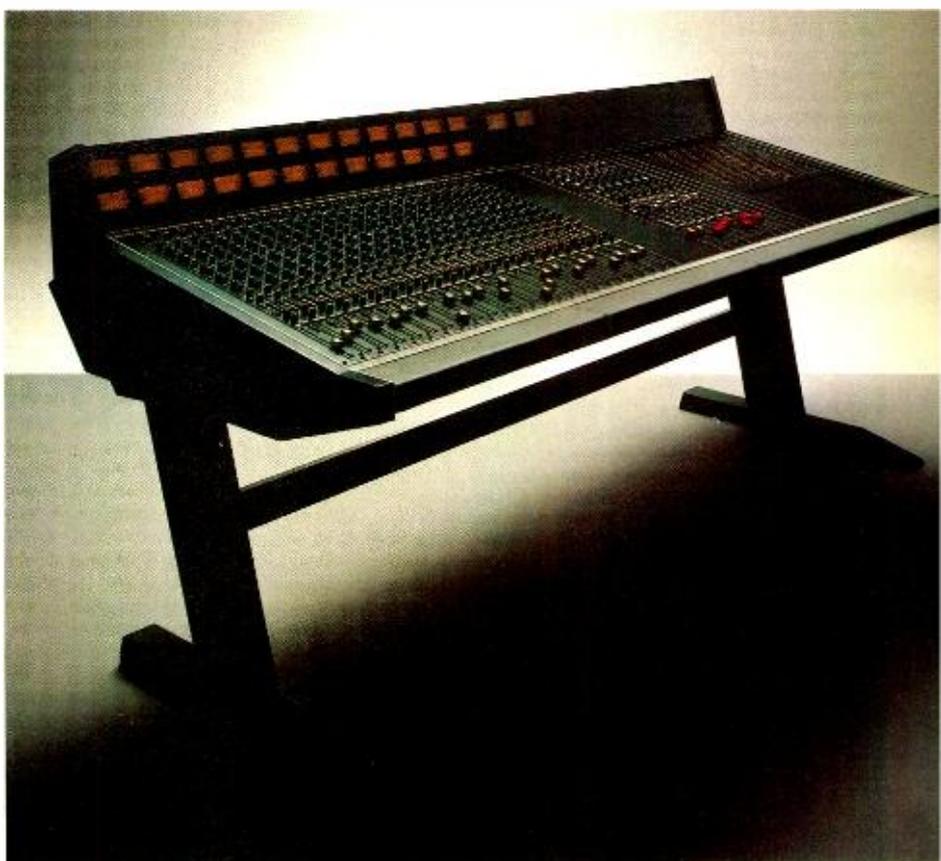
Quite a line-up.

Attention to detail is equally stringent with modular PCBs, no dual concentrics and a clear, logical layout that belies the sophistication inside.

But the most remarkable feature of the new TS12 is without doubt the price.

We suggest you call us today to find out just how remarkable.

ALL THE FEATURES YOU'D EXPECT IN A \$50,000 CONSOLE, EXCEPT ONE. THE PRICE.



LETTERS

Mic clarifications

From: Bruce Bartlett, microphone project engineer, Crown International, Elkhart, IN.

I enjoyed reading the helpful article by Fred Ginsburg on production sound recording in the November 1987 issue. It contained some statements that may be misleading, however, and I hope the author doesn't mind my pointing them out.

The article said that electret condenser shotgun microphones lack the sensitivity and reach offered by air condenser shotgun microphones. While this is true with current models, an electret capsule has the same sensitivity and reach as an identical air-condenser charged to the same polarization voltage.

The article also said that lavaliers come in two basic varieties: transparent and proximity-oriented. Since these terms are not used by microphone manufacturers, readers would have a hard time finding these types. I think the author meant omnidirectional and unidirectional.

In this same vein, the article states, "Until recently, all professional-grade electret lavaliers were proximity-oriented—in other words, they tended to add presence to close dialogue while rejecting background ambience." Since an omnidirectional microphone does not reject ambience, the author must mean that, until recently, all electret lavaliers were unidirectional. This is not so; unidirectional lavaliers are a recent development. The "transparent" or omnidirectional variety has been around a long time.

Thanks for the opportunity to clarify these facts in an otherwise excellent article.

State of the art

From: Nyya F. Lark, chief engineer, Galaxy Sound Studios, Hollywood.

I recently read Michael Fay's editorial in the January issue ["To Be 'State of the Art' Or Not"], and I find myself in total agreement with his views.

Since I came on as chief engineer at Galaxy Sound Studios, I had the challenge to bring the studio up to date and to maintain an image for our current and future clientele. A lot has changed over the years and as far as technology goes, faster than ever. There's a constant need to maintain an edge on the competition.

Here in Hollywood, you can imagine how stiff the competition is, as each studio has a like neighbor in less than a speaker's

throw away (less than a block). We have discovered that "negotiable" is a part of our business vocabulary, related to rates. What we do know is that it is vitally important to offer the best *service* to clients. Our policy is to make sure that all second engineers that are hired have a working knowledge of the outboard gear. If not, training becomes a part of the program. This is to ensure a smooth session.

Trying to keep up with all of the latest toys becomes a financial burden. We have an ample amount of outboard gear that is maintained, and it is also our policy to check out a new toy per month. But the emphasis is on the treatment of the client. If the toys function and the client's needs are met, then it's a far better guarantee that you will see the same client again.

Having toys up the yin yang and giving lousy service kills such a possibility. For years, we have emphasized the studio vibe in terms of comfort and accommodations. Then we found ourselves slipping into "he who has the most toys" syndrome at a sacrifice to the client. When I say "we," I am referring to the basic attitude of some professionals on the studio side of the business.

Obsolescence is definitely a word we are more than familiar with in the industry. It is aptly applied to the equipment, but you cannot attach it to the service aspect. Perhaps "state of the art" should be used in reference to client treatment.

Here at Galaxy, we have found much value in keeping our technical support staff well-educated by sending them to seminars, conventions and purchasing of audio materials. Again, all of this is to offer as much security as allowable in this industry.

Thanks again for an insightful and well-delivered editorial.

Workstation clarification

From: Kenneth W. Lonas, president, NASSA Design, Bearsville, NY.

On page 36 of your January issue, the caption accompanying the photograph of Reggie Lucas' MIDI workstation incorrectly stated that it was built by JB Audio. The rough application concept was presented by JB Audio, but the engineering design and construction of the workstation was actually done by NASSA Design.

NASSA has designed, built and installed workstations, production consoles and

other custom equipment enclosures for many studios and production companies, including Oasis Music, Electric Lady Studios, Bearsville Studios, George Benson, Sync Sound, Shelton Leigh Palmer & Co., Make Manieri's Centerfield Productions, the Cars, Jan Hammer and New England Digital.

In addition, NASSA specializes in the design and manufacturing of efficient workstations for the Synclavier Digital Audio System, similar to the unit that Reggie Lucas now uses at Quantum Sound Studios.

Mic Pre Listening Test

From: Gordon K. Kapes, president, Studio Technologies, Skokie, IL.

I would like to respond to the letter from Deane Jensen, president of Jensen Transformers, which appeared in your December 1987 issue. In the letter, Deane mentioned that an "important and relevant result of the Mic Pre Listening Test in your October 1987 issue was omitted."

Although the fact that Denny Jaeger chose the Jensen Twin Servo 990 mic pre for his studio is very nice, it has absolutely no relevance to the mic pre test results. Deane went on to ask, "so which mic pre really won the test?" Just look at the results, Deane. The Studio Technologies Mic-PreEminence was put through its paces along with seven other mic pres and came out with the top score (a full five points ahead of the Jensen Twin Servo 990, I might add).

I wonder if Deane is aware that our mic pre is being used by top sound engineers/producers in numerous studios throughout the country with reports such as "you saved me the cost of a new console... you've helped put fun back into recording...great product...very transparent and it's blowing us away."

Referring to the new Michael Jackson album, "Bad," engineer Bruce Swedien said, "I have been using your Mic-Pre-Eminence on Michael through the whole thing...tell people to merely listen to the sound of Michael's voice on the album and that is your pre amp. You can't miss!"

So Deane, which mic pre really won the test?

REP

Address letters to the editor to: Letters, *REP* Box 12901, Overland Park, KS 66212. Letters may be edited for length and clarity.

RECORDING

WHAT YOU DO WITH THE M-600 MIXER IS YOUR BUSINESS.

That's why we've designed it to meet or exceed your most demanding requirements. And made it the easiest, most flexible professional mixing console you'll ever work with.

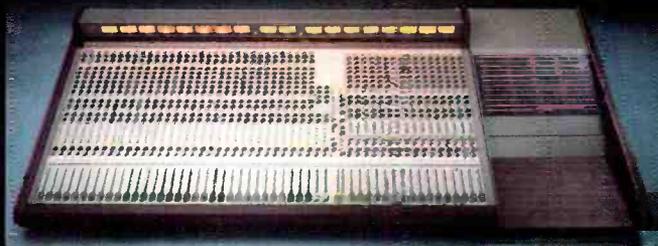
The M-600 is modular. Which means you can custom configure the console to *your* audio or video production needs. The M-600 lets you choose up to 32 input channels, or you can start with 16 or 24 input channels and expand the board as your needs change. Optional stereo modules can also be added to provide even more line inputs for MIDI instruments and video production convenience.

Installation and wiring is exceptionally easy. The M-600 is the only modular mixer that's available with all the necessary finished cables and installation hardware. And that can eliminate a lot of installation hassles and expense. At the same time, no other mixer at its price gives you multi-pin, computer-type connectors for quieter, more secure connections.

But the real pleasures of the M-600 will only be evident after it's in your studio. Up to 54 stereo or 128 mono inputs can be accessed directly from the top pane. A patch bay can be added for fast, flexible routing. That's convenience.

The M-600 has all the features you'd expect in a professional mixing console. Like balanced insert patch points on all inputs, PGM busses as well as the stereo master buss for increased signal processing capability. Plus sweep-type parametric EQ, balanced inputs and outputs, phantom power, talkback/slate channel and all the audio performance you'll ever need. Without the exorbitant price you don't need.

So check out the M-600 modular mixing console. It's ready for fame when you are.



TASCAM

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Circle (7) on Rapid Facts Card

AES announces awards program

The AES Educational Foundation has announced its 1988 educational grant program for university studies, with emphasis on audio topics. The awards are for graduate students only, and applications must be submitted by June 1 to be considered for the 1988-89 academic year. Additional information and application forms are available from the AES Educational Foundation, 60 E. 42nd St., New York, NY 10165; 212-661-8528.

Recording seminars scheduled for 30 cities

A series of 2-day seminars on selecting, integrating and maintaining recording equipment begins this month in Philadelphia on April 20-21.

Titled "Selecting, Integrating and Maintaining Recording Equipment to Meet Your Needs," the seminars will cover a wide range of recording systems, including analog, digital formats, compact disc system, audio for video, broadcast cart systems and standard cassettes. The seminars are being presented by audio consultants Irv Joel & Associates, and will be held in 30 cities in addition to Philadelphia.

More information is available from Irv Joel & Associates at 528 River Road, Teaneck, NJ 07666; 201-692-0010.

Electronic music workshop scheduled for June

The New England Conservatory in Boston will present its electronic music workshop from June 20-24, consisting of lecture demonstrations and hands-on sessions with analog and digital synthesizers.

Topics covered include the basic principles of sound; types of analog synthesis; frequency modulation; amplitude modulation and ring modulation; MIDI and digital sampling; Yamaha DX-7 programming; and a survey of various computer music programs. Afternoon sessions will consist of hands-on experimentation with various synthesizers.

No prior knowledge of electronic music is assumed or required, and the workshop is available for undergraduate/graduate credit or non-credit. Robert Ceely, the conservatory's director of the electronic music studio, will run the workshop.

More information from the conservatory at 290 Huntington Ave., Boston, MA 02115; 617-262-1120.

Video/storage industry announces trade shows

LaserActive, a trade show for the interactive video and optical storage industry, has announced its schedule of trade shows, including two regional shows.

The regional events, in Chicago and Toronto, are the result of an agreement between LaserActive and the International Communications Society. All shows will cover the latest developments in interactive video, CD-ROM, CD-I, CDV and write once optical storage.

The regional events are: LaserActive Chicago, Hilton and Towers, April 27-29; LaserActive Toronto, Constellation, June 8-10. LaserActive Boston, the industry's main show, will be at Boston's Hynes Convention Center from Oct. 25-27.

More information is available from LaserActive at Box 2401, Satellite Beach, FL 32937; 305-777-1609.

QSC announces leasing program

QSC Audio Products has announced a leasing program aimed at helping touring sound companies finance system purchases. The program will help clients lease entire packages couples with their amplifiers, including speakers, outboard gear and consoles.

The program is available to qualified QSC accounts; a brochure describing the program is available from the company at 1926 Placentia Ave., Costa Mesa, CA 92627; 800-854-4079 (in California, 714-645-2540).

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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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NOTHING REFRESHES A MIX LIKE A SIX PACK OF MIDIVERB II'S



Mixing is no picnic. Especially when you're in the hot seat. Consider the pressure. The fatigue. The late nights. And all the agonizing over what outboard to use on what tracks.

If you've ever sweated out a mix thirsting for more effects, the Alesis MIDIVERB II is pure refreshment. Whether it's the perfect room simulation for the hi-hat, or the perfect chorus texture for a last minute synth overdub, MIDIVERB II delivers. And, at an astonishing \$269, it's no wonder pro engineers are using multiple units to strengthen their processing 'front line.'

With 16 bit linear PCM, 15K bandwidth, and tons of musical character, MIDIVERB II is the #1 selling

signal processor in the business.*

That'll only surprise you if you've never used it. Those who have used it love the sound so much they can't resist buying several more. With 99 programs — 50 reverbs, plus choruses, flanges, delays, and innovative special effects — Midiverb II redefines the meaning of cost-effectiveness.

So after today's mix, you deserve something refreshing.

Ask your Alesis dealer to break open a sixpack of MIDIVERB II's. Your next mix could be a picnic.



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*Based on *ML* and *Sound Retailer's* monthly survey of 1,200 audio dealers

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www.americanradiohistory.com

SPARS ON-LINE

By Wilber W. Caldwell

The Death of a Myth

As most of us know, our business is more vulnerable to rumor and myth than most. Why is it that popular myths flourish so well in the modern recording studio? What role does illusion play in the technology of waveforms and electronic logic, of magnetic fluxivity and in the undeniable truths of 1s and 0s?

The reason lies in the fact that we wield the technology in the pursuit of art, and art itself is the most elusive of goals. There is no avoiding the arbitrary opinions and false impressions created by those engaged in the pursuit of art.

For a very long time one of the most powerful myths of the audio recording business has been the notion that the technology, the skill and the "art" required to create a commercial product are somewhat less than what it takes to achieve success in what we still archaically call "The Record Business."

It is my opinion that this myth is dying, but myths die very slowly. Myths survive without regard for facts. They are a product of human nature that often overlooks logic and reason.

Commercial production may require a different mind-set and motivation, but it is clear to many musicians, studio owners, engineers and perhaps to the general public as well that the pursuit of commercial forms is no less demanding than that of making records. It is simply different, and as time passes it becomes less different.

Those of us who began our audio careers in the 1960s have lived through the great crossfade. We have seen major changes in "The Record Business." A drastic tightening of the belt, brutal competition, some sloppy business practices and a demand for more technology without a corresponding increase in budgets.

Wilber W. Caldwell is president of Doppler Studios in Atlanta. He is also regional vice president and a member of the Board of Directors for SPARS.

Simultaneously, we have also seen a rapid expansion of the commercial production business. Recently there has been a great rush by almost all of the major studios and many major artists to embrace commercial production in one form or another. Even studios controlled by the big record labels are now actively soliciting commercial work. As Michael Jackson struts for Pepsi and Phil Collins croons for Michelob, major hits of the past 20 years are continually popping up in TV commercials.

Beyond radio and television, the field of commercial production includes many other sectors of the recording business. Surely advertising is a substantial slice of the commercial pie, but it is far from all of it. As the growth of the commercial production industry accelerates, it is growing out of its stepchild image.

**As the new markets
become more and more
diverse, the
technology follows.**

Our entire society is changing and *communication* is the buzzword. Almost every aspect of our personal, business and institutional lives is relentlessly bombarded by media of one sort or another. The demand for audio services in this diverse arena of commercial production is not only growing in sophistication and specialization, it is also growing in a broad-based groundswell of applications. Business, government, education and even religion have new needs that are creating new business that would have been unimaginable 10 years ago.

Elaborate A-V productions of motivation, information and training have created a dynamic new industry. Almost any kind of information imaginable—from Shakespeare to hypertension therapy—is now available on audiocassette. The demand of "in house" videotape presentations is creating a mushrooming industry at all levels of production sophistication. The success of cable television and home video has created an insatiable appetite for programming of every variety. The demand is limitless—and almost every form requires quality audio. In short, while the

media continues to grow and demand more sophisticated technology, we find massive decentralization and thus, specialization.

As the new expanding markets become more and more diverse, the technology follows. We are in an era of a great technological spread. The high end is characterized by digital multitrack recorders, larger and larger consoles, digital workstations and heaven only knows what will come next. The lower end is growing with an increasingly more sophisticated and more affordable array of "personal use" studio gear: inexpensive multitrack recorders, a variety of synchronizers, high-quality signal processors and effects, which are as flexible as they are cheap. And let's not forget about the ever-broadening applications of MIDI and sampling.

It should not go without note that, in the midst of all this change, SPARS has subtly redefined its acronym. SPARS is no longer the *Society of Professional Audio Recording Studios*. It is now the *Society of Professional Audio Recording Services*. This renaming, along with some changes in membership categories and costs, comprises a vigorous effort by SPARS to embrace the broadening audio industry. If you work in the commercial audio field, and if you (or your employer) are not a member of SPARS, you would be wise to consider joining. SPARS offers many ways to get more out of this industry. It also provides an opportunity to put something back.

Commercial production is a dynamic and expanding segment of the recording industry. Commercial producers and their product are gaining more and more respect within the industry. With this respect comes responsibility. This new respect must be earned by applying the same dedication, the same criteria for quality, and the same fanatical zeal for perfection that has created respect in the record business.

Myths die hard. They linger. The myth of the commercial producer as a second-class citizen is not altogether dead in the audio industry. But with the power of new technology and the momentum of the marketplace we may finally see this tired myth being laid to rest.

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UNDERSTANDING COMPUTERS

By Jeff Burger

Popping open the Hood

As promised in our first column last month, it's time to "pop the hood" and take a peek inside the average computer. Before we can effectively discuss the functions of the hardware and the software, we have to spend a moment reviewing the basic nature of the information in the digital medium.

While the analog world consists of fluctuations and amounts, things are either on or off in the digital world, and as the term "digital" implies, even this has to be put in numerical terms. It's really straightforward—zero represents off and a one signifies on.

You've no doubt heard the terms "bits" and "bytes" thrown around, and for good reason. Bits are the smallest pieces of data that computers can deal with, and a bit can be either on or off, period. It takes a stream of bits to amount to anything significant like a command or a string of meaningful data. This is Base 2 arithmetic—real exciting stuff! (If you're starting to wish that you had paid more attention to high school algebra, bear with me a moment. You don't have to go around counting in Base 2, but we do need to set the stage for concepts to come.)

Even the computer has to find a higher level of dealing with the tedium, so batches of bits are gathered and manipulated with greater importance. A byte is a group of eight consecutive bits, with each bit representing a different value position.

The rightmost bit represents a one, the next denotes a two, the third a four, the fourth an eight, the fifth a 16, the sixth a 32, the seventh a 64 and the eighth a 128. Each bit is still either on or off, indicating the presence or lack thereof of the number it stands for.

By combining the status of the 8-bit pattern, any number from 0 (00000000) to

255 (11111111) can be described. For example, the binary number 10001010 is the equivalent of 138 in decimal (128+8+2). Remember that the lower place values are to the right just like with a decimal notation. (See Figure 1.)

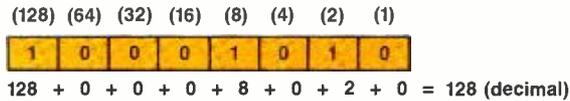


Figure 1.

decimal	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
hexadecimal	\$0	\$1	\$2	\$3	\$4	\$5	\$6	\$7	\$8	\$9	\$A	\$B	\$C	\$D	\$E	\$F	\$10

Figure 2.

Even this extension of bits into bytes is hard to deal with when the reference remains in binary. Because there's no easy conversion to our human decimal system, Base 16, or hexadecimal notation, is the default.

First, let's acknowledge that the only reason the human race has standardized on the decimal numbering system (Base 10) for our everyday needs is that we have 10 fingers. Since the inherent binary extension of 1, 2, 4, 8, 16, etc. doesn't provide an easy conversion to tens, more appropriate numbering systems had to be employed to deal with groups of bits. While one of the first was octal (Base 8), the hexadecimal system (Base 16) is now used.

A byte can be broken into two halves, called nibbles (cute, huh?). Each nibble holds a number from 0 to 15, 16 values in all (remember that 0 is a valid number in computerdom). Our decimal numbering system is augmented using the alphabet like this: 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, E and F. A "\$" is used to symbolize hexadecimal notation. (See Figure 2 for how hexadecimal numbers compare to decimal values.)

What happens when we run out of characters? The value is carried to another place value and the lesser number is reset to zero. So just as the number 10 in decimal represents 0 ones and 1 ten, \$10 in hex represents 0 ones and 1 sixteen. This system allows a single character to

represent each nibble and our range of 256 possible values for a byte can be symbolized as \$00 (0) to \$FF (255). In the number \$FF the rightmost place indicates 15 ones and the place to the left represents 15 sixteens. 16x15=240+15=255.

Our earlier example of 10001010 binary or 138 decimal would be \$8A in hexadecimal notation. To see this correlation, first divide the byte into nibbles and read each nibble right to left. (1000 binary equals \$8 in hex and 1010 binary equals 10 decimal or \$A in hex.) See Figure 3.

As you may have noticed, there are numbers larger than 255 in the world. Representing them simply takes additional bytes. The byte representing the values up to 255 is called the least-significant byte (LSB) and the additional one to the left is called the most-significant byte (MSB).

The term "word" is used for groupings larger than the 8-bit byte. For example, a "16-bit word" refers to just what it implies—a grouping of 16 bits (two bytes) of data capable of representing a number up to 65536 values—0(\$0000) to 65535 (\$FFFF).

Just as we keep adding place values in the decimal system to allow numbers to rollover and carry, the same applies to

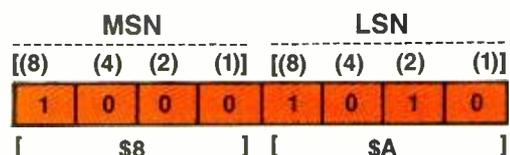


Figure 3.

Jeff Burger is RE/P's computer consulting editor, and is president of Creative Technologies in Los Angeles.

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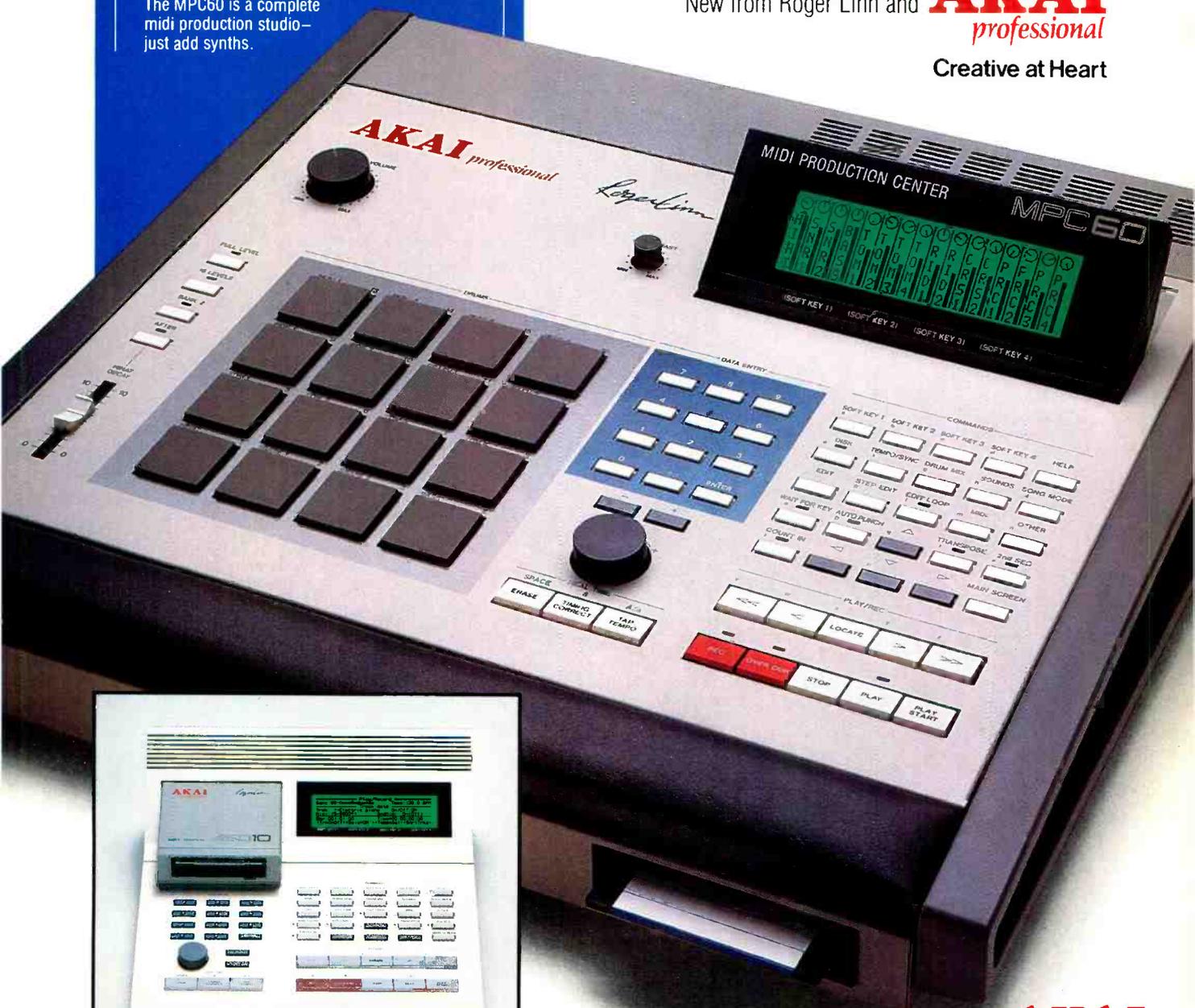
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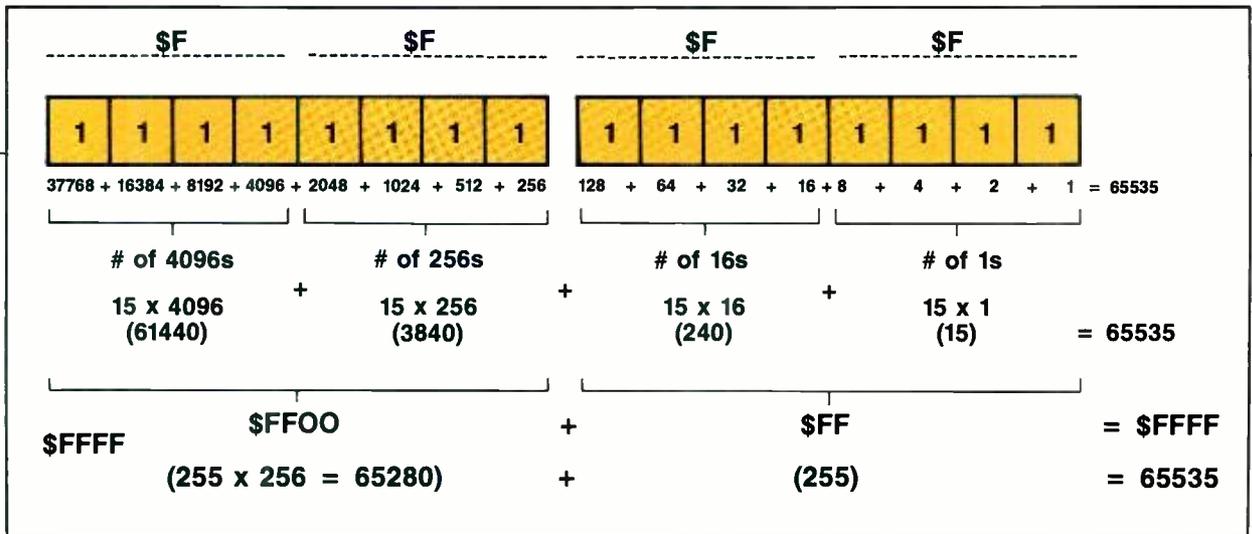


Figure 4. Example shows the value \$FFFF in hexadecimal, decimal and binary notation. Each \$F represents a nibble.

Base 16. In hex, the number above \$FF (255) is \$100 (256) with the new place value representing the number of 256's. The number above \$FFF (4095) is \$1000 (4096) with the new place value indicating the number of 4096's. The LSB represents the number of ones and the MSB taken as a whole represents the number of 256's. Given binary, decimal or hexadecimal there are many ways to calculate byte values, but as the following diagram illustrates, the answer is the same. (See Figure 4.)

But what about negative numbers? Glad you asked! A word can also represent positive and negative numbers with a trade-off of reduced range.

This is accomplished by having the most-significant bit of a word represent a sign bit for the value of the remaining bits. This allows an 8-bit byte to represent the values of +/-127 and a 16-bit word to cover a range of +/-37768. Whether or not a word represents a full-range of positive values or a restricted range of positive and negative numbers is up to

how the program itself translates these values.

Let's bring this concept of bits, bytes and words home by applying it to the audio domain. In the world of digital sampling and recording, we hear specs being thrown around like "8-bit sampling" and "12-bit sampling," with the cream of the crop being 16-bit sampling and beyond. The original analog waveform is broken down into a stream of zeros and ones—you guessed it, bits. Each grouping of bits represents the amplitude of one minute section, or snapshot, of the waveform.

The greater the range of possible values for each amplitude, the higher the dynamic range is. Eight-bit sampling offers a range of 256 levels, or +/-127. At 6dB per bit (trust me), that's a dynamic range of 48dB (what's wrong with this picture?). By using a 12-bit word (or a byte and a half) to represent an amplitude, we get 72dB or dynamic range. Now you can see why 16-bit sampling is so coveted: it yields 96dB, or the dynamic range of a CD.

At this point, it may become clearer why certain numbers keep reoccurring in our lives. Looking at the MIDI spec, we find 16 channels, 128 notes and velocity levels, 4096 values for pitch bend, etc. Programmable gear often has 16, 32, 64 or 128 memory locations. Computers offer palettes of 8, 16, 256 or 4096 colors. Computers usually have memory capacities such as 64K, 128K, 256K, 512K and 1Mb. These are all values that can be represented economically by small numbers of bits or shared in the confines of a byte or two.

Now that we've established some common terminology, next month we'll get down to business with the hardware that processes this digital data.

REP

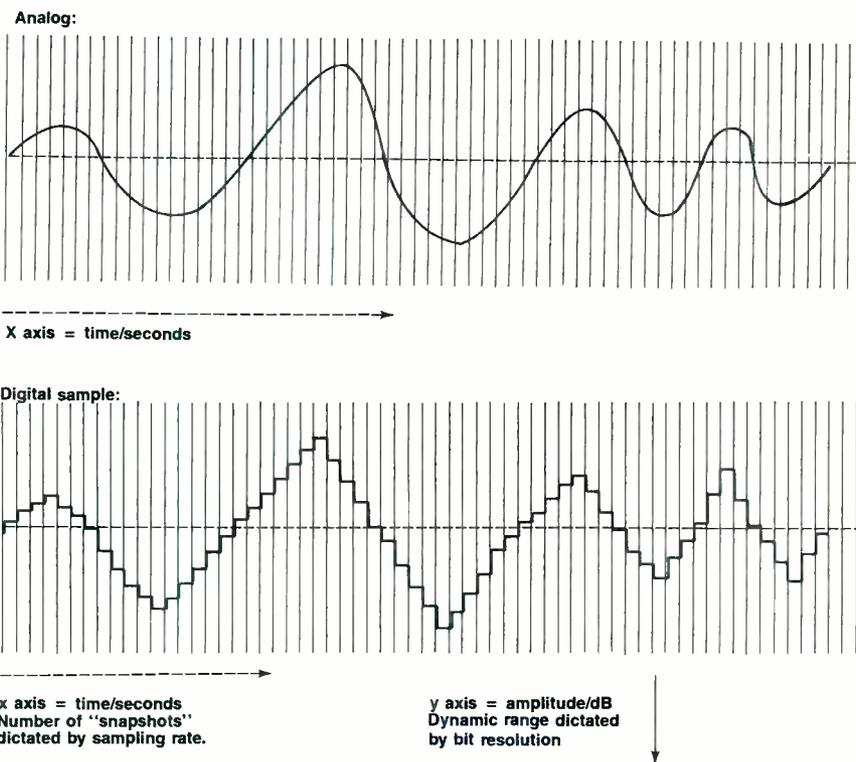


Figure 5.

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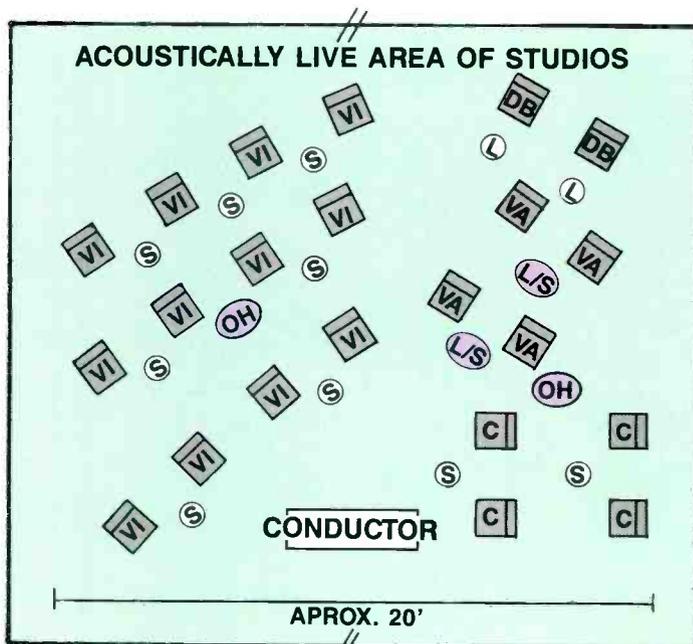
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The Engineering "Hot Seat:"

Commercial Music Production

By Michael Fay

Every aspect of a high-speed, high-pressure commercial production session has to flow through the "media" engineer.



VI = VIOLIN
VA = VIOLA
C = CELLO
DB = DOUBLE BASS
S = SMALL CAPSULE CONDENSER
L = LARGE CAPSULE CONDENSER
OH = OVERHEAD LARGE CAPSULE TUBE OR
CONDENSER MIC
LS = LARGE OR SMALL CAPSULE CONDENSER
□ = CHAIR W/BACK

Figure 1. Setup for string section.

There are as many ways to get good sounds on tape as there are engineers and studios. But in the world of commercial recording, it becomes necessary to develop a basic *style*, which includes a repertoire of proven techniques.

I would like to share some of the tricks and techniques that are part of my style. The methods you use or develop may be vastly different...what's important is that you know what you want and how to get it quickly. If you have confidence in your work, your clients will have confidence in you. Only experience allows you the time and perception to break new ground.

Scheduling

Effective scheduling is a must for the success of a project and is usually the job of the arranger/producer or music contractor. It is not uncommon for a commercial project to come together on short notice. I found it helpful to stay in constant contact with my regular clients, as this relieves them of the burden of tracking me down. A phone call every couple of days to "see what's going on" is all it takes. These calls can be as much social as business. *I need work* is never the tone of the call.

When I get a work call I like to know who the musicians are in advance of the session. This lets me plan the setups. For example, if I know the drummer all the right mics can be out and ready to set up when he arrives. Every musician uses a little different setup, and being prepared for the regulars makes me more efficient.

It is also my responsibility to note any problems I see with the proposed schedule. If the producer calls for one hour, and I know from experience it's going to take an hour and a half to set up for 12 strings, record three or four stacks, marry the tracks, tear down and set up for singers...I let the person know. They respect this, and it will probably save them money in the long run because their schedule won't get backed up.

More often than not the arranger will want relatively the same sound throughout a given project. So once the levels are set for a given ensemble, those levels can remain for that sections tracking. Variations are likely, but are generally the exception.

If it hasn't already been done, the rhythm section setup is next. Usually consisting of drums, bass, keyboards, and guitar, this will become the backbone of the music. Any timing errors or problems in the basic arrangement must be corrected now. As this is often the first chance the client has to really hear the "feel" of the

Michael Fay is editor of RE/P magazine.

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worked with want only rhythm, piano, and click in their phones. Any other instruments make it difficult for them to hear each other as well as play in tune with the track. For the same reasons, they also won't want to hear *their* basic track in the phones until they're ready for the second pass. Also they'll want the feed to their phones turned much lower than other musicians.

There are many other overdub elements that can be described, but for the sake of space I'll limit my descriptions to these remaining two—brass section and group vocals.

• **Brass**

A typical brass section consists of three trumpets, three trombones and possibly three or four French horns. Getting a good brass sound is a commercial production "must." Whenever possible I like to record the French horns separately from the trumpets and trombones unless I'm working in a very large and lively room. With this in mind I'll describe them separately.

I like to have the trumpet and trombone sections facing each other and about 15 feet apart. With the chairs placed in a slight arch and three-four feet apart, the trumpet setup involves two SCC set in an XY position, at a height of five feet and

centered about 10 feet from the middle chair.

You may need to explain to the players that they must control their own internal balance whenever possible, and that they shouldn't play into the music stands. Because the "lead" trumpet usually sits in the middle, I can "reach" for a little more of the second or third part if necessary. If you're not already using this mic technique—try it, you'll like it.

The distance from the instruments can be varied somewhat based on the room acoustics. Generally speaking, all acoustic instruments sound better in "live" rooms.

A trombone is one of the easiest of all instruments to record. Almost any good dynamic mic will do, placed about three feet in front of each instrument.

Getting a good French horn sound can be a little trickier. I line up the chairs side by side with about three feet between each one. Behind and to the right of each chair, I'll use a LCC in an omni pattern. These mics are on desk stands and back about four-five feet.

If you're fortunate to have a very live area in your studio, put the horns there. By moving the mics to adjust the ratio of direct to reflected sound, you can get a wonderful natural French horn sound.

When working with 24 tracks you usual-

ly can't afford more than one track for each of the three brass sections. I try to end up on tracks 12-14 if possible.

If the arranger/producer wants to stack any of the brass you must be very careful that the two stacks don't phase. The easiest way to prevent this is to ask the musicians to move their chairs either forward or backward about two feet. This changes the time domain and will eliminate most phase problems. Changing the mic position will accomplish the same thing.

On to the vocals. Groups of four or six are the most common, and can be dealt with quite easily with some basic understandings. Start by grouping the mics and music stands for pairs of singers. If you're working with two female and two male singers, have the pairs facing each other. For six, a triangular or three sided square works well.

Try to maintain clear sightlines and have them comfortably close. Like the string players, the singers will often use only one side of their headphones. The open ear will be listening for intonation and blend—as it is happening in the room.

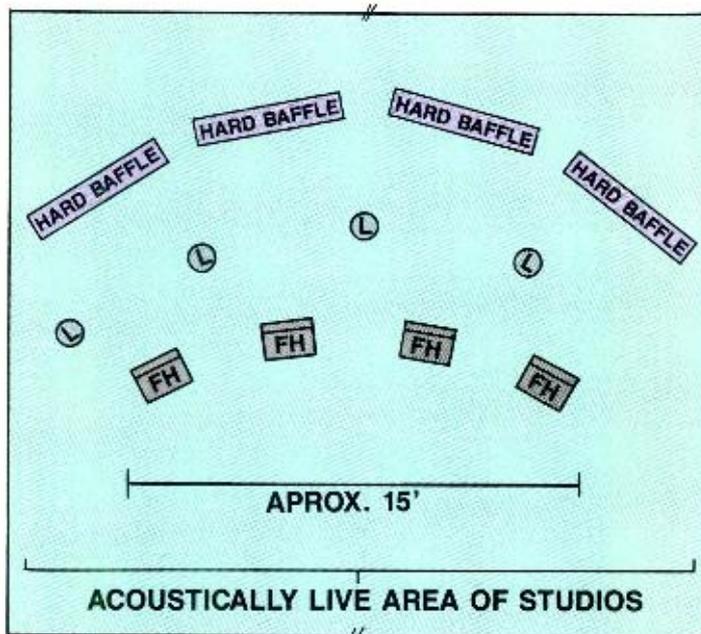
I have several favorites for mic'ing group vocals. All are either LCCs or LCTs.

My sidechain processing consists of a tube compressor(s), and a single ended noise reduction unit(s). Setting the noise reduction unit for low pass filtering caused a very gentle gating effect when the singers are not singing, which reduces the amount of headphone leakage that goes to tape. This proves superior to any other manual or automatic muting configuration available because it is both fast and inaudible. Because this type unit doesn't shut down the entire bandwidth, it works well for normal conversation (below gating thresholds) levels that occur when the tape is not rolling.

My clients like three or four stacks of group vocal. I record the first pass on two tracks so the clients and the arranger/producer can hear better. On the second pass I record over the number 2 basic track. The group vocals remain panned mid left/right to give a full sound and also help identify problems. Another common trick is to solo the second thru fourth tracks against the basic. When this is done, you can tell immediately if there are glitches or intonation problems. I generally use tracks 17-20 for the group vocals.

With sync on 24 and click on 23 that leaves tracks 6, 11, 21 and 22 for other overdubs. This isn't many but with careful planning and the help and understanding of the arranger/producer there will be room for everything.

Having the click and sync tones on adjacent tracks can sometimes be a problem. Part of the solution is to print the click



FH = FRENCH HORN

■ CHAIR W/ BACK

▭ = PORTABLE BAFFLE WITH HARD SURFACE FACING THE MUSICIAN.

Figure 3. Setup for French horn section.

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softer than normal. But with a soft click you'll need to turn up the "send" in order to deliver the levels the musicians need. A gate will clean up the click track. After

it's gated, there won't be any audible cross-talk from the sync track, and you've solved the problem. Another way to deal with this is to skip track 23 and put the

click on 22. This can present other cross-talk problems on tracks 21 and 23, so be careful. Don't forget, when recording the click track, its clock source must be the

Is your studio ready for commercial production clients

There are many areas of commercial production that a traditionally "record-oriented" studio can successfully compete. One such area is recording industrial music packages. These "custom music" packages are produced for applications like the shows at Magic Mountain or Cypress Gardens, and station "image" packages for radio and television. Not all commercial production requires a big investment in video and interlock hardware. In fact, most of these projects require no video equipment at all.

There are however, a few tangible requirements that are necessary to understand, in order for your facility to be considered as "usable" to a commercial client. They include the following:

- *An efficiently designed facility.*
- *Engineers with media experience.*
- *Well maintained equipment which can be operated quickly.*
- *A service oriented attitude.*

It is not uncommon to have dozens of separate "starts" included in one of these packages. As you might imagine, efficiency and comfort will be a high priority throughout the process.

Generally speaking commercial producers and arrangers like working in studios that are well-lit, clean, have fairly good elbow room and good sight lines. All these elements are conducive to entertaining clients and spending relatively long hours (and days) in a high pressure and high speed recording environment.

What is meant by an efficiently designed studio? This means more than good sounding or even roomy. It means plenty of parking; easy loading access in and out of the studio; plenty of light; enough control room space for the production personnel, clients, musicians and any instruments; adequate lounge facilities for clients, musicians, and guests; and ideally, warm-up and storage areas for the musicians and their instrument cases.

Equipment

There are certain types of professional audio equipment that are "faster" than others. This has nothing to do with the slew rate of

the pre-amp, but rather the time it takes to perform routine tasks. For example some tape transports are slower to search, locate and play than others. Some consoles require time consuming patching in order to "marry" or "ping-pong" tracks. Or, the input channel mutes aren't easily accessible. Perhaps the microphones and stands are still being attached with threaded connectors, rather than using the quick disconnectors that are readily available. The list could go on through most of the studios equipment. I'm not suggesting you go out and buy all new hardware just to address the commercial market, but I am suggesting you consider these points when making future purchases.

In a commercial production session the headphone system is much more important than you may think. A clean and powerful system can be the difference between keeping and losing a client. Your system should at least meet the following requirements.

1. *200W-250W of power.*
2. *Include stereo sends (desired but not mandatory).*
3. *Feature reverb sidechain access (desired but not mandatory).*
4. *Provide multiple sets of high-impedance headphones. Remember if you have a large string section, the number of phones can add up to quite a load on the amp.*
5. *The headphones should be capable of disassembly so that the musicians, who need to, can work with only one side connected.*
6. *Provide multiple distribution boxes. Within reason, it is good if each musician can have their own level control.*
7. *The headphones should have replaceable elements. Because of the constant pounding of a click track pulse, and sheer wear and tear, you'll need to replace elements on a regular basis. The cost of replacing the entire headset is prohibitive.*

As a facility owner, if you plan to pursue commercial clients, it is vitally important that you have an engineer available with media experience.

What does this mean? Even if you only plan to do industrial music

recording, (without SMPTE etc.) the engineer you hire had better have experience working under time and pressure constraints. This is not the place to break-in someone new. If you successfully sell your studio to a commercial client and you don't have an engineer with media experience, the studio will likely develop a reputation of not being able to cut it—even if the facility itself is quite nice. Invest the time to find a qualified engineer, then give that person ample access to the studio (in advance of the first session), so that he/she can familiarize themselves with your facility.

This is a good opportunity for assistant and intern engineers to pick up some of the techniques involved, but it's crucial that the assistants be chosen carefully. The stress associated with this type of production can be incredible. An ill-timed or ill-chosen comment can be disastrous.

Case in point: After the completion of my first commercial sessions I was taken aside by the producer and told that the arranger had requested I not be used again. The reason? During the course of the project I had made, what I thought was a constructive comment about a trombone part. As it was the arrangers first job for this production company, and he was under a lot of pressure too—the comment was not well received.

Your companies service attitude may well be the most important factor in enticing new commercial production clients. When these clients come to you they need more than a job done. They often need your help and understanding. If you can provide a "safe haven" for them it makes their already crazy jobs easier.

Take some responsibility to properly educate your less technical clients. Help them learn the proper terminology and capabilities of your business. If they are intimidated by the process, they will find it easier to avoid you than to deal with yet another uncomfortable situation. The more you can teach them, the more they will come to trust and rely on your expertise for all their future production needs.

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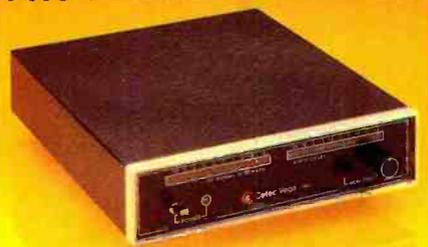
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sync track you've already laid down, in order to be in sync.

And last but certainly not least, the mix.

A commercial/industrial project might have anywhere from 10 to 40 separate "starts." Now that it is time to mix, you'll see why it's so important to keep as much consistency as possible in the tracking stage.

If the project is destined for broadcast, chances are you'll be making five or six "pullout" mixes for each start. A pullout is an alternate mix. It usually involves taking part or all of the vocal track(s) out, but may also involve alternate instrumental combinations as well.

Once you set the final rhythm mix there's a good chance you'll be able to let it run that way through the entire mixing session. Of course you might need to tweak it a bit here and there as you go along, but for the most part this establishes the foundation upon which all the music is written and recorded.

Like everyone, I have my EQ and panning preferences. The EQ I'll leave to you, the panning logic may be of interest. I tend to rely on two types of stereo perspective—wide and narrow. Though the locations tend to be similar, the pan widths will

come in toward the center for narrow stereo mixes.

The rhythm section. I like to view the drums from the perspective of the drummer. Seldom are they panned more than mid-left/right. The bass sits dead center. The keyboards range from mid-left to hard-right. The guitar is hard left to mid-right, if it's spread across two tracks.

Instrumental overdubs. Strings go full left/right as viewed from a conductor's podium. Low brass sit at about 10 o'clock, French horns in the center, and trumpets are at 2 o'clock. Woodwinds fit in the holes between 9 and 3 o'clock, and hand percussion lives in the gaps from hard left to 9, and hard right to 3 o'clock. Timpani and other low-frequency parts fit in between 11 and 1 o'clock. Other synth and sequenced parts are fit in as appropriate to the overall balance and blend.

Vocals. Lead vocals go in the middle with group vocals filling in around, from mid-left/right to hard left/right.

I am not a big fan of overly bright mixes, therefore I tend to track a bit on, what many would call, the dark side. Because of this I found that running the entire mix through a sidechain of 2:1 compression (for dynamic range control) and an "ex-

citer" device, gives me just the extra sparkle I needed without adding any significant EQ to the individual tracks. I found I could back the vocals into the track a bit more and still maintain intelligibility. The nature of exciter-type devices is such that the hottest signals (at and above the tuned frequency) are affected the most—because the vocals are usually the hottest element in a commercial mix, they stand out crisply without being 10dB hotter than the track.

My preference, and the preference of most of my clients, is to mix in stereo. Because so many commercial productions end up in mono this is still a touchy subject. My philosophy is that if I have done a careful job recording the instruments, and I can do the mixes in mono-compatible stereo, I will try my best to convince the client to accept a stereo mix.

There are probably several ways to measure the compatibility of a stereo mix. I judge by monitoring my mix in mono before the stereo print is made. By using my ears and experience I can tell if there will be any problems.

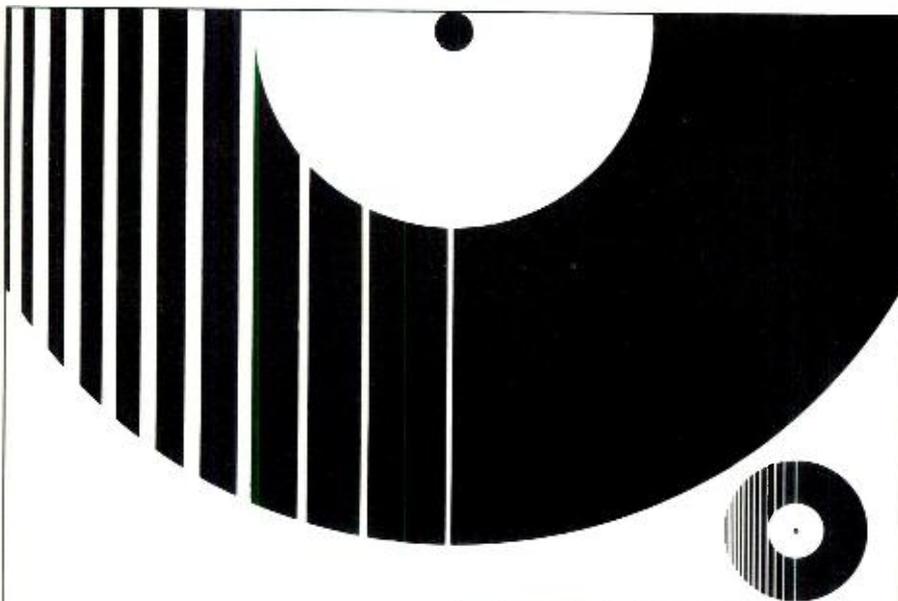
I know this doesn't take into account the next machine in line that might have a bad azimuth alignment. So, if I didn't know where the tapes were going, I made sure both the client and the next facility knew they had a stereo mix and were given adequate tones for adjustment.

Keeping the crucial elements panned at or close to center was another technique I found helpful in avoiding compatibility problems. For example, the kick, snare, bass and lead vocals were always at or very near dead center. The chordal instruments and group vocals were seldom panned past 9 and 3 o'clock. The remaining instrumentation would fill in the holes and take the hard panning. I called this my narrow stereo mix. This "compromise" worked every time, and allowed the client to have a stereo mix for presentation and if necessary, a mono compatible mix for broadcast.

As a perk to the client, arranger/producer and musicians, I might also go back into the studio (with fresh ears) and do a custom highlights mix in full stereo. Unless it was specifically requested, this would be done on my own time. As a promotion tool I convinced the studio to give me time in the off hours...so the cost was minimal.

Today there is a great deal of commercial work done using nothing but computers, synths and samplers. I don't mean to neglect their importance, but I felt an article on mic'ing technique was needed. Getting good sounds through a microphone is the backbone of the recording art. If you can capture the sounds of orchestral instruments, you'll have little trouble handling the machine outputs.

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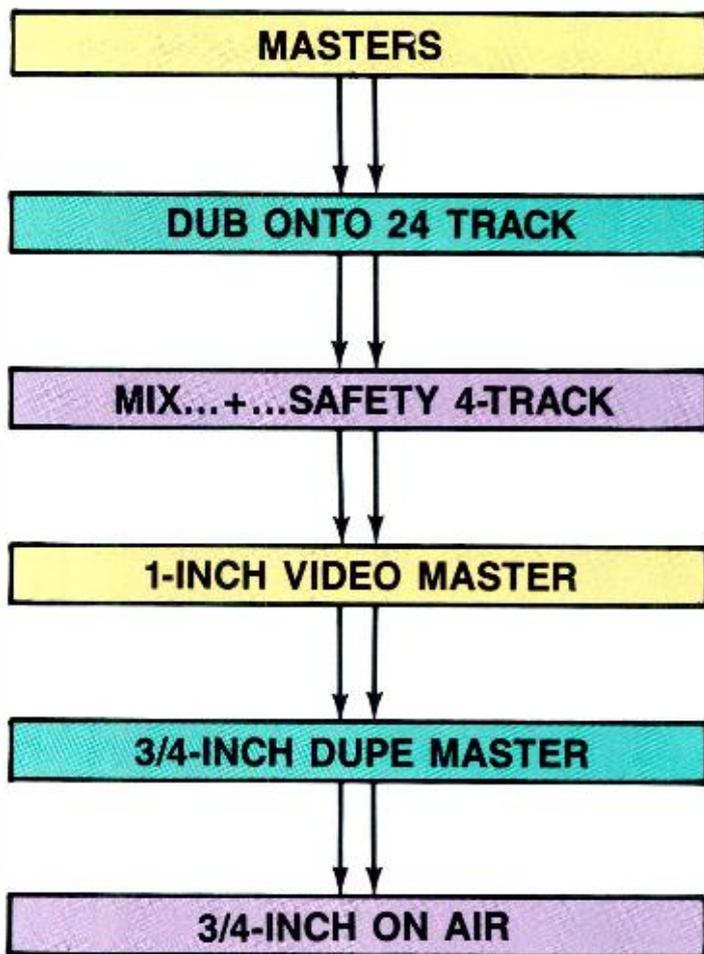


Figure 2. Sweetening session.

assembled in an on-line edit suite with all the fancy effects required. This is called conforming the video.

The sweetening session

The next step is to do the same with the audio: Go back to the original first-generation tapes and resync them to the finished picture, adding music effects, voice-overs, etc. In modern commercial production, there is usually a slight twist in that almost every element has been chosen for a specific reason, and with great care. For example, a commercial with a gorgeous girl applying her favorite makeup while talking about the product is very likely to be one girl with a gorgeous face, another with perfect hands, and a third with the perfect, *believable* voice.

The session begins by locking up (via SMPTE) a 24-track machine with a specially prepared 3/4-inch reference dub of the conformed 1-inch master. (The 24-track tape is blank except for the SMPTE.) A sweetening room has many more audio source machines than a standard music studio. The basic setup for a sweetening room is one 24-track machine, one 1-inch 8-track, one 1/2-inch 4-track, one 1/4-inch 2-track, one 1/4-inch center-track machine, one 3/4-inch video machine and one 1-inch layback or video recorder. The extra machines are to facilitate laying back the elements.

This process, also known as building up the track, is relatively simple in theory. First, the engineer locks up one of the source machines—with an element on it—

the reference video and the 24-track. When the element and the video are in sync, the elements are simply dubbed on to the 24-track. This process is repeated with each element until all of the music, sound effects and dialogue are on the 24-track machine. Variations (regional tags or dated offers, etc.) can be accommodated at this time by simply using additional tracks.

If you send a "sync tone" stripe (program on track 1, sync tone on track 2) to a video house, they may dub it onto a SMPTE stripe but will most likely use an event controller. Event controllers simply close or open a relay at some offset. They are used to start the tape machine. Once the machine is running, it receives its servo clock from the transport controller and so stays in sync. Getting the sound and picture to work together is then a matter of cuing the 1/4-inch machine up to the correct start point manually. This is fine if you only have to fly the track one time or if the subject is a short TV spot. But if an industrial video of several minutes is involved, the task is better served by a SMPTE stripe and a chase-lock machine.

Laying back the voice-over becomes more complicated when the client wants to change the phrasing of an announcer take. This is accomplished by starting the source machine at different offsets and punching in on the 24-track. To avoid this problem, the voice-over is often recorded at the beginning of the post session so the talent can look at the final picture and read with the music. In this case, the AVO (announcer voice-over) is recorded directly to the 24-track and a 1/4-inch (or 4-track) at the same time.

Note that complicated announcer or actor sessions would be extremely expensive in a post room, so this is only possible when the client is absolutely certain that they will get a good reading immediately.

Mixing a sweetening session

When all of the elements are positioned precisely in time, the next step is to balance the music, dialogue and effects. The most important aspect to accomplish is a perfect voice-over/music/effects balance. There can be no question that each and every syllable is prominent and clear.

Most of the following techniques allow the engineer to keep the voice-over at a reasonable (read low) level and still get the message across: soft gating or expanding the music track using the voice-over as a trigger, or compressing not only the voice-over and the music but the entire track.

Other processes that are commonly done to the voice-over to ensure a clear message include exciters that allow the voice to stand out in the track at a lower

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Preparing for a session

Preparing a 3/4-inch video for the music session.

- Use 30 non-drop frame SMPTE. 30 drop frame corresponds to elapsed time, but trying to calculate the offset is hard enough without having to remember to drop two frames every minute (except the 10th minutes).
- Start the SMPTE numbers at a convenient time, such as the first frame of picture at 1:00:00:00.00. Again, this will make it quicker and easier for the engineer to calculate the offset points.
- Make sure that there is at least 10s of pre-roll. This will allow all machines to resolve before the start of picture.
- It is a common misconception that the audio must start 15 (video) frames after start of picture. Film mixers often line up the music by simply cuing the start of the audio at 12 (film) frames. If this is not the case, be sure to include extra line up information in the take sheet.
- Leave 10s of post-roll. Otherwise, the machine will not know which way to go at the end of the spot and you will waste time.
- Visual and audio time code should match.
- Whenever possible use the SMPTE from the edited 1-inch.

At the music session:

- Is the video final cut? If not, what might change?
- Where are the tapes going? Often, this is the only information that a non-technical agency person should have to know. Once the music engineer knows where the post mix is, he can be sure to provide the optimum configuration of formats.

At the audio post session:

- Bring the elements on audiotape. Any elements that were previously transferred to mag are not likely to be of any use.

Bring the originals.

- Bring all of the derivative elements. For example if you have built up a music track or a dialogue track previously, bring the individual components in case you suddenly decide to change something.
- Try to edit and post in the same house. Every house has its own special methods and quirks. Also, if any problems arise later, one house will always blame the other. You could wind up with dueling engineers arguing on your time.
- The pressure of a post session has the tendency to change minds and make quiet people speak up. Everyone who was told, "We'll fix it in post" will be ringing your bell.

volume; short delay programs and early reflection programs to add depth and apparent loudness to the voice; and tight chorus and pitch shifting programs to "fatten" the AVO.

When audio is broadcast over the air, it is usually very heavily comp-limited. Special care must be taken not to lose the dynamics of the mix. It is not unreasonable to monitor and even print the mix through a 6:1 compression ratio, as the entire audio will be compressed again when it is broadcast.

When the final audio mix is completed, it is then laid back onto the 1-inch video master. A safety audio mix is simultaneously recorded onto an 4- or 8-track machine in case any changes or remixes are required later. It is also a common prac-

tice to print the "buy mix" and an additional mix with the music 3dB higher. This is a time saver if the client calls the next morning and says, "It's great, I love it, but the music should be a little louder."

Mechanics of the film mix

The film mix is similar to a sweetening session with the biggest difference being that all of the audio elements are first transferred onto mag. Mag is 35mm film with audio tape glued to it. A mag "fullcoat" has either three or four audio tracks—depending on the dubber used. A mag "stripe" has one audio track. During this dub process (see Figure 2.) all of the SMPTE numbers that the audio people have been working with become lost, and all of the "hits" will have to be re-aligned

by the mix engineer's eye.

In order to resolve the transfer from audiotape to mag, the transfer engineer needs sync tone or SMPTE, which should originate from the music 2-inch tape. Sync tone, often referred to as Vsync, 60 cycles or color sync, is actually the 59.94 cycles per second NTSC color video sync tone. It is *not* 60Hz from the power mains (called off-the-wall-sync, for good reason).

Once the mag transfers are completed, the mix engineer starts syncing, building and mixing simultaneously. As in a sweetening session, the reference video is usually on a 3/4-inch cassette. This means that while the VTC (visual time code) is 30fps, all your adjustments to the audio elements will be in 24fps. Instead of dubbing each element onto a multitrack, the mix engineer simply keeps locking up more machines called mag machines or mag dubbers. As he locks up more machines, he bounces track combinations onto a virgin mag fullcoat running in sync. Therefore, all of the elements will lose at least two generations in the notoriously poor film mag domain before they end up on the master 1-inch.

Comparing the two

From a technical quality standpoint, the sweetening session seems to make the most sense. This concept, when taken to the limit, produces arguably the highest quality sound tracks.

At Sync Sound in New York, audio post-production is done in the digital domain. Any intermediate mixes can be made during the process without the problems of generation loss. Instead of making a 4-track music sub-mix, the music 2-inch and all other first generation elements can be mixed at one time.

The in-house AudioFile system is capable of editing and playing back up to eight element tracks in sync with the picture. All of the editing is done simultaneously by one editor. Usually the Foley, sound effects, dialogue, and music cutters work separately and are not fully aware of each others actions.

The film mix engineer works with dubs. The sweetening engineer is able to work with the masters. This may sound obvious, but it is often the case in a film mix that the music 4-track (three audio tracks plus time code) is transferred onto three separate stripes instead of one fullcoat. This type of stripping off can lead to misunderstandings.

For example, I know of a music session where there were several hits in a 30-second spot. One hit at the top was two frames early, and on a separate stripe (which was originally just a different track on a 1/2-inch 4-track) a second hit fell two frames late. In context, the composer had

decided that these were "close enough."

But when the mix engineer was asked to slow down the motion, the clients made him slide the two tracks in opposite directions. The result was two tracks of musical effects that were four frames off. (At 120 beats per minute, that is just over 1/16th note.) You can imagine the sound of two tracks off by a 32nd note in opposite directions, while the bed track was running in sync.

If the music had been laid in during a sweetening session, the engineer would not have questioned the timing because there would only have been one piece of tape.

Notes for the music engineer:

1. *Splitting the mix.* Commercial music mixes are usually done at least two ways. A mono mix is mastered to a 2-track machine with time code or Vsync on track. At the same time, a 3-track split is mastered to a 4-track machine with the time code on track 4. The important thing to do with the 3-track mix is to give the post engineer as much flexibility as possible.

For a jingle session the 3-track split is usually band, lead vocal and group vocals. The split should be mastered using the console busing so that the post engineer only has to bring all the faders up in a line to hear your intended balance. In an instrumental mix the split would be rhythm section, melody instrument and sweeteners. (Sweeteners are strings, brass, cymbal rolls, etc.)

Often, the music is done before, or even simultaneously, with the final video edit. This causes confusion and makes it even more important to build flexibility into the mix. If an event might move in the final video edit, its corresponding sound may not fit in the 3-track mix (isolated in time) and may be left out of the mono mix entirely.

In this case, each event that might need to move will be sent out on a separate stripe. Whenever stripes are made, a sync point sound should be recorded roughly 2s before the start of picture on all mixes (popping the slate mic across several tracks on the 2-inch works fine.) This sound can be used to check the alignment of various elements by simply listening to the flam at the top instead of having to go all the way through the spot and then guess (where does a cymroll start?).

But, if the beep or click is too early in the pre-roll, the machines may not be locked up yet and the sync point will be useless.

When sending out a SMPTE center track stripe, remember that a 4s sound occurring at the end of a spot will have 26s of blank tape plus 10s of pre-roll before it.

If possible, at the end of the music session make a simple audio dub of the music back onto the working 3/4-inch so that the post engineer can see what your intentions are. This will help eliminate embarrassing questions like—is this supposed to fade up or fade out (for example, do I line it up with the end of the spot or the beginning?).

2. *Think ahead to the next mix.* The commercial music engineer rarely becomes involved in the voice-over or non-music sound elements. It may sound like the post engineer has a simple task involving only a few tracks, but this is not the case when each sound effect is on a separate piece of tape or on a separate machine. It is the post engineer's task to make all the elements work together.

For example, this can be difficult if the snare drum (and/or horn stabs and guitar solos) is loud in the music mix, and there's no room for the voice-over.

Needless to say, in a commercial the AVO is considered more important than the music. The entire music track could end up barely audible as a result of the music engineers "hot tracks."

3. *Know where the tracks are going next.* If the post session is a film mix, it is a good idea to send out the music master with sync tone instead of SMPTE, because not all film mix houses are able to resolve from SMPTE.

There always seems to be audible cross talk on 2-track machines and the sync tone (which does not sound the same as ground loop buzz) is less annoying than high frequency SMPTE. Check with the transfer house before the music mix.

Also, if you know where the tape is going you can find out how the machines are aligned (+6dB/185nW/m for Ampex 456 or 3M 226). In the real world, because of laziness or a "good enough" attitude, the only tone likely to be checked is 1k. Remember that the right way is not your way, but the way of the next person in the chain who will have an effect on the final product.

One dramatic example happened while I was working on stereo FM radio campaign. I learned that some of the regional duplication centers had source machines that were only capable of playing back channel 1. It never even occurred to them that someone would send a stereo half-track. The result was a near-disaster as the product name in the tag was hard-panned.

It often seems to outsiders that commercial music production is an engineer's fantasy. Big budgets, top session players (who show up on time), real string sections and expensive designer-equipped studios with

the latest gear. But from a music engineer's point of view, there is often frustration. Most mixes are still mono and digital technology is rarely used.

The music engineer, looking ahead, sees his mix going to post-production facilities that typically spend more money on one piece of fancy video equipment than on entire audio rooms. In fact, many post facilities are still using archaic audio equipment (witness the continued use of film mag for video). Why is quality and technology so often suppressed?

Blame it on the agency? We all have stories about the agency agonizing over a sound for hours at a music mix and then seemingly leaving it out at the post mix. (Or the classic example of the ad execs "favorite track.") The client liked a part so much at the music session that he wanted a stripe to play with at the film mix. Unfortunately, during the film mix, he decided to use it twice: before *and after* the modulation!

Why isn't the audio quality controlled throughout the process the same way the picture is? Often, by the time the music is even considered for a (post score) spot the agency personnel have been living with the "concept" for a long time. They have already spent thousands of dollars, and in some cases staked their professional reputations. Jobs are clearly on the line. Yet when it comes to audio, agency personnel seem unaware of, or unconcerned with the available options.

Isn't that why they hire us?

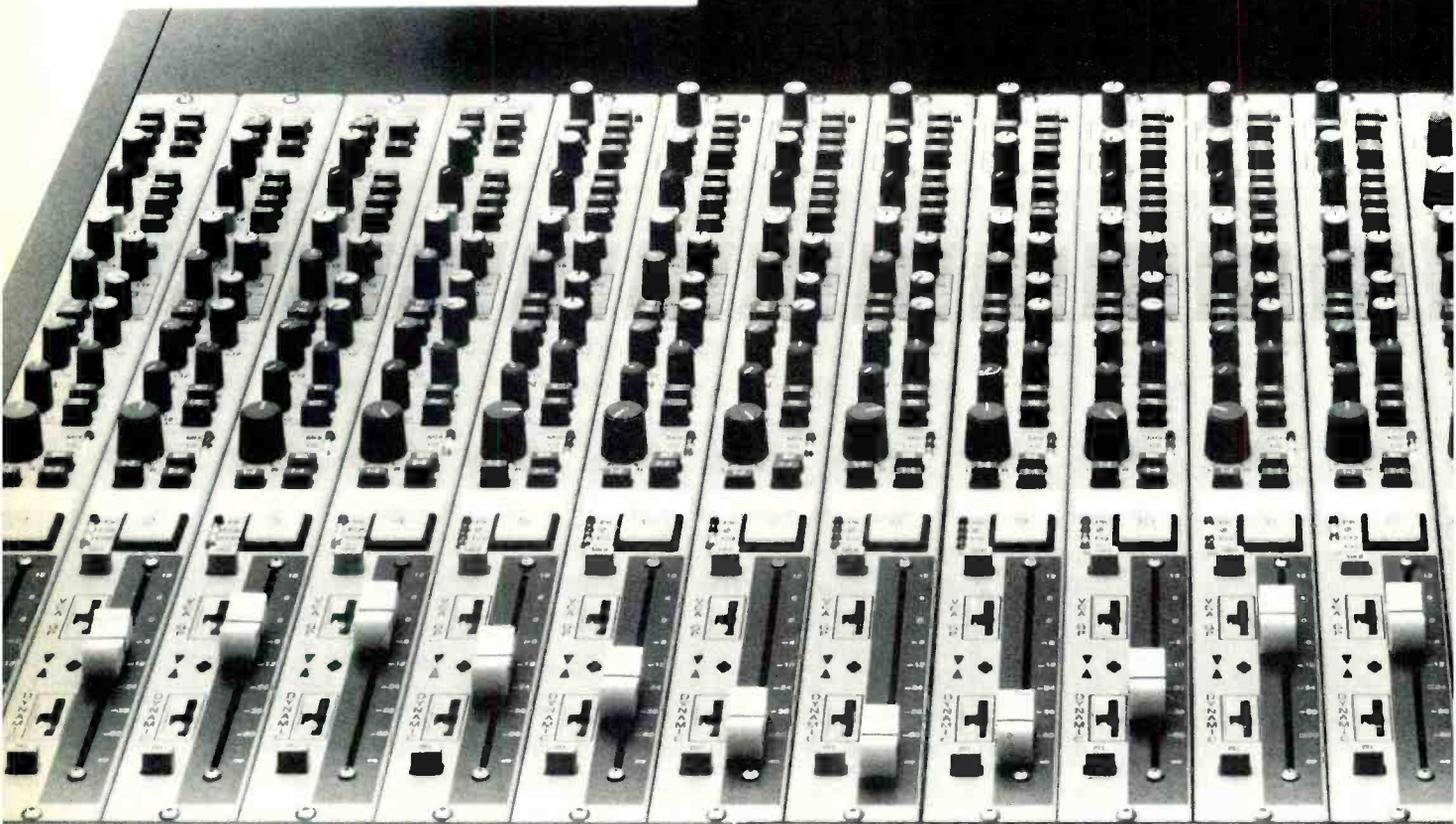
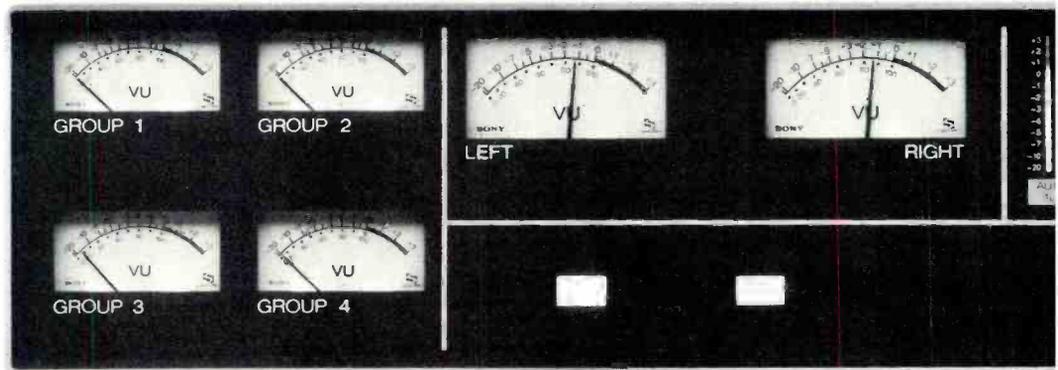
It is the responsibility of the engineer to get the best sound on tape. Included in this is a responsibility to inform clients about beneficial changes in technology. Engineers who like to complain that quality audio is not appreciated, must take some responsibility for not adequately convincing their clients of its benefits. The kind of pressure mentioned above makes it very difficult to push the boundaries of quality and creativity—but it also makes the demand for change and a competitive edge very loud.

Why don't advertisers insist on digital recording and stereo broadcast networks? Certainly considering the commercial industry finances television, they must have some power to persuade. After all, in this business, if the client wants fresh-squeezed pink grapefruit juice at 38.6 degrees, they get it.

REP

The author would like to thank Bill Ivie, chief mixer at National Video Center, New York, for his help.

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$$c(t) = 1087.42 \sqrt{1 + t/273.16}$$

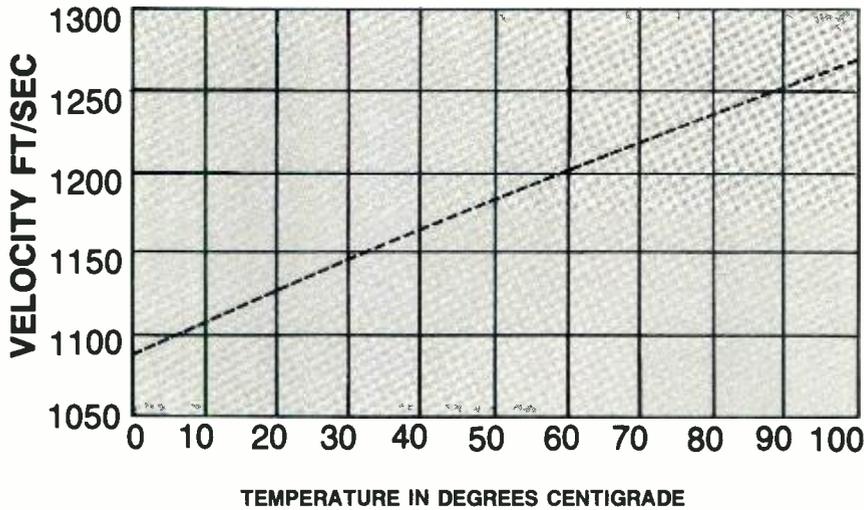


Figure 2. Graph of Speed of Sound in ft/sec. vs. temperature.

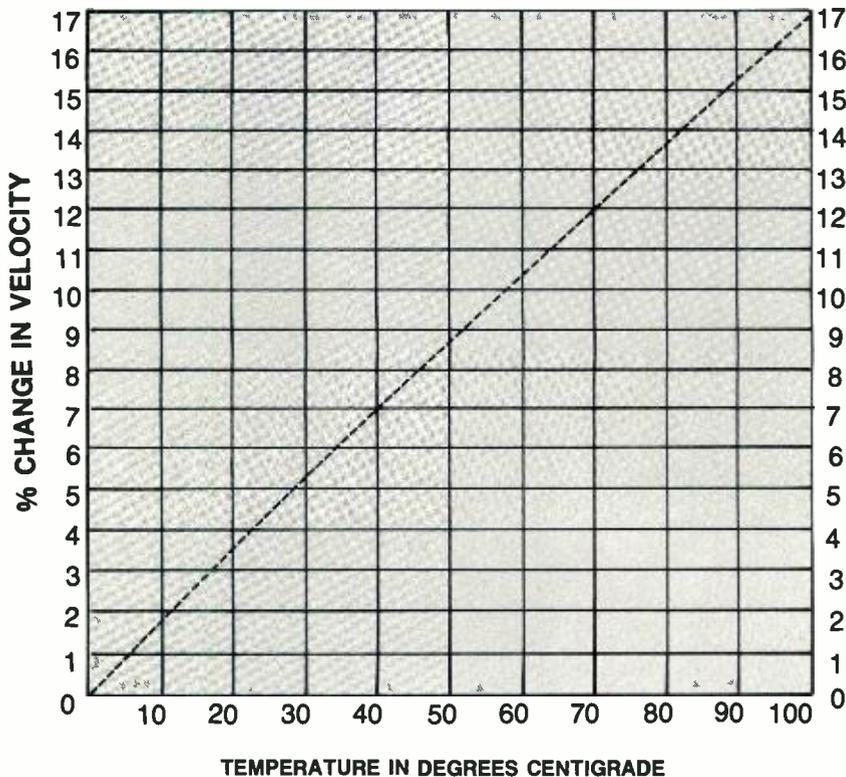


Figure 3. Graph of temperature vs. percentage change in the speed of sound (re 0°C) in dry air.

In one sense, there is nothing new here. The major effects described and equations presented all exist within published books on acoustics. Some, from the *Journal of the Acoustical Society of America*, are 45 years old. However, as familiar as many members of that learned group may be with the information, unfortunately very few working sound reinforcement professionals are members of that society. This article is for those who are in the trenches every day and need all the assistance they can get.

What's new

What's new is the table and graphical treatment of the material. Everything available regarding temperature and humidity effects on the speed of sound appears in this new form, as does the material on sound absorption. Experience shows tabulated and graphed data to be more useful than equations. Practical applications require concise, easily checked, look-up facts.

Before presenting the detailed analyses, a question should be answered: Why bother? This is not a facetious question. Many professionals realize that sound velocity depends upon temperature, barometric pressure, relative humidity, altitude, air composition and so on. The problem is that somewhere they learned they may ignore these effects—make the assumption that they are not significant.

Well, 30 years ago, I may have agreed with you. Thirty years ago, we were just beginning to understand what room response meant, much less do anything about it. Then, we developed ways to view and alter room responses. Graphic equalizers and real-time analyzers opened up a whole new window of opportunity for improving reinforced audio.

Progress continued slowly until Richard Heyser gave us time delay spectrometry (TDS). We then experienced one of those step function jumps in our ability to view our acoustic environment. For the first time, we could actually see what a mess we had been dealing with all along!

Today, we have a whole new army attacking room problems with a vengeance. Racks of equalizers and delay units arm those combatants as they wage war on all

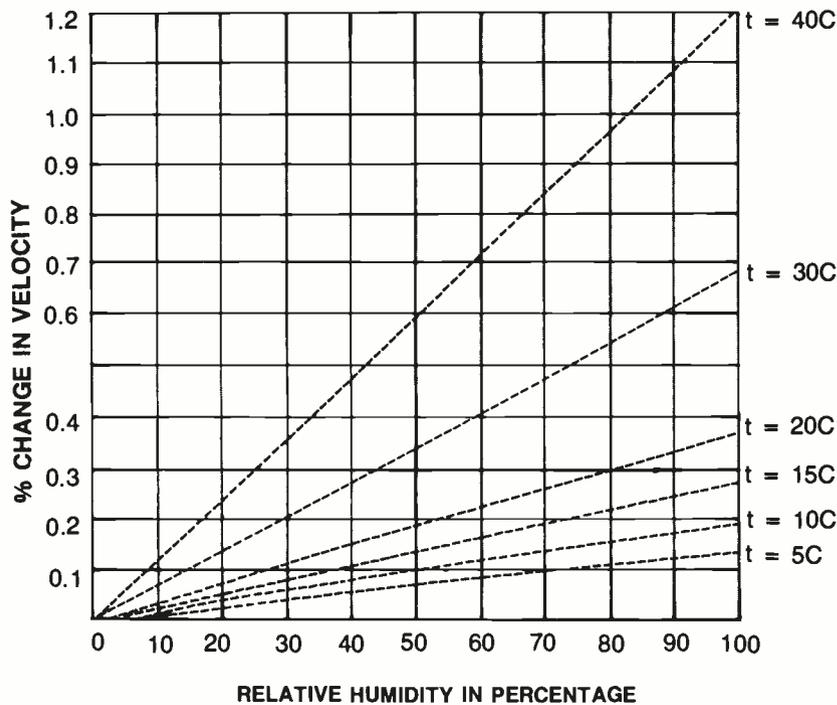


Figure 4. Graph of relative humidity vs. percentage change in the speed of sound as a function of temperature.

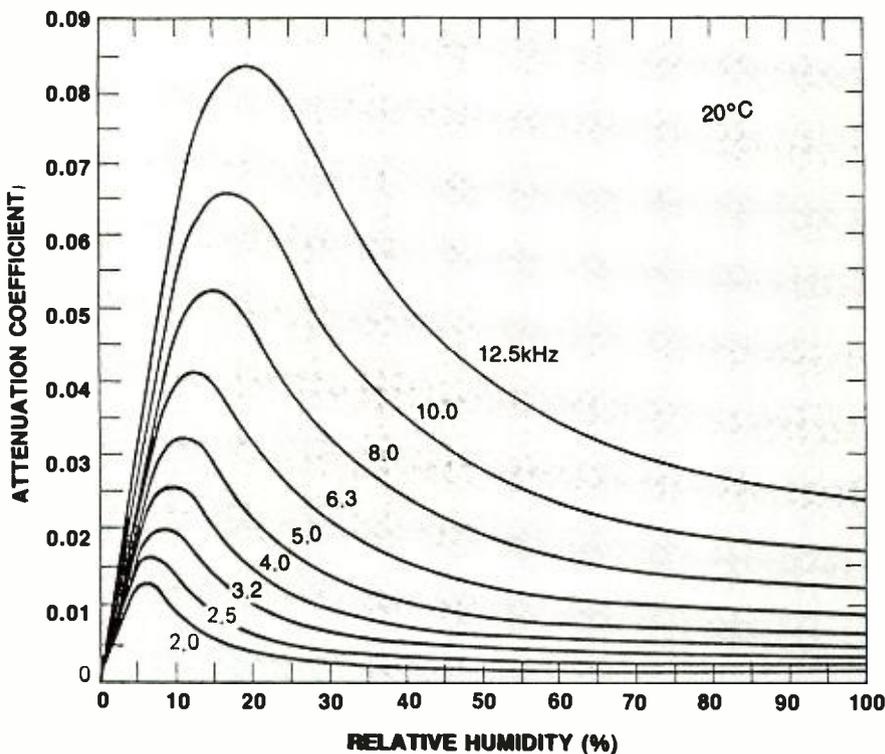


Figure 5. Values of the total attenuation coefficient m vs. relative humidity in percentage for air at 20°C as a function of frequency. [From C.M. Harris, *J. Acou. Soc. Am.*, vol. 40, p. 148 (1966).]

those response peaks and valleys. Each year, they demand finer equalization tools and smaller delay increments with which to continue the fight. All this is fine. Only, *don't forget Mother Nature.*

TDS-based test equipment allows us to see far more than is probably good for us. And there is a natural tendency to fix something if we can see it—without regard to relevancy.

The point here is that tight control of temperature and relative humidity *must* accompany use of very small time delay increments to fix room response problems. Perhaps an example best illustrates the importance of controlling the environment for sound systems.

An example

For this example I will jump ahead and use data from the various graphs and tables presented. I hope this approach will encourage you to do the same.

This simple example does not even require diagrams. Consider a listening spot positioned so that the direct sound must travel 50 feet to the listener. This same spot receives one reflected arrival that travels 140 feet, say 70 feet to a side wall and another 70 feet back to the listener's ear. Ignore all other delayed arrivals. The reflected wave arrives with some sort of phase relationship to the direct wave. This relationship is a function of the distance traveled, the frequency involved and the speed of sound.

Assume the room temperature was 20°C with 30% relative humidity when the measurements were taken. Table 3 shows the velocity of sound is 3.71% faster than the reference standard velocity (1087.42 ft/s). Using a test tone of 10kHz, calculate the following information:

Velocity of sound: $1087.42 \times 1.0371 = 1127.763$ ft/s.

Wavelength: $(1127.763)/(10\text{kHz}) = .1127763$ ft.

No. of cycles traveled for 50 feet: $(50)/(.1127763) = 443.36$

No. of cycles traveled for 140 feet: $(140)/(.1127763) = 1241.40$

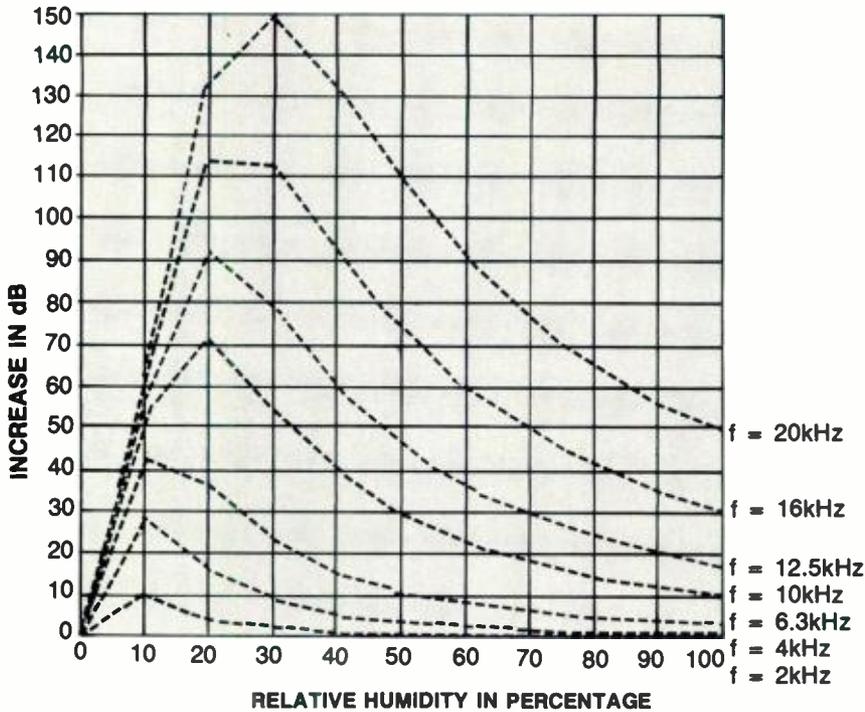


Figure 6. Graph of sound absorption increase in dB/1000 ft vs. relative humidity as a function of frequency at 20°C.

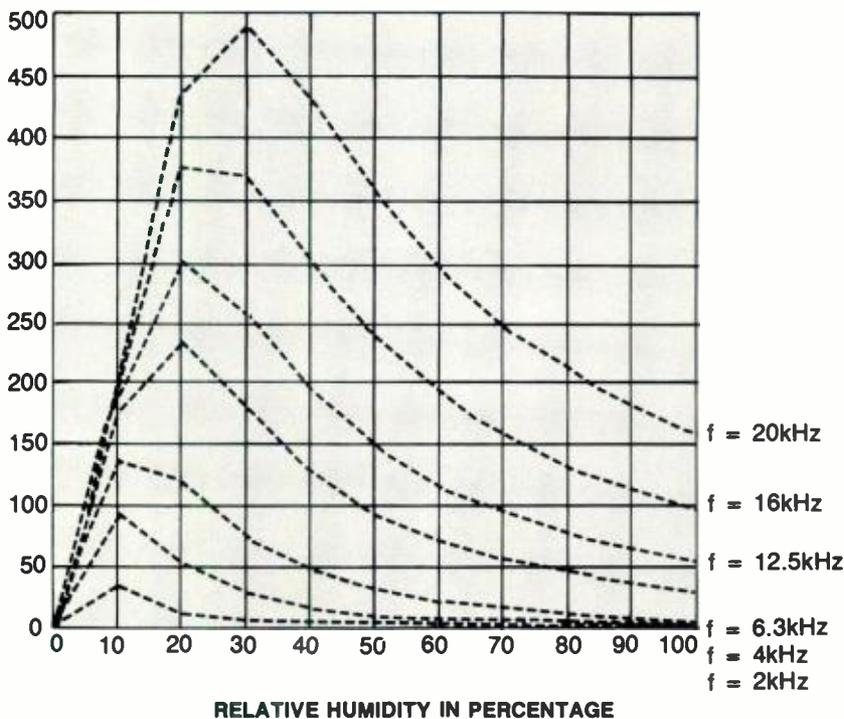


Figure 7. Graph of sound absorption increase in dB/km vs. relative humidity as a function of frequency at temperature equal 20°C.

For purposes of this example, the only point of interest is the decimal fractions of a cycle. For all practical purposes, the two waves are in phase (.36 cycles vs. .40 cycles). That is, the delayed and attenuated reflected wave arrives essentially in phase. So, the two waves will add. A little equalization easily corrects this bump and everyone is happy.

Until the environment changes. Assume the temperature rises to 30°C with 80% relative humidity. Again, checking Table 3 shows the velocity of sound now is 5.9% faster than standard. The casual observer mistakenly figures its only a difference of 2.19%, so there is no problem. *The casual observer is wrong.*

Recalculation gives:

Velocity of sound: $1087.42 \times 1.059 = 1151.578$ ft/s.
Wavelength: $(1151.578)/(10\text{kHz}) = .1151578\text{ft}$.
No. of cycles traveled for 50 feet: $(50)/(.1151578) = 434.19$
No. of cycles traveled for 140 feet: $(140)/(.1151578) = 1215.72$

OK, the velocity of sound increased. This creates a longer wavelength. So, traveling the same distances takes fewer cycles. Nothing too interesting yet. However, careful examination of the two decimal fractions of a cycle reveals they are essentially *out of phase*. The difference between them is .53 cycles, or about 180°. Even to the casual observer, this is not good. The applied equalization is now in the wrong direction.

The above example illustrates the fallacy of thinking you can ignore velocity changes because they affect direct and reflected waves equally. This simply is *not true*.

Complicating things further is the change in absorption due to the change in relative humidity. Table 6 and Figure 6 show a drop of 39dB/1,000 feet, due to the increased relative humidity (ignoring the temperature effects of 30°C). Because the example involves a distance of 140 feet, there is 5.46dB less absorption. So, not only does the signal arrive out of phase, it

is about 5.5dB bigger!

The point of all this is that even a small percentage change in the speed of sound can have disastrous effects on a sound system. Often overlooked is that *the small percentage change is for every cycle undergone by the wave*. It is a trap to think of the change as only a few percent and dismiss it.

Think of the hundreds, nay thousands, of cycles existing within any room. Each one has its wavelength altered by this percentage. If a 1% change affects hundreds of cycles, it alters the acoustics of the whole system. No wonder all those hours spent equalizing are sometimes in vain. Much work lies ahead in understanding how to control environmental effects so room equalization, once done, will remain satisfactory for prolonged periods.

Historical Background¹

Investigation into the nature of sound dates back to earliest recorded history. Indeed, ancient writings show Aristotle (384-322 B.C.) observed two things regarding sound: First, that the propagation of sound involved the motion of the air; and second, that high notes travel faster than low notes. (Batting .500 isn't too bad for the ancient leagues!)

Since in the transmission of sound air does not appear to move, it is not surprising that philosophers later denied Aristotle's view. Denials continued until 1660, when Robert Boyle in England definitely concluded that air is one medium for acoustic transmission.

The next question was, how fast did sound travel?

As early as 1635, Pierre Gassendi, while in Paris, made measurements of the velocity of sound in air. His value was 1,473 Paris feet per second. (The Paris foot is equivalent approximately to 32.48cm.) Later, Marin Mersenne (1588-1648), a French natural philosopher often referred to as the "father of acoustics," corrected this to 1,380 Paris feet per second, or about 450 meters per second. Gassendi also demonstrated conclusively that velocity is independent of frequency, thus forever discrediting Aristotle's view.

In 1656, Borelli and his colleague Viviani (both Italian) made a very careful measurement and obtained 1,077 Paris feet per second, or 350 m/s. It is clear all these values suffer from the lack of reference to the temperature, humidity and wind velocity conditions.

It was not until 1740 that Branconi (also Italian) showed definitely that the velocity of sound in air increases with temperature. This was two years after the French gave us our first good velocity figure.

The first measurement judged precise in the modern sense occurred under the di-

rection of the Academy of Sciences in Paris in 1738, using cannon fire. When reduced to 0°C, the results were 332m/s. A rather remarkable feat considering that careful repetitions during the rest of the 18th century and the first half of the 19th century gave results differing by only a few meters per second from this value. And 200 years later, the best modern value² recorded was 331.45 ± 0.05 m/s in still dry air under standard conditions of temperature and pressure (0C and 76 cm

of Hg pressure)—a scant 0.5 m/s difference from the French value.

Laplace was the first to show why temperature was important. He suggested that in all prior calculations errors occurred due to the assumption that the elastic motions of the air particles take place at constant temperature (isothermal law). In view of the rapidity of the motions, he reasoned that the gas molecules experience a small change in temperature. In 1816, he demonstrated that the compres-

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Table 1. Velocity of sound in dry air vs. temperature.

Temp °C	Temp °F	Velocity m/sec	Velocity ft/sec	Temp °C	Temp °F	Velocity m/sec	Velocity ft/sec
0	32.0	331.45	1087.42	21	69.8	343.96	1128.46
1	33.8	332.06	1089.42	22	71.6	344.54	1130.37
2	35.6	332.66	1091.39	23	73.4	345.12	1132.27
3	37.4	333.27	1093.39	24	75.2	345.71	1134.20
4	39.2	333.87	1095.36	25	77.0	346.29	1136.11
5	41.0	334.47	1097.33	26	78.8	346.87	1138.01
6	42.8	335.07	1099.30	27	80.6	347.45	1139.91
7	44.6	335.67	1101.26	28	82.4	348.02	1141.78
8	46.4	336.27	1103.23	29	84.2	348.60	1143.69
9	48.2	336.87	1105.20	30	86.0	349.18	1145.59
10	50.0	337.46	1107.14	31	87.8	349.75	1147.46
11	51.8	338.06	1109.11	32	89.6	350.33	1149.36
12	53.6	338.65	1111.04	33	91.4	350.90	1151.23
13	55.4	339.25	1113.01	34	93.2	351.48	1153.13
14	57.2	339.84	1114.95	35	95.0	352.05	1155.00
15	59.0	340.43	1116.88	36	96.8	352.62	1156.87
16	60.8	341.02	1118.82	37	98.6	353.19	1158.74
17	62.6	341.61	1120.75	38	100.4	353.76	1160.61
18	64.4	342.20	1122.69	39	102.2	354.32	1162.45
19	66.2	342.78	1124.59	40	104.0	354.89	1164.32
20	68.0	343.37	1126.53				

Table 2. Percentage increase in speed of sound (re 0°C) due to moisture in air (only). Temperature effects not included except as they pertain to humidity

Temp °C	RELATIVE HUMIDITY IN %									
	10	20	30	40	50	60	70	80	90	100
5	.014	.028	.042	.056	.070	.083	.097	.111	.125	.139
10	.020	.039	.059	.078	.098	.118	.137	.157	.176	.196
15	.027	.054	.082	.109	.136	.163	.191	.218	.245	.273
20	.037	.075	.112	.149	.187	.224	.262	.299	.337	.375
30	.068	.135	.203	.272	.340	.408	.477	.546	.615	.684
40	.118	.236	.355	.474	.594	.714	.835	.957	1.08	1.20

Table 3. Total % increase in speed of sound (re 0°C) due to temperature and humidity combined.

Temp °C	RELATIVE HUMIDITY IN %					
	0	30	40	50	80	100
5	0.91	.952	.966	.980	1.02	1.05
10	1.81	1.87	1.89	1.91	1.97	2.01
15	2.71	2.79	2.82	2.85	2.93	2.98
20	3.60	3.71	3.75	3.79	3.90	3.98
30	5.35	5.55	5.62	5.69	5.90	6.03
40	7.07	7.43	7.54	7.66	8.03	8.27

sions and rarefactions did not follow the isothermal law, but instead follow the *adiabatic law*, in which the changes in temperature lead to a higher value of the elasticity. (Adiabatic refers to change in which there is no gain or loss of heat.)

Elasticity is the product of the pressure by the ratio of the two specific heats of the air. The ratio of the specific heats is symbolized by the lower case Greek letter gamma. Laplace originally used results by LaRoche and Berard giving gamma=1.50. His results were off from the measured velocity, but not enough to discourage the theory. Later in his chapter on the velocity of sound in his "Mechanique Celeste" in 1825, he used the accurately measured value of gamma=1.35 by Clement and Desormes (1819). The revised calculations agreed very closely with experimental results. Some years later, the revised value of gamma=1.40 led to complete agreement with the measured velocity.

The Laplace theory is so well established that it is now common practice to work backward to determine gamma for various gases by precise measurements of the velocity of sound in the medium.

Temperature and humidity effects

[Author's note: All data presented are based upon results published in the "CRC Handbook of Chemistry and Physics," 67th edition (CRC Press Inc., Boca Raton, FL, 1986), and H.C. Hardy, D. Telefair, and W.H. Pielemeier's definitive paper, "The Velocity of Sound in Air," J. Acou. Soc. Am., vol. 13, pg. 226-233 (Jan. 1942).]

General equations

The theoretical expression for the speed of sound, c, in an ideal gas is:

$$c = \sqrt{\delta P/p} \quad (1)$$

where P is the ambient pressure, p is the gas density and δ^3 is the ratio of specific heat of gas at constant pressure to that at constant volume.

The term δ (ratio of specific heats) is dependent upon the number of degrees of freedom of the gaseous molecule. The number of degrees of freedom depends upon the complexity of the molecule.

Since air is composed primarily of diatomic molecules, the speed of sound in air is

$$c = \sqrt{1.4 P/p} \quad (2)$$

The velocity of sound, c , in dry air has the following experimentally verified value:

$$c = 331.45 \pm 0.05 \text{ m/s} \quad (3)$$

or

$$c = 1087.45 \pm 0.16 \text{ ft/s} \quad (4)$$

for audio frequencies, at 0°C and one atmosphere (760mm Hg).

Temperature dependence

Substituting the equation of state of air of an ideal gas ($PV=RT$) and the definition of density, p , (mass per unit volume), Equation 2 may be written as

$$c = \sqrt{1.4RT/M} \quad (5)$$

where R is the universal gas constant, T the absolute temperature and M the mean molecular weight of the gas at sea level.

Equation 5 reveals the temperature dependence and pressure independence of the speed of sound. An increase in pressure results in an equal increase in density. Therefore, there is no change in velocity due to a change in pressure. But this is true only if the temperature remains constant. Temperature changes cause density changes which do not affect pressure. Thus, density is not a two-way street. Changes in pressure affect density but not vice-versa. Humidity also affects density causing changes in the velocity of sound.

Because R and M are constants, the speed of sound may be shown to have a first order dependence on temperature as follows:

$$c := C_0 \sqrt{T/273} \quad (6)$$

where T is the temperature in degrees Kelvin and C_0 equals the reference speed of sound under defined conditions.

The speed of sound is seen to increase as the square root of the absolute temperature. Substituting centigrade conversion factors and the reference speed of sound gives

$$c := 331.45 \sqrt{1+t/273} \text{ m/s} \quad (7)$$

or

$$c := 1087.42 \sqrt{1+t/273} \text{ ft/s} \quad (8)$$

where t is the temperature in degrees centigrade.

Graphs of Equations 7 and 8 appear as Figures 1 and 2 respectively. Table 1 gives tabulated results for Equations 7 and 8. A more useful presentation of this data is

shown in Figure 3. Figure 3 graphs the percentage increase in the speed of sound due to temperature.

Humidity dependence

All previous discussion assumed dry air. Attention turns now to the effects of moisture on the speed of sound. Moisture affects the density of air, and hence, from Equation 1, the speed of sound in air. Moist air is less dense than dry air (not particu-

larly obvious), so p in Equation 1 gets smaller. This causes an increase in the speed of sound. Moisture also causes the specific-heat ratio to decrease, which would cause the speed of sound to decrease. However, the decrease in density dominates so the speed of sound increases with increasing moisture.

The literature is painfully lacking in practical specific treatments on the correlation between relative humidity and

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Table 4. Total sound absorption in dB/1000 ft vs. relative humidity as a function of frequency at 20°C (68°F).

Freq. Hz	RELATIVE HUMIDITY IN %										
	0	10	20	30	40	50	60	70	80	90	100
2k	1.26	11.7	5.31	3.33	2.54	2.18	2.00	1.92	1.89	1.89	1.92
4k	2.70	31.0	19.0	11.9	8.52	6.75	5.71	5.06	4.68	4.34	4.14
6.3k	4.54	47.1	41.2	27.6	20.0	15.6	13.0	11.2	9.98	9.10	8.45
10k	8.01	6.16	79.7	62.5	47.4	37.5	31.0	26.6	23.5	21.1	19.4
12.5k	10.9	68.1	103	89.7	70.9	57.0	47.5	40.8	35.9	32.3	29.5
16k	15.9	76.2	130	129	108	89.6	75.5	65.2	57.6	51.8	47.2
20k	23.0	85.6	156	172	155	133	114	99.4	88.1	79.4	72.5

Table 5. Total sound absorption in dB/km vs. relative humidity as a function of frequency at 20°C (68°F).

Freq. Hz	RELATIVE HUMIDITY IN %										
	0	10	20	30	40	50	60	70	80	90	100
2k	4.14	38.2	17.4	10.9	8.34	7.14	6.55	6.28	6.19	6.21	6.29
4k	8.84	102	62.3	38.9	28.0	22.2	18.7	16.6	15.2	14.2	13.6
6.3k	14.9	154	135	90.6	65.6	51.3	42.5	36.7	32.7	29.8	27.7
10k	26.3	202	261	205	155	123	102	87.3	77.0	69.3	63.5
12.5k	35.8	224	338	294	232	187	156	134	118	106	96.6
16k	52.2	250	428	423	355	294	248	214	189	170	155
20k	75.4	281	511	564	508	435	374	326	289	261	238

Table 6. Increase in sound absorption (dB/1000 ft) due to relative humidity as a function of frequency at temperature equal to 20°C (68°F).

Freq. Hz	RELATIVE HUMIDITY IN %										
	0	10	20	30	40	50	60	70	80	90	100
2k	13.9	10.4	4.05	2.07	1.28	0.92	0.74	0.66	0.63	0.63	0.66
4k	20.0	28.3	16.3	9.20	5.82	4.05	3.01	2.36	1.93	1.64	1.44
6.3k	21.7	42.6	36.7	23.1	15.5	11.1	8.46	6.66	5.44	4.56	3.91
10k	22.8	53.6	71.7	54.5	39.4	29.5	23.0	18.6	15.5	13.1	11.4
12.5k	23.4	57.2	92.1	78.8	60.0	46.1	36.6	29.9	25.0	21.4	18.6
16k	24.4	60.3	114	113	92.1	73.7	59.6	49.3	41.7	35.9	31.3
20k	25.7	62.6	133	149	132	110	91.0	76.4	65.1	56.4	49.5

Table 7. Increase in sound absorption (dB/km) due to relative humidity as a function of frequency at temperature equal to 20°C (68°F).

Freq. Hz	RELATIVE HUMIDITY IN %										
	0	10	20	30	40	50	60	70	80	90	100
2k	45.7	34.1	13.3	6.76	4.20	3.00	2.41	2.14	2.05	2.07	2.15
4k	65.6	93.2	53.5	30.1	19.2	13.4	9.86	7.76	6.36	5.36	4.76
6.3k	71.2	139	120	75.7	50.7	36.4	27.6	21.8	17.8	14.9	12.8
10k	74.7	176	235	179	129	96.7	75.7	61.0	50.7	43.0	37.2
12.5k	77.2	188	302	258	196	151	120	98.2	82.2	70.2	60.8
16k	79.8	198	376	371	303	242	196	162	137	118	103
20k	84.6	206	436	489	433	360	299	251	214	186	163

sound speed. And Hardy, et al., warn of the many inexact expressions existing in the textbooks, handbooks and tables for the change in sound speed due to moisture. A rigorous analysis does exist by Pierce⁴ and will be used to develop a useful and accurate graph and table directly relating relative humidity to the percentage increase in the speed of sound.

Figure 4 plots the percentage increase in sound speed due to relative humidity only, the temperature values are for accurately specifying the relative humidity. Table 2 gives calculated results.

Combined effects

The results graphed in Figures 3 and 4, and also shown tabulated in Tables 1 and 2, can be added together to show the combined effects of temperature and relative humidity on the speed of sound. Doing so produces Table 3. Here the total percentage increase in sound speed is tabulated for easy reference.

Effect of relative humidity on the absorption of sound

To a certain degree, everything absorbs sound, especially air. And wet air absorbs sound better than dry air. This section presents the latest findings on the absorption of sound in air.

Air absorption

Sound propagates through air as a wave in an elastic medium. Because air is not a perfectly elastic medium, this pulsating action causes several complex irreversible processes to occur. The wave action of air causes minute turbulence of the air molecules it passes through. Each affected molecule robs the wave of some of its energy until eventually the wave dies completely. If this were not so, every sound generated would travel forever and we would live within a sonic shell of cacophony.

Absorption works with divergence. Divergence of sound causes a reduction in sound intensity due to spreading of the wave throughout the medium. The sound pressure level will decrease 6dB for each doubling of the distance, i.e., it is inversely proportional to the square of the distance. This well-known fact occurs simultaneously with absorption. Absorption describes the energy exchanging mechanism occurring during divergence. So not only is the wave spreading, it is also dying.

Air absorption mathematics

The strict confines of the ideal fluid-dynamic equations cannot explain attenuation of sound. Theoretical predictions must include bulk viscosity, thermal conduction and molecular relaxation for agreement with measured results. Conservation of mass, entropy for the gas and

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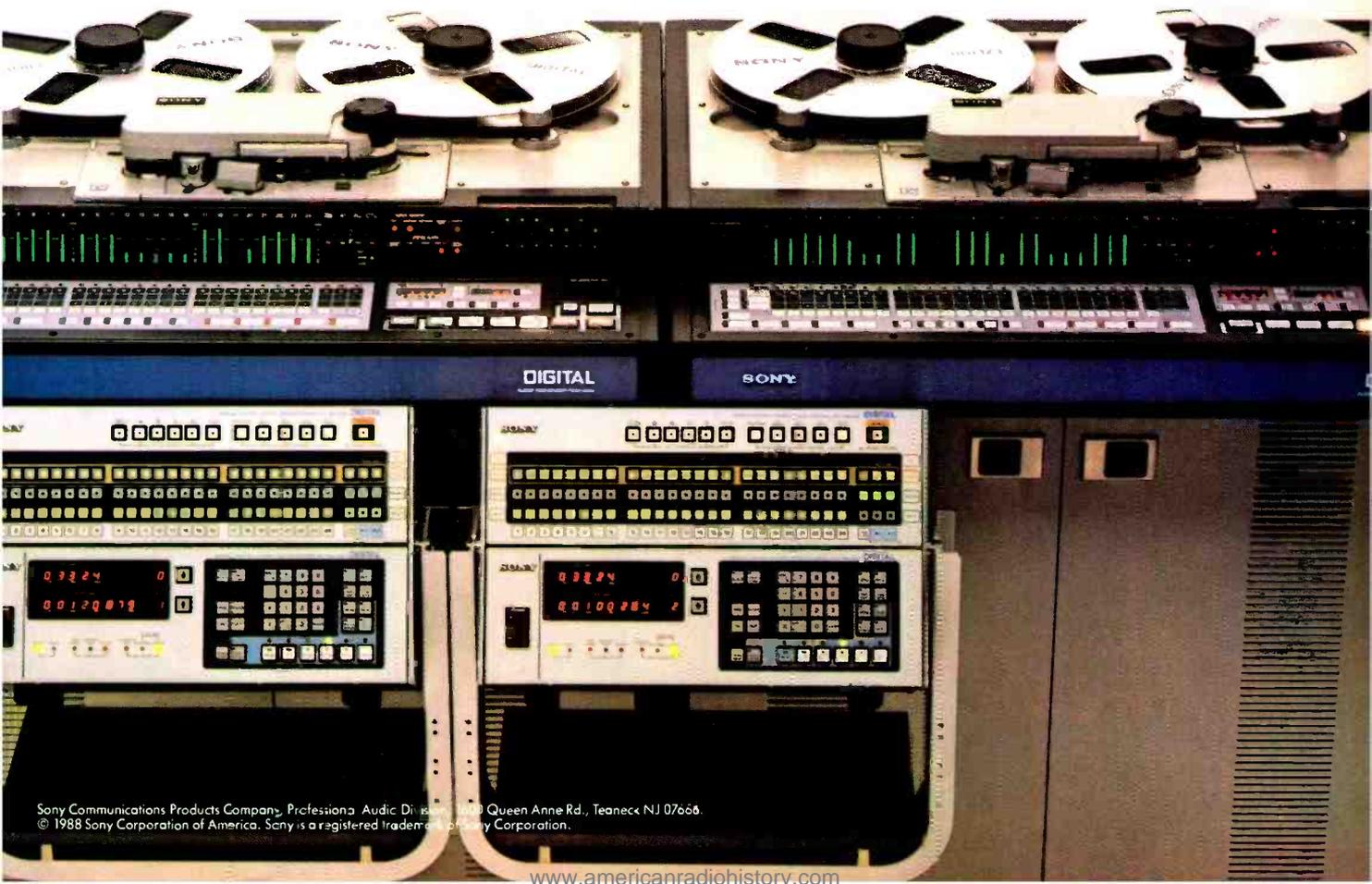
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Equipment Rental:

Fantasy and Reality

By Tom Behrens

The key to successful rentals is researching your equipment needs and stating them clearly to the rental company.

To begin, a story:

Arnie waited in his seat until the last of the passengers exited the plane. He had been fidgeting throughout the whole flight, wondering what horrors awaited him at the studio.

His last project approached near-disaster, with looking for cables at the last minute and that so well-timed "intermittent" problem with the vocal limiter. The only time it worked was when the tech finally got around to actually listening to it. And it was a sure bet that it would start acting up again during the best vocal take.

Arnie shivered. "Maybe this new brokerage service will iron things out. . ." He paused. "Naah!"

Back in Los Angeles, Arnie's secretary, Rita, had just finished dialing the toll-free number for Equipment Referrals, the worldwide rental network. Arnie had called the company for all the equipment he would need. She gave Jean, ER's customer representative, the confirmation number and waited.

Jean turned to her terminal, which was on an open link to an electronic mail network, and she executed a search routine through their password-protected database to find Arnie's order. A few seconds later, she had Arnie's complete itinerary on the screen.

"OK, Rita, I have one digital 32-track for three weeks, a PD format digital 2-track for three days for mixdown. I also show

Tom Behrens is vice president and co-owner of Equipment Pool, an equipment rental company based in Nashville.

ice industry. Most people forget that notion because they are renting something tangible, a piece of equipment. Accompanying the rental of the equipment are many intangibles, including years of experience, a broad technical knowledge and expertise in interfacing a wide variety of equipment.

Most rental companies take a great deal of pride in providing a high level of technical support to the industry, allowing recording studios to streamline their inventory and invest available resources more effectively. This simply means that the studio is not required to have a significant investment in equipment to fulfill a job requirement. Instead, the rental company provides short-term access to a large inventory without the big initial investment.

In return for this level of service and commitment to inventory, the rental company expects to be able to make a reasonable return on its substantial investment. Multiple-day discounts are offered as an incentive for you to keep the equipment for longer periods of time. As the rental term increases, the cost of doing business decreases.

Most rental companies will turn that savings around to you in the form of a long-term discount. It simply costs more for a rental company to rent on the shorter term, so don't expect discounts for one- and two-day rentals, especially if the job was originally quoted as a two- or three-week rental. In return for giving an established rental company the opportunity to profit from your business, the renter should expect to receive at least the following:

1. Courteous attitude and intelligent responses to your questions (no matter how dumb you think they are).
2. Knowledgeable personnel you can rely on for experience, suggestions and solutions to your individual job requirements.
3. All necessary or requested interface cables.
4. Complete technical support for the equipment you are renting and its interface.
5. Equipment in good working order.
6. Quick response to problems.
7. Prompt delivery.
8. Setup and verification of operation on major pieces.

Basically, it comes down to communication. If you and your rental company completely understand each other and the job you are attempting to do, you should attain success and profit together. This will lay the groundwork for a mutually beneficial relationship for both of you, now and in the future.

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A Bits Eye View of Digital Signal Processing

By Richard C. Cabot, Ph.D., P.E.

Signal processing is an inexact science that requires the use of various techniques that often achieve an only adequate response.

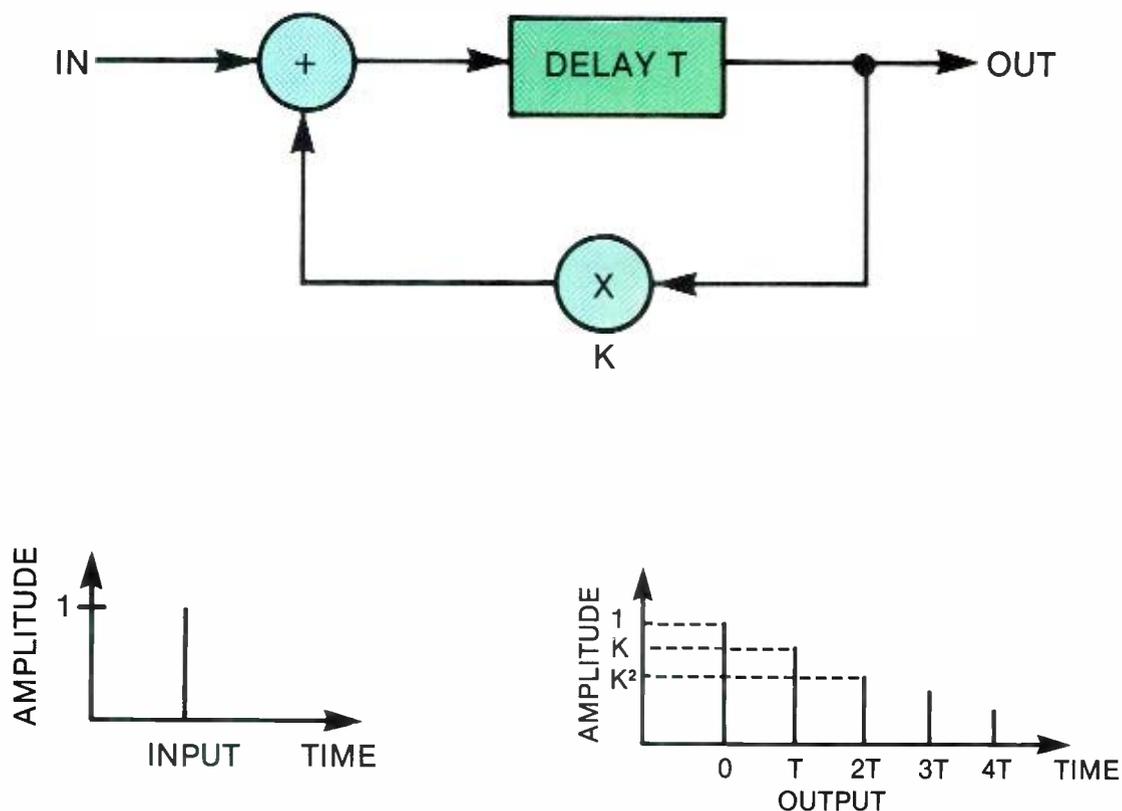


Figure 1. A simple single-stage reverberator.

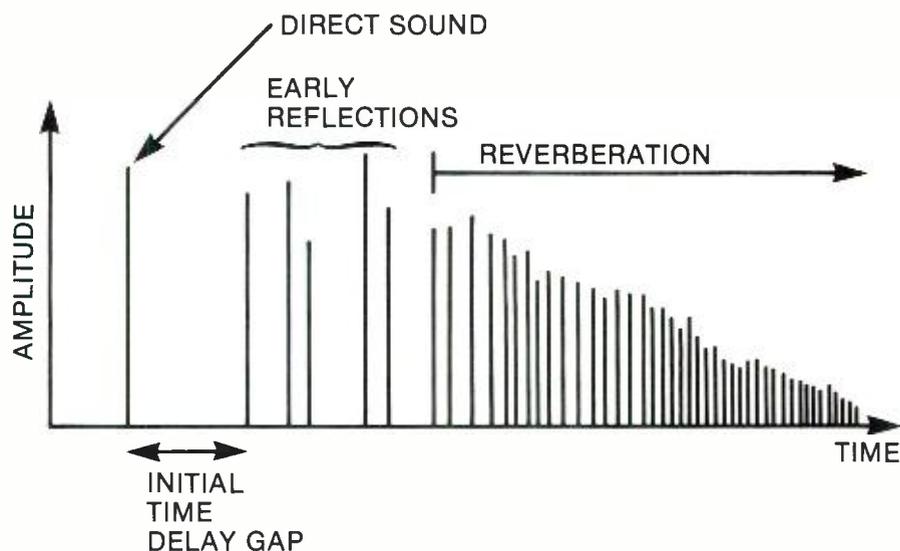


Figure 3. A simplified room impulse response.

Although the steady state frequency response of all-pass reverberators is flat, the sound is still not very natural. This is because the echo density is low and the spacing is very regular, that is, there aren't many echoes and they are all the same amount of time apart. Cascading several all-pass reverberators will improve matters but is not a cure-all.

As with the original reverberator design,

the delay time of each cascaded stage must be set to a different value. The selection of delay times for optimum uniformity of reflection timing requires all times to be prime number ratios so that none of the reflections will occur simultaneously. This is true also for other schemes with multiple-tapped delays.

The next step in the evolution of reverberators was to combine the generic re-

verberator schemes described above with specific early reflection patterns. This becomes easy when using a software-based reverberator. The early reflections tend to give the reverberation the character of a particular room or hall. The impulse response of Figure 3 shows the pattern of reverberation in a hypothetical room.

The direct sound from the source to the listener is followed by the first reflection

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Vic & Linda Radulich,
Digital Post, Los Angeles.

 "On our latest film, 'Lords of Magic', we recorded all of our production sound digitally. The AudioFile is used to handle the dialogue and music editing and will be used as a playback source in the final mix."
David Marsh,
Marsh Films, Los Angeles.

 "Commercial production forms the bulk of our business. The AudioFile has proven to be easy to use and now makes it possible for us to realise our goal of digital audio from start to finish."
Jay Scott,
Producers Color Service, Detroit.

 "When we took delivery of our AudioFile, we got it out of the box, powered it up, and did a project with it, it really is that simple."
John Wiggins,
HBO Productions, New York City

 "Over the years we've built up a very comprehensive digital audio effects library and we're now building two complete new rooms, each equipped with an AudioFile to get the very best results when laying audio to picture."
Wylie Stateman & Lon Bender,
Soundelux, Los Angeles.

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Editel, Chicago.

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Hank Newberger, & Tim Butler
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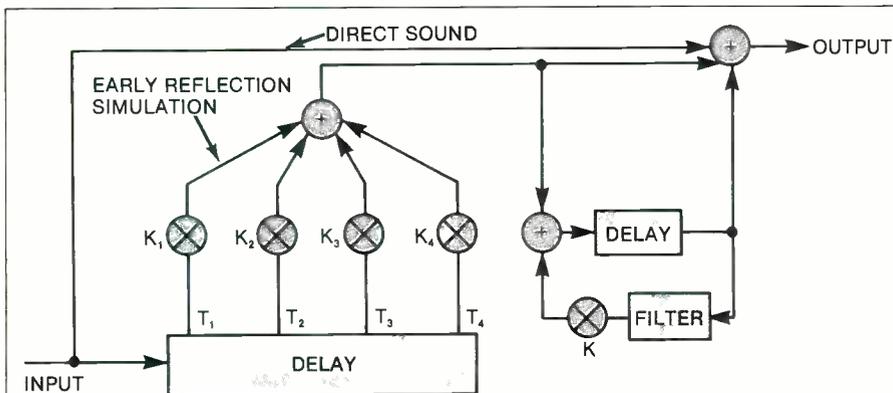


Figure 4. A room simulator.

after a delay of several tens of milliseconds. This delay is called the initial time delay gap. Several reflections will follow spaced fairly wide apart. Sometime later the main body of the reverberation will arrive. This is composed of a large number of reflections, spaced closely in time, which decrease in amplitude at an exponential rate.

The spacing of the early reflections depends greatly on the characteristics of the space being simulated. Because these reflections are introduced by the first bounce off the surfaces of the room, the initial time delay gap will linearly scale with the size of the room. The spacing between the early reflections will also scale with the room dimensions, but their exact distribution will depend on the shape of the room and hence will dictate the sound character of the room.

If the main body of reverberation is fairly smooth and uniform, only its decay rate (the reverberation time R_T) will have much effect on the perceived sound. In most rooms, this decay rate will be a function of frequency.

To simulate the response of real rooms, it is necessary to imitate many of these characteristics. One approach for doing this is shown in Figure 4. The early reflections are simulated by adding suitably delayed versions of the input signal. These are then mixed with the output of a reverberator, which produces the main body of the reverberation. The signal being recirculated or fed back around the reverberator is passed through a filter. This allows the frequency response of this path to be adjusted, resulting in a variable reverberation time with frequency.

The delay taps for the early reflections, the reverberation time, and the frequency response of the reverb may be set for the characteristics of the hall to be simulated. This is usually done by front panel selection from a number of preset choices or by entering numbers representing hall size, high-frequency reverb time, low-frequency reverb time, and so on. The selection of the non-adjustable parameters, and even the implementation of the adjustable ones, is not a trivial design task. Exact details of the reverberation algorithms in commercial units are closely guarded secrets of the respective manufacturers.

Digital filtering

Digital equivalents to conventional analog filters have been within the capability of available hardware for several years. It has achieved widespread use in compact disc players as a replacement for high-complexity analog anti-alias filters. It has also been applied very successfully to equalization requirements inside digital reverberation units and fully digital consoles.

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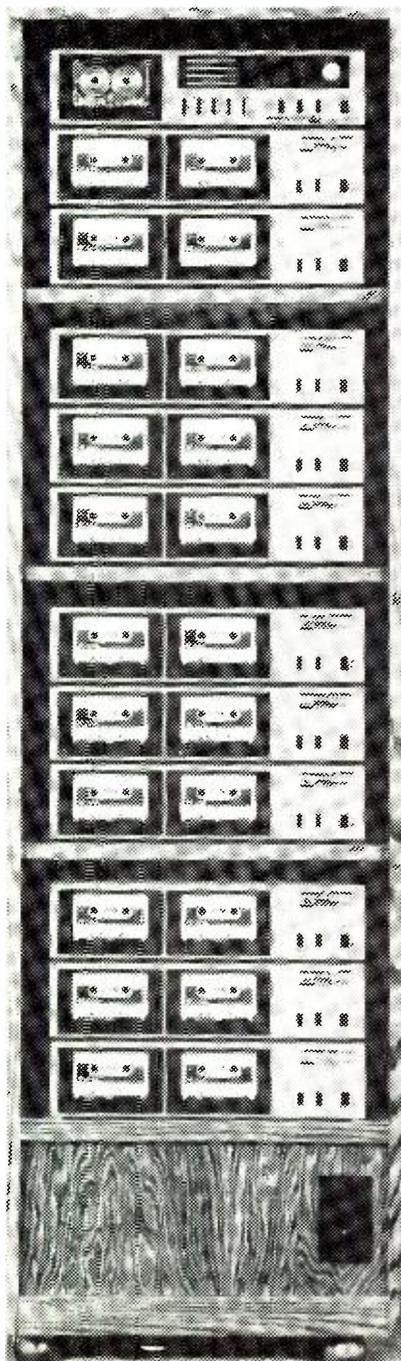
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These are all applications where the signal is already digital for other reasons and the filtering is easier to perform in the digital domain. Indeed, the costs of converting an audio signal to the digital domain purely to provide equalizing functions are just now becoming affordable.

The block diagram for a simple digital filter is shown in Figure 5. The block diagram's nature implies a hardware design. But, as with the reverberation topologies described earlier, may be implemented on a general digital signal proc-

essor. Several alternative topologies exist for digital filters. They differ in the trade-off between multipliers and adds as well as some performance attributes we will discuss later.

Analog and digital filters

The fundamental characteristic of a filter is its amplitude vs. frequency response. This is easy to define for analog filters because there is no upper limit to the available frequency range. However, in digital systems, aliasing occurs above

one-half the sampling frequency. This aliasing will also occur on filter responses, folding the portion of the response above the sampling frequency back down. The most straightforward way to make a digital filter is to take an analog design and convert it. This usually results in a digital filter shape different from the original analog design. In fact, only low-pass filters whose cutoff frequency is far below the sampling rate can be directly converted without modification. This direct type of conversion from an analog filter to a digital filter is called an *impulse invariant design* because the impulse response remains unchanged. Because the two filters have the same impulse response, they will also have the same frequency response.

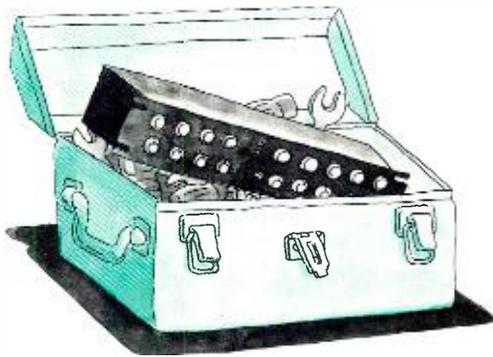
To realize other filter designs, transformations have been developed that convert an analog filter alignment to a digital filter alignment. These transformations map the entire frequency range of the analog domain into the digital range from dc to one-half the sampling rate. Therefore, there will be no aliasing problems in the response.

However, the *free lunch principle* comes into action. These transforms distort or *warp* the frequency axis to achieve this mapping. The resulting frequency will be different from the original analog one. This requires the analog response (the pole and zero locations) to be predistorted before the transformation. As you might expect, this involves trade-offs and the original analog response can never be achieved in digital designs for both amplitude and phase response over the entire frequency range.

Because digital filters contain delay elements, they can be designed to do things that are difficult or impossible in conventional analog filters. A good example of this is the phase response of a high order, low-pass filter. Typical analog filters have a square wave response that peaks on the front corner. This is due to a non-constant delay vs. frequency through the filter. A digital filter can be designed for constant delay vs. frequency to linearize the phase response and make the leading and trailing edges on the squarewave symmetrical. This can be done with analog filters by adding several all-pass phase shift stages to flatten the delay.

As mentioned above, there are several topologies that can be used to implement a digital filter. They differ in the order that delays, additions and multiplications are performed, and in the relative number of each operation. A greater number of delays requires more memory operations, and more multipliers introduces more round-off errors (and with some previous generation hardware took more time). Round-off errors will introduce distortion,

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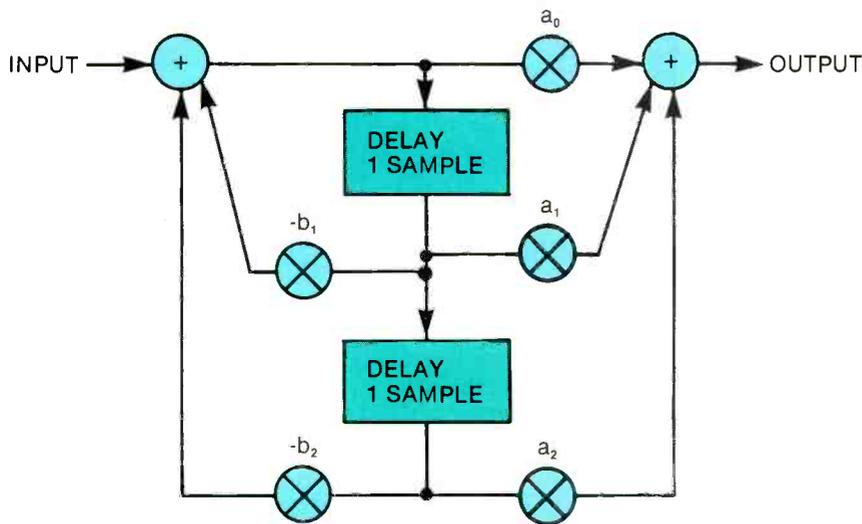


Figure 5. A basic 2-pole digital filter.

noise and very low amplitude oscillations that never die away.

Modern integrated circuits have eliminated the difficulty of implementing multiplications and the extra time they took. Architectures are now viewed in terms of the total number of operations they require and the effects of round-off errors.

Other digital signal processing

There are many units on the market that perform other digital signal processing tasks such as chorus, flanging and echo. However, it is likely that a commercial limiter using digital signal processing is just around the corner. A limiter that introduced a several hundred millisecond delay into the signal path would have this time available for the control circuits to sense the level. The limiter could then completely eliminate over shoot because it effectively knows the signal level several hundred milliseconds in advance. Such a unit has been built and used by the BBC and appears to work as expected. These units will appear on the market in the near future as the cost of A/D conversion decreases.

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W Winter NAMM Replay

By Paul D. Lehrman

Although no single product dominated the show, many companies exhibited improvements to existing products.

The winter meeting of the National Association of Music Merchants, at the Anaheim Convention Center in January, was a serious affair. There were few major innovations or surprises (except perhaps on the merger and acquisitions front, as Wurlitzer pianos became part of Baldwin, Sequential Circuits was absorbed by Yamaha and the two former arch-rival guitar manufacturers, Guild and Gibson, became one), but instead many improvements and strengthenings of existing products and technology were on display.

The convention floor seemed less crowded than usual, due in large measure to newly tightened admission rules that discouraged non-members from attending, and also due to an unseasonable monsoon—reported to be the worst storm Southern California has seen in 100 years—that flooded the streets of Anaheim on the final day of the show. Nonetheless, business was brisk, and many exhibitors commented favorably on the more serious overall attitude displayed by the attendees. By all reports, the show was an unqualified success—so much so that there was a fair bit of talk among attendees and exhibitors alike that they were thinking of skipping the upcoming summer NAMM show in Atlanta.

Mixing

Being a music-industry show, the big guns of mixing consoles do not exhibit at NAMM, but a number of small manufacturers showed mixing automation products that can bring major-studio capabilities to smaller facilities. Tucked away in a small booth at the Marriott Hotel, around the corner from the Convention Center, was ProMix, a MIDI-driven console automation system from a California company called Microsystems Inc. The system uses high-quality VCAs, insertable into any console. The control panel looks like a typical console automation system, with

100mm faders, switches for various editing modes and status lights.

When a fader on the panel is moved, it sends out MIDI commands in the form of notes, polyphonic pressure, pitchbend or continuous controllers (the user can decide which), which can then be recorded in any sequencer. When the sequencer plays back, the VCAs follow. Mixes can be edited by addressing the individual events in the sequencer, or in a special edit mode, in which the system automatically matches levels at punch-in points, regardless of the physical fader position, and will automatically cross-fade to previous levels at punch-out points. The minimum ProMix configuration is 16 channels, and it is expandable in blocks of eight channels.

J.L. Cooper Electronics showed a somewhat similar system, the MixMate, which a spokesman described as being a miniature version of the company's MidiMation system. The MixMate also communicates with MIDI controllers, but instead of storing them in an external sequencer, it puts them into the unit's own memory, which itself is a kind of sequencer.

The mix instructions can then be synchronized to tape or another sequencer using SMPTE, MIDI Clock with Song Position Pointer, or the "smart FSK" format used in the company's PPS-1 synchronizer. In addition, the unit can generate the smart FSK code, so it can be used to synchronize a sequencer to tape at the same time it is handling the mix. An option allows the unit to interface directly to a Macintosh or Atari ST computer for expanded memory, disk storage, and real time graphic readout.

The MegaMix system from Musically Intelligent Devices, which uses a dedicated PC—IBM, Atari or Mac—has been around for a while, but a new front end for it was being shown, called the IFI-8 Intelligent Fader Interface. The unit features eight faders with channel and group assign, and editing and solo switches on each channel. Up to eight units can be slaved together for a total of 64 channels.

Perhaps more importantly, the IFI-8 can also be used on its own, without the mixing system, as a dedicated real time con-

Paul D. Lehrman is *RE/P*'s electronic music consulting editor and is a Boston-based producer, electronic musician and free-lance writer.

troller for operating MIDI devices such as reverbs, delays, equalizers, etc. Shipping was due to start at the beginning of February.

Korg showed a smaller MIDI-controlled mixer, the C2. It has eight channels of VCAs made by Sound Workshop, and stores 64 programs, each with settings for level, equalization and two effects sends on each channel, along with crossfade and mute controls. Delivery is scheduled for May.

Yamaha discretely showed two small black boxes called "format converters." One of them, which is being manufactured for the company in America and was scheduled for delivery in March, will plug into the DMP-7's digital "cascade" jacks, and allow direct transfer in the digital domain from the DMP-7 to Sony PCM-1610/1630 ("SDIF/2") format. Not only will this allow output from the DMP-7 to be directly recorded on digital tape, it will also allow Yamaha's DEQ-7 digital equalizer (which uses the same data format as the DMP-7) to be incorporated into a Sony digital recording system without any analog conversion.

The other box, made by Yamaha and dubbed the FMC-7, will convert DMP-7 digital signals into DAT, AES/EBU or

SDIF/2 formats, although it will only operate in one direction at a time. It also incorporates a clock generator that can drive the DMP-7 at 48kHz (it is normally 44.1kHz) so the mixer can serve as the system master clock.

Synchronization

The era of the under-\$500 SMPTE-to-MIDI convertor has definitely arrived, with about half a dozen such devices on display.

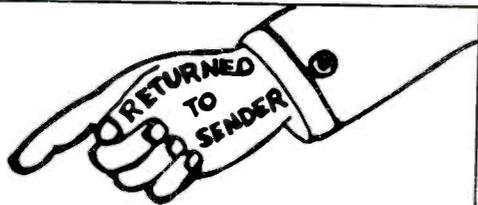
At the Imagine Computers booth, Michael Stewart, inventor of the Human Clock (which seems to have run into some distribution problems, and was not being shown anywhere) demonstrated his newest device, SMPTE City. The unit allows for 150 programmable cue points, each one containing SMPTE start and stop, tempo, offset, preroll and sequencer bar number information. It can display all information either in terms of SMPTE numbers, or in bars and beats and reads all SMPTE formats at levels ranging from -25dB to 0dB.

Passport Designs showed its MIDI Transport for the Apple Macintosh, which reads and writes SMPTE and converts it to MIDI Time Code. The company plans to include MIDI Time Code capability in the next version of its sequencing program Master Tracks Pro, which will enable the user to

continue to use the program's graphic tempo editing features, rather than have to deal with constructing a tempo map in hardware. It can also read and write "smart FSK," and is compatible with Cooper's PPS-1.

Southworth Music Systems, which was displaying at the Apple Computer booth (it was the first time Apple has had its own booth at a NAMM show), showed its new JamBox 2. Unlike its predecessor, the JamBox 4, which interfaces only with the Macintosh, this device is designed to work with any computer. It can generate an internal tempo map by recording tempo directly from a sequencer as it plays, and the map can then be stored, through MIDI system-exclusive, on a computer, or recorded (in digital form) on audiotape.

SMPTE synchronization is also a feature of Korg's new Q1 Workstation, which is a full-featured hardware sequencer with 512K of dedicated RAM, and a virtual disk memory of 1.4Mb. The unit reads and writes all SMPTE formats, and deals with MIDI Time Code and song pointer as well. It can store 16 tracks, each of which can drive all 16 MIDI channels, and each of which can have its own start time and loop status. The same functions, along with 16-bit stereo sampling, are available in the



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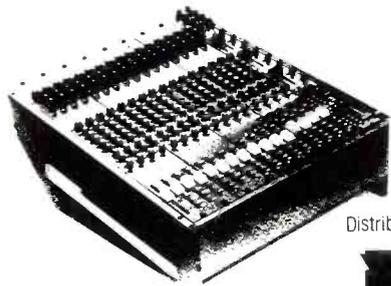
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company's S1 Production Workstation.

Opcode Systems showed version 2.0 of its Cue program, with a number of new features. While locked to SMPTE via MIDI Time Code, the program can freeze a video frame, record its SMPTE number from the vertical interval and store it as a cue. Cues can now contain up to 40 MIDI commands each, for accessing samplers, starting sequencers, etc. The program can also create tempo and meter maps and export them into other sequencers using the MIDI File format.

Vertical Interval time code was also the topic of discussion at the Fostex booth, where the company displayed a new VITC reader/generator and character inserter. The device can read code and generate streamers in two directions at once so that, for example, you can see elapsed time and time remaining simultaneously. It will also display user bits at the same time.

Processing

MIDI-controlled signal processing is moving ahead slowly, with no revolutions in evidence at the show, but a few interesting developments.

Lexicon, for example, has a new add-on board for its 480L Digital Effects System called the Sampling Memory Expander, which holds up to 10.9 seconds of 18-bit, 48kHz stereo samples (21.8 seconds mono). An interface is provided for direct transfer of samples to and from Sony SIFD/2 devices. The board allows pitch shifting and playback rate change, and provides full MIDI control over sample playback. Reverse playback will be available soon.

AKG's digital products division exhibited version 4.0 of its ADR 68K reverb and effects generator, which is now shipping. The new version, which consists of both a hardware and a software upgrade, allows up to 32 seconds of sampling (16 seconds in stereo), a comprehensive MIDI implementation including a special automation mode, MIDI Sample Dump Standard capability and 100 new factory presets.

ART announced a lower price for its ProVerb and a new addition to the line, the ProVerb II. The new unit has 200 presets, a footswitch for bypassing or stepping through programs and it responds to MIDI program changes. A more elaborate unit, the MultiVerb, is scheduled for delivery in April. It will do simultaneous multiple effects, and will feature increased bandwidth. The MIDI implementation also promises to be interesting, although it wasn't finalized at the show.

Roland showed two high-end products, the E-660 2-channel parametric equalizer and the R-880 4-channel reverb. Both devices feature digital inputs and outputs that conform to the new AES standard for signal transmission in the all-digital studio,

which means the units will be able to interface with digital tape recorders, samplers, consoles, etc., without ever having to convert the signal to analog and back again. MIDI control of the devices will be extensive, although final details were not available at the show.

Digidesign, makers of high-end editing software for samplers, showed a new 16-bit Sound Accelerator digital signal processing card for the Macintosh SE and Macintosh II computers. Although it's expected the card will eventually have myriad uses, a spokesman for the company explained that its first application will be to allow programs like Sound Designer and Soft-synth to run in real time—changes in a sound will not have to be communicated to the sampler over relatively slow serial data lines in order to be heard. An effective demonstration of the card consisted of an orchestral sample being equalized in real time with a click of the Macintosh mouse. The card will be available in "a couple of months."

Another sampling software company, Blank Software, had an interesting item: Alchemy, a stereo waveform editor for use with many samplers, although at first it will only work with Ensoniq and E-mu systems. Maintaining 16-bit integrity through-

out, it will do harmonic analysis of samples and resynthesis, and will convert samples among several formats, including Digidesign, Dyaxis, Blank's own formats, and even the new Macintosh II "SND" sound resource, and it can deal with several devices at a time in a DAN, or digital audio network.

Other toys

Several other new items at the show, although they don't fit neatly into the above categories, were worthy of notice.

Kurzweil, finally responding to a crying need from professional users of its K250 sampling keyboard, has come up with a Separate Outputs option for the instrument, so that 12 individual sounds can be mixed and processed outside the machine. The company also introduced its 1000 series of MIDI modules, which contain a variety of sampled sounds and offer extensive editing capabilities, multiple scale tables and the company's public-domain MIDIScope program built right in. However, they do no sampling themselves.

You probably thought that the old debate between 8-track and cassette was long dead, but Tascam has really nailed the coffin shut with the first 8-track cassette deck. The rack-mount model 238

runs standard cassettes at 3¾ips, and has built-in dbx noise-reduction.

Sonus Corporation, which make software and hardware for a number of computers, showed an intriguing fiber-optic based MIDI distribution system, which looks like it could be useful in many applications. Unfortunately, most of the attention in the booth was focused on a continuous live stage show, so there was nobody to talk to about it, and apparently no literature on it either.

Julian Systems, an Apple systems house in Northern California, quietly showed a Macintosh II-to-NTSC converter that could have some interesting implications as video software for the computer develops. For serious video applications, a genlock card will be available in a few months.

Finally, hi-fi and video fans are used to cute little remote controls, but so far musicians and studio engineers have been left out. That may soon change, thanks to AMR, a daughter company of Peavey, which showed the MIDI Director, a 1lb hand-held battery-powered device that will send MIDI program changes, Song Select, Start, Stop and Continue. It uses a 15-button keypad and a 3-digit LED. Maybe next year they'll have a wireless version.

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A Question of Service

By Laurel Cash

What's most important to commercial producers when they book studio time? It's not the equipment; it's the engineer.

If you were to ask commercial producers why they had or had not booked time at a particular recording facility, they would probably reply, "It's a question of service."

But what is service? Although the word has an agreed upon dictionary definition, the meaning is often intangible. But for most recording studios, particularly those that specialize in media and industrial recording, service has a tangible benefit: the money you receive from those clients.

So the question really becomes: What constitutes service to a commercial producer? To obtain a clearer view of what influences clients of commercial produc-

tion facilities to choose a particular studio, *RE/P* contacted 10 advertising agencies and creative services from across the country.

It is interesting to note that although the respondents were from different parts of the country, worked for companies of different sizes and produced a variety of projects, the responses were similar: if one took for granted that the equipment is of high quality, then it is the engineer that is paramount.

Creative input

Vince Manze, executive vice president of Steve Sohmer Inc., which does the advertising and promos for KNBC-TV (Los Angeles) and NBC's owned and operated stations, says that the engineer/editor is the one thing, by far, that he looks for.

"I want someone who will make suggestions, who is creative and proficient in his craft," he says. "It helps if the engineer has a good personality and can make the session pleasant."

Betsy Flynn, a producer with Hal Riney & Partners in San Francisco, agrees. "When we're working on big projects for Swanson, MJB Coffee or Alamo Rental Cars, a good engineer is a must. He must have a feel of what we are trying to accomplish, and have a good rapport with the producer."

Eric Rice Productions, Atlanta, produces and directs commercials, documentaries and corporate films. Rice prefers an engineer who doesn't treat you like a sub-human if you don't know technical lingo. But mostly, he says, he likes to work with an engineer "who knows what he's doing and can take the ball and run with it."

Richard Perlmutter, president of Radio Play, Los Angeles, which specializes in creative radio commercials for Hertz, careAmerica, Pic & Save and Murata

Pearl, says that the engineer is of primary importance.

"To me, the engineer is a co-producer. Engineers generally have more experience than most producers, only most producers won't admit it. The engineer is also more familiar with sound effects and music libraries. I looked for and found someone with the same sensibilities I have, and I rely on that person for his artistic judgment."

Craig Hazen, senior music producer at Young & Rubicam, New York, which is responsible for Lincoln Mercury, Kentucky Fried Chicken, *Time* magazine and AT&T, says it depends on the job.

"Sometimes, I want to use a state of the art digital music recording facility, and sometimes a jingle house." There are times, he says, when he wants to be able to experiment without having to pay \$350 to \$500 an hour for studio time. If he's doing a very large orchestra or choir, then he wants a very good studio, not just for the equipment, but for the room. All in all, he thinks that both the engineer and the room are the most important to him.

Scheduling flexibility

The next major consideration to most of these producers seems to be scheduling flexibility. Paul Fey of Paul Fey Creative Services, which produces commercials for the "Oprah Winfrey Show," "Jeopardy," "Wheel of Fortune," and Paramount Television, says that developing a relationship with the studio personnel is a necessity.

"I like it when they know who I am when I'm calling and can anticipate what I need," he says.

Manze adds, "It's important to be able to get a booking on short notice and that they can accommodate me on the back end of a session if I go overtime."

Pat Douglass, executive producer for

Laurel Cash is a free-lance writer in Los Angeles. She also holds a degree in engineering.

Identifying Client's Needs

By Fred Jones

The approach we've tried to maintain over the years is to spot a niche in the industry and fill it before anyone else. In fact, our business was conceived on the premise that producers need a place to get very high quality radio commercials and audio-for-video post-production work done, in a cost-effective manner.

In retrospect, it seems that most of the services we now offer were a result of either identifying a need in the industry or just plain listening to the feedback of our clientele.

Some of these realizations occurred in the initial planning of the facility. For example, in some cases,

Continued on page 71

Fred Jones is the owner of Fred Jones Recording Services, Los Angeles.

BBD0, Chicago, the people who bring you Wrigley Gum, Centel Corporation and Kemper Insurance, looks for reliability. "I work with studios I trust and that I know will stand behind their work," he says. "The facilities I use will work me in when I'm in a bind and that means a lot to me. The engineer is important, but to me the overall responsibility of the company is most important."

For some, the technical capability of the studio is most important. Karl Westman, vice president and music producer for Ogilvy & Mather, New York, prefers to work in a facility that is well-maintained. The chosen studio must have the ability to interlock 3/4-inch video to multitrack for film scoring and sport an engineer who is conversant with computerized mixing. He adds that it is important that the facility can accommodate extra people who are not there in a technical sense.

"It should be less intimidating and more user-friendly," he says.

Dave Hoogenakker, executive producer at Phillips-Ramsey, San Diego, says it's important that there be enough equipment available to get the job done.

"You shouldn't have to go out of the facility for transfers, the total job should be done at one place," he says. "Another thing

that a studio can do is give you different ideas for getting something done."

Walt Kraemer of Walt Kraemer Creative Services, San Francisco, who had created ads for Apple Computers, Amtrak/Caltrans, Dole and Mug Root Beer, has a unique way of working. Although he works from a loose script, much of his work consists of improvisational dialogue, at times using as many as five actors simultaneously.

Strong editing skills, therefore, are as necessary as good signal processing. Kraemer, who has been doing dialogue recording on 2-track analog, now uses digital audiotape, a capability that has been added to the facility at which he works.

Paul Fey also enjoys using DAT. "The studio I work at has a DAT machine in each room," he says. "It makes reviewing or comparing takes, a breeze. Now when I want to hear take 72, I don't have to wait while the tape shuttles."

Satellite technology

Satellite recording is another area where Kraemer and Fey are looking forward to using. They both do a fair amount of phone patch sessions, and are happy that satellite recording is now being offered.

"Phone patch is very important," Fey

says. "I use it in better than 50% of my sessions. But the fact that I can, for not all that much more, sit in an L.A. studio and record my talent in New York is something I like having the option of offering my clients."

"Phone patch or satellite sessions," Kraemer says, "allow me to use talent that would not normally be available to me because they might be in another city when I need to use them."

There are other items on the wish lists, but are not a big influence on why these producers choose to book a facility. Vince Manze asks for fast, quality dupes made with special attention to labeling. Paul Fey thinks that a talent payment service within the studio would be handy. Food and drinks are nice, but are not the sole reason why they book time.

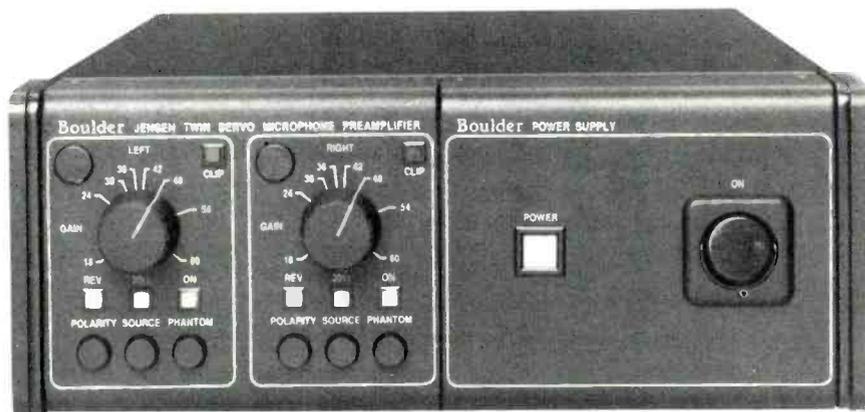
But having such amenities can be used as an indication of how well a studio will perform.

"The amenities are nice if they are there," Manze says. "It says that the studio's personnel want to take care of you and that the facility is a class act and of a professional level."

"Service is everyone in the facility being very, very accommodating."

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Finding a niche

By Michael Laskow

During the music boom of the early to mid 1970s, 24-track studios proliferated at an unprecedented rate. Everyone in the major markets (and some not so major) wanted to be the proud owner of a large, sophisticated, and most importantly, glamorous rock and roll facility.

Not Howard Schwartz. Armed only with his Mastercard and a healthy dose of chutzpah, he did the unthinkable. He opened a tiny facility on the 19th floor of the Graybar building in the heart of New York's ad agencies.

Everyone asked why, but Howard saw a niche that didn't require 24 tracks, a lot of startup capital or an enormous amount of square footage. He decided to go after a market that was largely untapped, if not a little unglorious compared with rock and roll: advertising.

Some of his contemporaries quietly smirked. Some laughed out loud. Howard plugged away 18 hours a day, editing mono voice-overs. Fourteen years and seven rooms later, nobody's laughing.

Many of the glamorous rock and roll studios have gone belly up. Many of those remaining are experiencing lean years, and diversification has become one of the industry buzz words. And to what areas are experts recommending studios diversify? Advertising and audio post production.

It's not as easy as it might seem. For starters, don't assume that by buying a few synchronizers, or merely announcing to the world that you now do production and post-production work, that you are in business. You may use the same consoles and the same digital multitracks, but for the most part, the similarities end there.

Different world

Ours is a world in which client services may be more important than automated consoles, and those services are quite different from what I encountered as a rock and roll studio manager. A typical call from a music client might have been a request for a certain brand of signal processor to be available (make sure that it can sample 2.5s) on tomorrow's session.

A typical call from one of our clients begins with, "I hope you can help me...we just finished shooting

our video and we need to resync the music to it. We've tried it at two other places and failed.

"You were recommended by company XYZ. The problem is, we shot to a wild music track without referencing the camera's to the Nagra's crystal, and now we can't get it to lock up. What will this cost to fix?"

As of this writing, this little beauty has just walked through the front door: "I shot four vignettes (10 to 30 seconds each) in England. I want to replace my British-speaking actor's dialogue with American voices."

Sounds easy? Not really. The client shot his video in PAL, the European video standard. We can't play their 1-inch on our NTSC video machine. It becomes our task to get the video converted to NTSC so that we can see what we're supposed to synchronize to. Only then can we begin replacing the dialogue.

Another service we provide is the satellite transmission of voice-overs and music. It has become routine for us to send to and receive from other cities in the country. Most recently, the request came for a satellite hookup with Tel Aviv, Israel.

Richard Crenna was there shooting "Rambo III" and his voice was needed here for a dozen or more spots for the Chrysler Jeep/Eagle division. This required finding a studio in Tel Aviv with a 15K phone line going to an uplink facility.

We also had to arrange the use of two satellites to bounce the signal halfway around the world. Dealing with the language barrier, the Sabbath and a 6-hour time difference also presented some problems. All in a day's work.

A typical day

Keep in mind that while this situation was being dealt with, we had all of our other rooms buzzing away with their own little problems. A typical day at Howard Schwartz can be comprised to 10 to 20 sessions ranging in length from a half-hour to 12 hours. Each of these sessions requires various elements, such as ¼-inch voice-over tapes, 1-inch video masters, music tapes, sound effects tapes, 35mm mags of who knows what.

Everything eventually ends up in the appropriate control room. With any luck, this happens before the client arrives. We have such an immense flow of tapes that it is necessary for us to have a full-time staff of five messengers to get them from place to place. These tapes also

have to be sent back to the proper agency, production house or network after the session.

And everything in New York has to happen very fast. The finished mixes for television are usually laid back to 1-inch videotape and sent back to the producer. Radio mixes, however, are a different story. From the session master we frequently make dubs. These dub orders range from about five to 5,000, which necessitates another room full of people to take the orders, make the dubs, cull the list of radio station addresses, fill out the waybills and ship the tapes.

If we were to make a mistake, we would end up paying for the missed air date. We don't make mistakes. As in rock and roll studios, our clients expect a very high level of quality in the control room. But we must also extend that "no mistakes" level of performance to all of the departments in this company.

Technical saviors

One department deserves a special mention. You might have wondered how we are able to maintain all of these rooms when they're always busy. You start with a very capable technical department that knows how to do quick fixes under fire during the day. Then they stay during the wee hours of the morning to fix it the right way.

As in rock and roll studios, these guys are often the unsung heroes. If we have downtime in the morning, it can throw off all of the sessions that follow. Our clients and their deadlines are far more inflexible than what I experienced as a rock and roll studio manager.

It's not like the good ol' days, when a group was in doing vocal overdubs and something went down for the count. I could simply ask them if they minded moving things back a day, and it usually wasn't a problem.

If something breaks at Howard Schwartz, we **must** have it fixed or replaced in a matter of minutes. Our technical department is worth every penny we pay them, but why they can't find the time to fix my Walkman is beyond me.

Maybe the reason that Howie's contemporaries laughed at him 14 years ago was not because they thought his studio was tiny, but because they thought that he was crazy for ever getting involved in this end of the business in the first place.

Michael Laskow is the studio manager and an engineer at Howard Schwartz Recording, New York.

Continued from page 68
producers would take completed commercials to other facilities for duplication and distribution to broadcast stations.

It was obvious that it would be easier on our clients if they could just leave the duplicating, labeling and mailing to us. So we built a real time cassette and reel-to-reel duplicating room with computerized labeling.

We have found it to be a great benefit to identify a need or trend before other facilities do. When this happens, we make it available to our existing clients and aggressively market the new services to potential clients.

For example, a lot of producers really love to create their own explosions and door slams, so we built a room with digital samplers and other keyboards for custom sound effects production. We also had more than 20 separate music and SFX libraries, so we developed a computerized listing and retrieval system to quickly locate and play sound effects or music cues.

Installing a fax machine when it

first came out is an example of a non-audio service we were able to provide, so our clients could send new or revised scripts. In the past, the scripts would arrive via express mail or (horror of horrors!) they were dictated over the phone and someone typed the script. The fax machine has been a lifesaver to both our clients and the staff.

Another service we provide is computerized scheduling and billing. With this, we can view and print out our bookings for the next month, week or just the next day. This allows us to better schedule our engineers and studios.

Then, through various reports, we can tell which client has been booking the most time, or what type of sessions each room is being used for such as radio commercial, TV commercial, film trailer, corporate video or sweetening for a TV show.

These are advantages for the studio owner and the client, as it provides more management control for the studio and accurate and fast billings for the clients.

Then there is DAT. This is obviously a piece of hardware, but it can

also translate to a service. Not only do the clients get the high quality of digital recording, but also almost instant access to different takes—which saves time. The producers love it.

The ability to do phone patch sessions is something we have offered from the beginning because there are a lot of out-of-town clients who like to take advantage of the Los Angeles talent pool. The next natural step was to get a satellite uplink.

Unlike a regular phone patch session, where a client can only hear the talent reader over conventional phone lines, with a satellite link the same client can now hear and record the talent with full fidelity at his end.

We don't consider ourselves to be just a recording studio. We're part of the production team. We can help acquire talent or step in and help at any point of the job from start to finish. All our engineers can act as producers, if necessary. They are there to give suggestions and do whatever is required to turn out a quality product—all of which adds up to effective client services. **REP**

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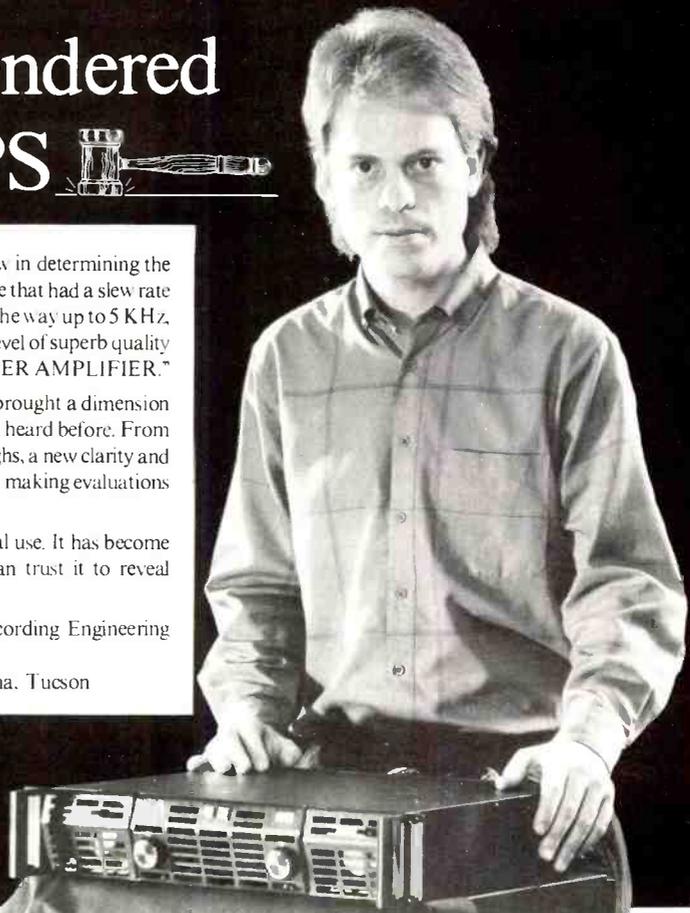
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The Basics of Music Libraries

By David Brooks

The sophistication of music libraries has grown in recent years, and titles are now available for any purpose.

Library music, or "stock music," or worse yet, "canned music," is an essential part of most modern post-production and recording facilities. Yet few in the business really know much about the source of this music.

Doug Wood is the principal composer, producer and owner of Omnimusic, New York, and he talks about some of the basics.

RE/P: *Library music, in many circles, seems to have a cheap connotation. What really is library music?*

DW: Library music is prerecorded music that's designed to help producers communicate more effectively with their audience. It's designed to elicit some kind of emotional or subconscious response. Just as the music from "Jaws" means danger to people, or the music from "Twilight Zone" means uncertainty. That's what we do with library music. We produce music that sets a mood that producers use to enhance their productions.

RE/P: *Why not use commercially available music?*

DW: Well, the copyright law prohibits the use of any music in films or commercials without the written consent of the copyright owner. In the case of commercial music, obtaining permission can not only

be difficult and time-consuming, but expensive as well. That's why producers turn to music libraries, where they know they can get original copyright-cleared music for a minimal fee, and with no hassles.

RE/P: *How has library music changed in the last few years?*

DW: Well, for one thing, the market has gotten competitive as the audio-visual and video production businesses have grown. People are more sophisticated now, and they want music that sounds as good as commercial recordings. That's why we went to compact disc as the medium for distributing our music. People want good sound, not just good music.

RE/P: *There have been a lot of new entries into the library music business in the past few years. Some have been very successful, while others have failed. What does it take to succeed?*

DW: I'd have to say that experience really counts. After leaving music school, I was working as a film sound editor for an industrial film company. We were doing films for clients like IBM and Grumman Aerospace, and I was in charge of selecting the music. When the guys in the white shirts and narrow ties came over to the editing room to approve the music, I learned quickly what worked and what didn't, where the boundaries were and how people who didn't know anything about music reacted to music.

That experience guides me today when I produce library music. Additionally, though, I would say that you've got to listen to your clients and get feedback to find out what works for them and what doesn't.

RE/P: *Do you still write most of the music yourself?*

DW: I write some, but we have 27 other composers who write for us, and their different styles and techniques make the library fresh and exciting. But because I produce everything here in our studio, we get a consistent sound.

Control room equipment

RE/P: *Let's talk about the technical aspect of recording library music. What equipment do you use in your control room?*

DW: A lot of people equate a big sound with a good one, so one of the most important items in our control room is our reverb. It's an EMT Gold Foil plate, which we bought from Kendun a few years ago. Although digital reverbs have become standard in most studios, our clients seem to prefer the old fashioned plate reverb sound.

I usually delay part of the reverb signal with a DDL, so the reverb becomes part of the rhythm. The EMT has a little less high-end emphasis than the digital machines, and sounds more like a concert hall, so when we have a multitrack running with 45 musicians, it sounds real.

RE/P: *What else is in your control room?*

DW: Our board is an Amek Angela 28x24, a real no-nonsense board that sounds great. And because I'm a real button-pusher and I like to work fast, both of our multitracks are MCIs, and our analog 2-track is an Otari MTR-10. We monitor using Tannoy Super Reds, powered by a Yamaha amp.

David Brooks is a musician, audio producer and writer in New York.

RE/P: Can you describe a typical production?

DW: Most of our composers are coming into the studio with lots of parts already down using either a computer or a sequencer. All of the keyboards, electronic and sampled drums, and synth parts go down together. After that, we often dub in real drums and add brass and strings. It depends on the piece, and what the composer has in mind.

Sampling/sequencers

RE/P: Has the revolution in sampled instruments changed the way you produce?

DW: Well, I really like what's happening with samplers, and I think it offers a lot of new tools to a composer. We use the Akai S-900 a lot, as well as the E-max and the Synclavier. The problem with samplers is that to really get the most out of them, you still have to understand how the great masters used orchestration to achieve emotional response.

You can sample a French horn, track it four times and have it play a nice melody, but unless you have a firm grasp of the technique of orchestration, it will never deliver the emotional response that French horns can deliver. I hear a lot of sampled sounds these days and they're interesting sounds, but they don't carry much impact.

RE/P: Why not?

DW: Every note on a real instrument has a different character to it. In addition, professional performers naturally phrase things in a musical way. To get even a great sampler like the Synclavier or the Fairlight to mimic the expression of a good flutist, you have to work all day, going in to the data and changing a little bit here and there in a random way to get a "real"

sound. A good flutist could play the piece in two minutes.

RE/P: So you continue to use a lot of real instruments?

DW: Absolutely. Not only do you get a superior sound, but also the music inflections and nuances that professional players can bring to a performance. Our musicians are really top-notch people, and they bring to the session more than just an instrument in a case. They understand what we're doing, and they're anxious to help us produce a good product.

It's not unusual for a brass player, for instance, to suggest a change in a part that he thinks would improve the piece. I don't know of a sampler that will do that.

RE/P: Let's talk about orchestration. Do you use different instrumentation for different styles of music?

DW: Yes. One nice thing about this business is that you get to produce a whole variety of music, from legitimate classical to the most modern pop/rock sounds. The style of the music dictates our choice of instrumentation, recording techniques, mixing techniques and resulting sound.

For instance, our sports music usually has a rhythm section, synthesizers, timpani, four trumpets, four trombones, four horns, 16 violins, four violas and two cellos. For the high-tech stuff, it may be just two synthesizers. One style of music that seems to be very popular with our clients is a hybrid of heavy synthesizers with orchestra. It lets a producer say, "This product is high tech, but it has a human dimension." Everybody seems to want that these days.

RE/P: You mentioned compact discs. How has digital technology changed the way you produce?

DW: Well, it hasn't, really. We still record on 2-inch 16- or 24-track machines, and mix to either the Sony 1610 or a modified F-1. The great advantage that compact discs have for us is the quality of sound that we can deliver to our clients. Also, access time is really fast, and when a client is looking for that "perfect" piece of music, the CD lets him find it in a hurry.

The other exciting part of CDs is the new interfaces that are being produced that allow computers to access CD changers. With these new units, you can select certain types of music, keep track of what pieces have already been used for which clients and keep a current database of what's in your library. We've been fooling around with a setup from Geffen Systems out of California, and we're really excited about the future uses of this technology.

RE/P: So this is where you see the future of library music going.

DW: No doubt about it. With the growing number of libraries and music available, the days are over when one guy could carry around in his head all of the information needed to find a piece of music at the drop of a hat. I would think that in a few years, most serious post-production houses and recording studios doing production work will have a computer-accessed CD changer as part of their basic equipment.

I mean, it all comes down to saving time, and saving the client money. Anything that will do that will be successful. And that's why we're continuing to keep an eye on these new innovations.

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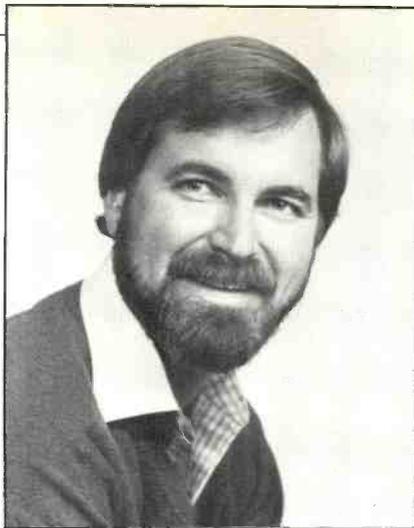
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Producer Profile:

Lars Clutterham



By Geoffrey T. Williams

The composer/producer of IDs for the Wave format created not jingles, but signature songs.

The newest trend in radio formats is the Wave. Born in Los Angeles in early 1987 on KTWV (formerly rock super-legend KMET), it has made its way into many major markets. No matter what people call it, from "the most innovative format in years," to "elevator music for yuppies," there's no denying that it is one of the fastest growing formats.

As any programmer will tell you, one of the fun parts of developing a new radio format is producing new jingles. And, as it turns out, one of the most memorable fixtures in the Wave is the musical identification. Most jingles for a new format, especially one breaking in a major market, would be produced by a major commercial jingle producer (just to relieve the program director from something more to worry about).

But, like many aspects of the Wave, producing the jingles was different. Instead of hiring a jingle producer in Los Angeles or any other city noted for its commercial production houses, the format's creators chose to travel south to San Diego, to a relatively unknown company, Lars Clutterham Music.

Producer/owner Lars Clutterham created a series of pieces that went beyond being jingles and became the signature sound of the new format. Most certainly, it was the first time that the music notes of the actual jingle were used in billboard advertising.

It started in late 1986, when Frank Cody, Paul Goldstein and Howard Bloom of Me-

tropolitan Broadcasting, creators of the Wave, formulated their plans for the new format. The jingles—what Goldstein calls "wavesongs"—are a product of the successful collaboration between the trio and Clutterham.

Goldstein booked time at Seacoast Recording, a 24-track studio in San Diego, and told studio owner Jack Elliot he wanted to bring in a few jazz musicians and just let them, you know, play for awhile. Just jam, until they had some stuff they could use for radio jingles.

That's a nice idea, but this can turn into chaos unless you have someone in charge who understands arranging and producing, as well as the timing intricacies of radio and television ID production. Elliot put the Wave people in touch with Clutterham.

Goldstein knew of Clutterham, at least by reputation, through his work on a jingle package for KNX-FM, in Los Angeles. Goldstein calls Clutterham "gifted," and Clutterham soft-spoken and articulate, accepts the compliment with good grace. But he says that in many ways what he did for the Wave was no big deal.

"Little songs"

"It's a radio jingle," he says. "You sing the call letters to advertise the station. That's not a new concept. Coming up with a logo melody to do that is not a new concept. There are no additional lyrics beyond station call letters or frequency. So they're just like shotgun IDs (those nanosecond blasts of music and singing developed by Bill Drake for KHJ in the early 1970s)."

Because this was a new format, featuring new and unfamiliar music, Clutterham asked Goldstein for samples of the titles the Wave would play. He knew some of the titles, and familiarized himself with others.

"I got back to him with two basic suggestions," he says. "Write all light groove tunes and all solo vocals. Little songs that sound like the music the station plays."

And, indeed, they seem to emerge from the records. KTWV believes they are so effective that it bought billboards in the Los Angeles area that are nothing more than the musical staff and their logo notes. (People in Southern California humming billboards?)

Clutterham says he thinks that Metropolitan came to San Diego because of cost and secrecy. Although they weren't sure, they thought the format was unique and they didn't want the idea to get out.

"Goldstein said, 'we want some jingles, but we don't want them to sound like jingles,'" Clutterham says. "They loved the demo I sent. They thought it was air quality. They said it might be six months before they got back to me for the finished product.

"It was less than two months. I did 30 cuts of music initially, from a minimum of 10 seconds to a maximum of almost a minute long." Commercial jingle recording is vastly different from a record date in one major way: time compression. Where an album can take several months to record and mix, the typical jingle package is over and done with in a matter of days—a week at the most.

The speed is because of several factors, the two biggest being the client's rising expectations—the closer you are to having something, the more you want it—and budget. Even though 24-track time in San Diego is under \$100 an hour, you still need to get things done quickly.

From his experience working for other jingle production companies, Clutterham was accustomed to a fast pace. But when he formed his own company he made a concerted effort not to let himself be rushed. He took two weeks to write the 30 starts (under other circumstances he might only have seven to 10 days). He says it was a comfortable time period. He could have taken more time, but it was enough.

The Wave session lined up like this: five days at Seacoast for live rhythm (bass, drums, guitar, piano), synth and other instrumental overdubs. Synthesizers included Yamaha DX-7 and TX modules, and Roland Super Jupiter, JX-8P, D-50 and S-50.

Vocals, at Weddington Studio in North Hollywood, took about three days. The final mix, completed back in San Diego, took about a week. It was a relaxed, even

Geoffrey T. Williams is president of Mainstreet Productions, a San Diego-based commercial production company.

laid-back, pace by most commercial recording standards.

"Paul Goldstein was the perfect client," Clutterham says. "He came to all the sessions and didn't say anything unless he had something to say worth listening to. We've done one other big session since then, 30 cuts last summer, and he only came to the vocals and part of the mix."

Non-conformist

In this day of the one-man-band, Clutterham is a bit of a non-conformist. A classically trained keyboard player, he admits to feeling more comfortable producing than playing during a session. He says he is still in the developing stages as a programmer, so he brings in a synthesist-programmer.

"We program right in the studio. I'd say, 'this cut needs to sound like this,' and we'd work it out right there. I feel a little like a painter who's asking somebody else to choose the colors in his palette. But I'm trying to build my business, so I have to wear as many hats as possible. So I'd rather have somebody else do the playing. That way I can give my time over to making sure all the pieces come together right.

"I still exercise a great amount of control during the sessions, but sometimes, a

suggestion from someone else can make all the difference. Other times I use my judgment, as any producer does, to moderate their suggestion."

All of the instrumental solos are real instruments.

"I don't think sample sounds make it," he says. "Even with the most expensive sampling synthesizers, by the time you come very close to the real sound, you've spent a lot more money than it would have taken to hire the real thing.

"I think it's an artistic decision, in the new age music, whether it's synthetic or real instruments. Which is the way I think it ought to be. The evolution of synthesizers is a natural evolution, and I'll eventually do all my serious sequencing on a Macintosh. I think the trend in studios is going to continue—bigger control rooms, smaller live areas—because the synthesizer trend is going to continue."

Other packages

After his success with KTWV, Clutterham sold new packages to WNUA-FM in Chicago and KNUA-FM in Seattle. Mike Donovan, general manager of the Chicago station, calls it "the most creative package for radio I've heard in 17 years." High praise indeed from the head of a station

whose last package was done by Jan Hammer.

"We don't call ourselves 'the Wave,'" Donovan says. "We have a different music mix. Funkier grooves, more texture. The package identifies our format so effectively." The trend toward longer, more musical jingles seems to be spreading. WXKS in Boston, a contemporary hit radio (CHR) station, is airing one of Clutterham's new packages. Other jingle companies, including the traditional powerhouses in Dallas, are writing and selling packages with similar approaches.

It's really nothing revolutionary, only evolutionary, a musical image tailored to fit its time. Chuck Blore did it with Color Radio KFVB in 1958, Bill Drake did it with Boss Radio a few years later, changing the way people listened to radio. And as the jingles for a new format, Clutterham's Wave package reflects the way that people listen now.

"The differences are that the music is longer," he says. "It's a softer sell and it's meant to hook the listener on the idea that he's about to hear a piece of music from the station. And before he realizes, he's not hearing a piece of programming, he's heard a jingle."

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April 1988 *Recording Engineer/Producer* 75

Facility Profile:

HLC

By Jeff Burger

A self-described "utilitarian" facility, HLC's commercial production success is borne out by its client list.

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Jeff Burger is *RE/P's* computer consulting editor, and is president of Creative Technologies in Los Angeles.



HLC is located in an art deco structure on Sunset Boulevard that was originally built in 1926 to serve as the Hollywood Chamber of Commerce offices and later as a movie theater.

of America's TV and radio advertisers? Yes, and it's also the playlist of HLC, one of Hollywood's hottest commercial production houses.

The H stands Ron Hicklin, one of the most-heard voices on America's airwaves for decades. Whether singing, vocal contracting or both, Hicklin has worked on more than 300 motion pictures and 200 records that have reached No. 1—everything from Gary Puckett and the Union Gap to Alvin and the Chipmunks to "The Music Man."

The L stands for partner Joe Lubinski. The two joined forces in the fall of 1982, as Hicklin was looking to map out the next phase of his career and Lubinski wanted to move to the major leagues. Their first collaboration saw their demo for the Wheaties "What The Big Boys Eat" commercial go all the way to air, winning a Cleo Award along the way for Best Music Commercial Of The Year. That led to Gatorade and the first 11 spots in the Levi's 501 Jeans campaign. The hits have just kept on coming.

While the C stands for company, family

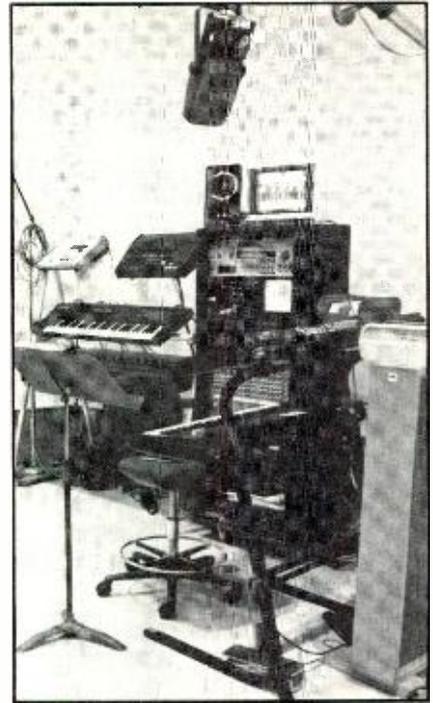
may be a more appropriate word. "Over the years I've worked with a lot of great talent and they're all equally important to our success," Hicklin says.

Frank Nadasty and Gary Joost share the engineering responsibilities with chief engineer Dick Hart. Hicklin's assistant producer, Debbie Hart, assists in the vocal casting duties, and a team of "permanent free-lance" session players, vocalists and writers who are used to working together are on constant call.

"We don't generally do a lot of one-man band stuff like many records are now; we go for more of a live band sound unless it's absolutely dictated," Hicklin says. "One reason is speed; it takes a long time to sit there and program drum machines and keyboards to do something. Commercials are often less than a 48-hour process from inception to finished product. There's just no time to sit and program keyboards and sequencers."

Facility layout

HLC's current home is an elegant art deco structure on Sunset Boulevard that was originally built in 1926 to serve as the



In each room, the console has exactly the same relationship to the monitors and center-front sliding glass doors opening to the performance areas.

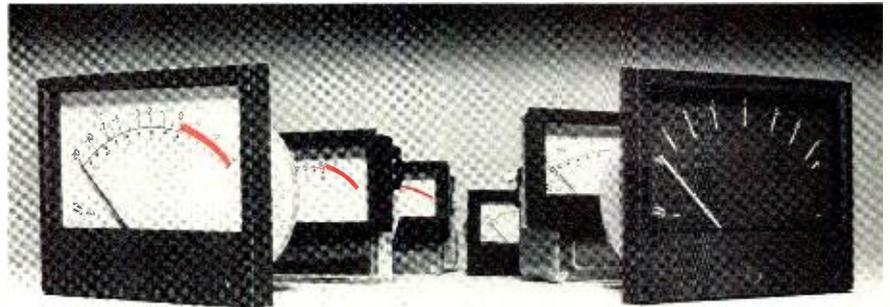
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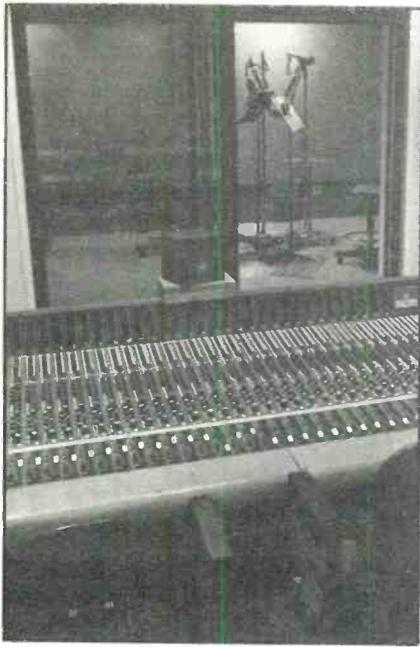
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The facility is subdivided in four separate studios, each dedicated to a different phase of the production cycle.

Hollywood Chamber of Commerce offices and later as a movie theater.

"We wanted a '10' facility," Hicklin says, "where everything had a bit of class even if it meant taking all the money we made and putting it right back into it. How many times have people recorded all night long only to find that their car has been ripped up and the radio stolen?"

"When it happens to you that's something you don't like but you can live with. When it happens to your client it's God-awful! I didn't want to have my staff and clients go down in a dark alley in the middle of the night or knock three times and say that Joe sent you, so we have a 50-car parking lot with security cameras and key locks. It helps people to feel safe and comfortable and that makes them more productive."

The 3,000 square feet of office space is augmented by a 12,000-square-foot facility subdivided to four separate studios, each dedicated to a different phase of the production cycle—tracking, overdubs, vocals and voice-over.

"It's sort of the Henry Ford process of making a jingle," Lubinski says. "The projects keep moving from studio to studio so

we can have as much work going through here as possible. The big difference between this and other facilities is that most studios are designed for outside rental. We don't rent to anybody.

"This facility is designed only for HLC and the volume of music we do. We could do about 40 spots a week but our normal output is about 20."

This is such an old building that the first task awaiting designers Jack Edwards and George Augspurger, construction manager Scott Putnam and general contractor Roy Hall was complying with California's earthquake laws. A \$100,000 power regulation system was also required to handle Hollywood's propensity for brown-outs. While basic design philosophies, such as non-parallel walls, were adhered to a great deal of attention was not given to specific acoustic measurements.

Nadasty says, "Isolation between rooms has not been a problem, but we're still in the process of doing some final room tuning with George Augspurger to iron out some small reflections."

The entire operation seems to have two bywords that are ingrained in the design—standardization and specialization. The designers intentionally created a common denominator in the form of matching control rooms. With minor variations, each control room is 19' 6" x 18' x 10'.

"In normal houses," Lubinski says, "there are a lot of changes between each studio because they're built at different times and people are always trying something new. Here an engineer might start off a jingle in Studio 1, move to Studio 2 for brass overdubs, move to Studio 3 for the singing and may even go down to Studio 4 for voice-overs. Our biggest concern was consistency; the product had to sound the same no matter where you were mixing."

Control room equipment

Each control room houses a Harrison 32 series console, MCI JH-24 machine, UREI 813 monitors tuned with White and UREI EQs, obligatory Auratones and a combination of Yamaha 2200 and UREI 6300 power amps. Studios 1, 2 and 3 all have a pair of Otari MTR-12 2-tracks and an MTR-12 4-track, while Studio 4 has an Ampex ATR-100.

In each room, the console has exactly the same relationship to the monitors and center-front sliding glass doors opening to the performance areas. (The glass doors don't pose an acoustic threat unless the doors are left open because of the lower monitoring levels used in commercial production.)

The only major differences between control rooms are where the machine alcoves are positioned and client amenities. For example, the main tracking control

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Each control room houses a Harrison 32 series console, MCI JH-24 machine, UREI 813 monitors tuned with White and UREI EQs, Auratones and a combination of Yamaha 2200 and UREI 6300 power amps.

room has a couch and the vocal control room has a desk for lyric sheets and the like, while the overdub room is less oriented to comfort because clients typically spend little time there.

Chief engineer Dick Hart designed a wiring system where all four control rooms could be interconnected using tie-lines that are terminated in "idiot-proof" 12-pin Elco connectors for tasks as mundane as 24-track transfers or as sophisticated as linking all four machines together.

"We're set up for easy 24-track transfers for when we're doing the same production piece but want to get into different lyrics and copy without losing the master," he says.

Each studio has two sets of tie lines—one to send and one to receive—and appropriate combinations are jumpered together as required in the central tech room. These connections come up as Deck B on the Harrisons' patch bays. An additional 16 pairs of tie-lines are available in every room for non-stationary equipment such as effects and video gear and a pair of MIDI lines is included as well.

The prohibitive cost of duplicating signal processing gear inspired the concept of

mobile racks that are patched externally with a single plug, while connections inside the racks can be rerouted on a panel of banana connectors. HLC's standard compliment of effects includes Lexicon PCM 70s, Yamaha REV-5s and SPX-90s, UREI limiters, Drawmer noise gates, Aphex units and a variety of Roland and Korg delay units.

Each control room has a Zenith System 2 video monitor above the forward sliding glass door to facilitate communication with each other as well as exterior security cameras.

"They were the only 25-inch monitors that fit in the space we had," Nadasty confesses, "but they're amazingly clear and have good color."

House-sync is planned for the future as HLC's video requirements become more sophisticated, and SMPTE lock is currently handled by Audio Kinetics pacers. Each type of wiring is routed through a separate conduit to preserve signal isolation; the wiring itself is C33 made by Belden, and was selected largely do to its cost effectiveness. Canare cabling is used for all mic cables.

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Each control room has a Zenith System 2 video monitor above the forward sliding glass door to facilitate communication with each other as well as exterior security cameras.

lar nature of the facility, and HLC has chosen to standardize Pin 2 as being hot on all XLR connectors.

Beyond the common ground of the control rooms lay the specialized configuration of each studio. Studio 1, the largest of the four at 42' x 22' x 9', is dedicated to basic tracking and includes an iso booth for the 7' x 4" Yamaha grand piano.

HLC's keyboard mainstay, Howard Pfeifer, chose that particular model "because Yamaha is known for its consistency. It just stood out above the rest—it was brighter, the bottom was deeper and it just had a much bigger sound."

Nadasty adds, "Elaborating on the drum baffeling, we originally had the drums in where the piano is but didn't like that sound. Then we tried them on an adjacent alcove and we weren't nuts about that sound either, so we moved them out into the main room to take advantage of the high ceiling and volume. We may sacrifice a little isolation but it's not a problem; most everything else goes direct and we just record the drums separately if we have

to. Things like brass overdubs are recorded separately anyway, so that's not a problem."

At 27' x 22' x 19', Studio 2 serves as a catch-all for overdubs ranging from brass to synthesizer. A movable drape in front of a hard rear wall serves to change the room's ambient characteristics to a more live sound. Studio 2's control room is also often used for final mixdowns.

Studio 3 is Ron Hicklin's domain—the vocal room. At 12' x 20' x 12', this room offers plenty of elbow room for what we'll soon see to be the highlight of the show. A pair of UREI 809s are mounted in the right wall in such a way that vocalists can do their take and audition it with relative ease.

"This room was designed acoustically for vocals," Lubinski says, "which gives us an advantage over having to sing in a big room, where the voice gets lost, or in a vocal booth which is usually a tiny little place. This room has just the right size and a warm feeling to it for singing—you can hear a whisper."

Studio 4, devoted to voice-overs or solo vocal, is the smallest at 10' x 16' x 12'. Larry Miller of the The Recording Place, who leases that space out, fills HLC's need for a voice-over specialist.

Everything at HLC seems to center on Hicklin's production style. "The attitude of the performer should be to give his best effort and leave his mark of excellence on a project; if you've got more to give, for God's sake give it! As a director, I try to cast the talent properly in the first place, so that all they have to do is go out there and be themselves. I then try to understand what the clients needs are and what our needs are so that if there's any communication problem between the booth and the floor, I can be the guide."

Ron Hicklin's perspective typically comes across as that of an artist. "The top of the line for a studio singer, where the most professionalism is demanded, is in the production of music for commercials... because they force you to utilize all your experiences. You have to be able to give it a record-like feel, articulate and worry

about delivering a convincing soundalike. So you end up with a wide variety of skills, which allow you to sell to many different demographic groups."

In capturing the best performances, Hicklin discards the more accepted techniques.

"Traditionally, people will say, 'let's do it again.' Well how about all the great things that were in that take? You end up with tunnel vision on this one thing that was wrong and you do it over, only to find out the rest of the new performance is not up to the standard of the first take(s). And if you punch-in, you can hear that work technically. I want performance to flow from beginning to end with the best feel, lyric, pitch and rhythmic pocket.

"On my Harrison board I set aside at least six tracks that I record the singers on and I try to get beginning-to-end performances on each one of them. Then I 'Frankenstein' a performance out of what they're doing—putting together all these wonderful bits and pieces, even down to syllables, and the only thing that's missing is the bolts because none of that should ever show."

The six vocal takes are then bused to, and monitored through, a single channel/track where the reverb is applied. Everything is standardized right down to the



Studio 1 is dedicated to basic tracking and includes an iso booth for the 7'x 4" Yamaha grand piano.

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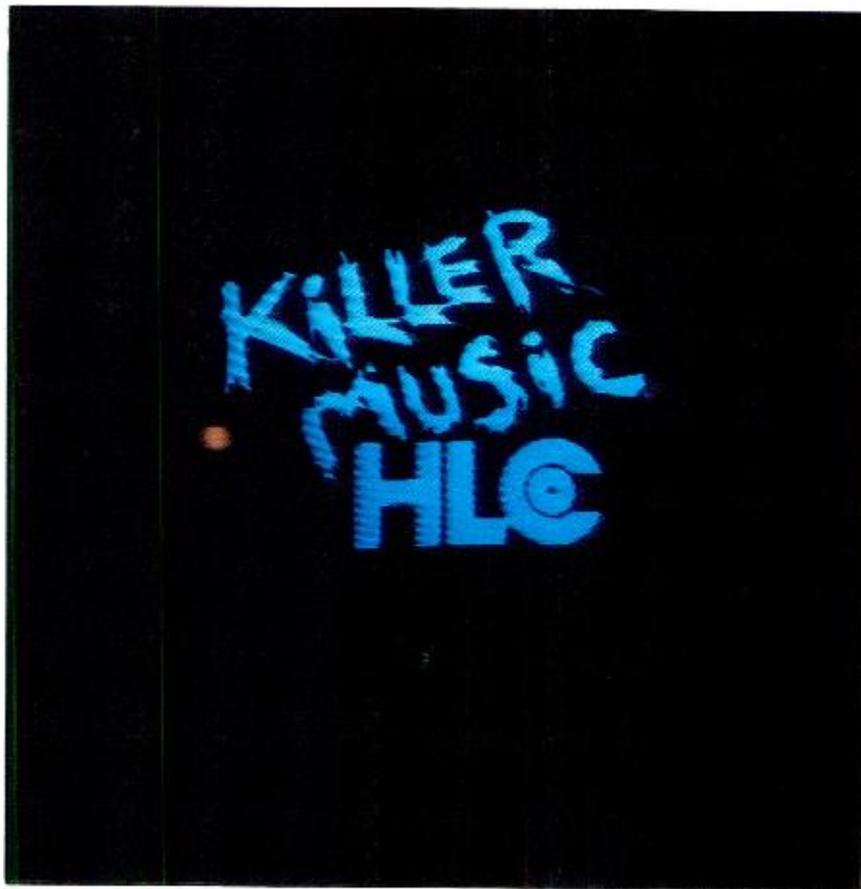
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track assignments, so that any engineer can pick up any tape and know what to expect, such as overdubs like synthesizers and piano on 11 and 12. With the exception of 14, which is always reserved for the lead vocal composite, 13 through 16 usually contain background vocals, 17 through 22 are always Hicklin's six vocal takes, 23 is click and 24 is sync.

Hicklin's tracking and mixing techniques influenced the choice of equipment significantly. He says, "If I can't do it with faders, the Harrison in Studio 3 lets me play the mute buttons just like a keyboard, and it's very quiet. The 24-track in that room has also been modified to be fast and noiseless so we can punch-in even in the middle of words."

Hart adds, "We've taken the auto-ramps out. They had ramp-up punch-in and ramp-out, which is supposed to make it smooth, but all that adds time delay. Ron can go from a vowel to a consonant!"

Automation and digital equipment doesn't play a role in this complicated

process at this point.

"We've got the capability to use the old Audio Kinetics/Allison Research system," Nadasty says, "but we'll go over a mix so many times that by the time we've got it where we want it, it gets to be useless; every time you do a new mix you bounce data between two tracks and each one adds a delay of 30ms or more. After 30 or 40 mixes, you have an audible delay and it also takes up two valuable tracks."

Hicklin adds, "The digital stuff we've used from time to time is marvelous, but it was a whole different process for us. The expense is probably what keeps us from going digital all the way in four matching rooms—that's a healthy nut."

Purchase decisions

The decisions regarding equipment purchases were largely influenced by previous experiences when renting other facilities. Specifically, HLC used United Western's Studio 2 extensively before moving into its current location, and much of the

equipment was bought out when that facility closed its doors as a studio. The HLC crew has also learned that all equipment is not created equal.

"Some of the differences in the Harrison has to do with how the design in the VCA chains has progressed," Hart says. "Some of them have the old Allison or dbx 101 series cans. Then they came out with the gold cans and the associated circuitry that goes with that, in conjunction with the automation on the rest of the console. That's what we've got in Studio 3."

Reinforcing the difference between seemingly like models, Hicklin adds, "We bought the 'Thriller' board right after that album and discovered that the magic didn't come with the console! (Laughs) Even though it was a Harrison just like our others, the mute buttons were not the same. I would try to 'play' the buttons the way I usually do... there was such a delay that I had to do it with the faders."

Nadasty says, "On some consoles, they have throw switches that are very hard to

work and others have these giant buttons that you have to physically push down about a quarter of an inch. It's a matter of speed."

The Harrison C-3232 and 4032 boards were also ideal for "3-stripe" mixing to 4-track where the music, vocals and voice-over (or sound effects or backing vocal) are each treated as separate elements for easy remixing, with the fourth track usually being 60-cycle sync.

The choice of MCI machines was also due to experience. "With the exception of our old 110, all the machines are Sony MCI JH-24's," Nadasty says. "We had been using a very old MCI 24-track that was just such a workhorse. We put it through 18 hours a day and the downtime on it was less than 1%. "Response was also a consideration. On vocals we have to go back and forth over a 1- or 2-second portion of tape over and over again, so the transport really gets a workout."

Monitoring

The choice of UREI 813s for the four control rooms was dictated by both performance and compatibility.

"At least in Los Angeles," Nadasty says, "there are as many 813s as there are everything else put together. Our philosophy here is not necessarily the absolute best of everything, but compatibility between our rooms and with other places so that agency people can be in a comfortable environment."

The monitor installations themselves are basically straightforward, and each has a baffled, air-conditioned cabinet beneath that houses the amps for both the control room and studio. Studios 1 and 2 have 811s in the main rooms, while Studios 3 and 4 sport 809s for their smaller vocal rooms.

The volume of business at HLC required the staff to computerize the business us-

"Over the years I've worked with a lot of great talent and they're all equally important to our success."

ing PC-compatibles and Q&A database software.

"For example," Lubinski says, "if a client calls in who is working on something for Disney, we can call up everything we've ever done for Disney and talk accurately about it. Or, we can look at everything in our reference library. A lot of times, people will call up on a jingle and want something that sounds like Whitney Houston.

We have a data base number for all our albums and CDs. We can locate items in our library by title, artist, or any number of other descriptions—pull it out and play it during a sales presentation." In addition, the computer helps locate specific HLC products in a library quickly growing beyond 2,500 cuts. Hicklin also finds the computer useful in cataloging the extensive vocal talent he's worked with and auditioned over the course of three decades.

"I've designed a form that goes all the way through peoples ranges, different sound-alike things they do, their reading skills, their language skills, their management and producers' comments like the fact that they do a great Aretha Franklin imitation.

"Then, if I'm sitting with a client who's looking for an Aretha Franklin type, I can type in 'Aretha' and up comes five names. I can just go to the drawer, pull out those tapes and play them for the client. If somebody's looking for a singing Elmer Fudd, we'll find it!"

Word processing software is also used to document the production summaries with regard to client, spots, personnel, time, musicians and equipment, as well as keeping a log of every connection in the building. Accounting is the next target area in computerizing HLC's specialized needs.

Utilitarian concept

Joe Lubinski summarized HLC's overall philosophy in their dedication to commercial production.

"These studios are very utilitarian," he says. "You see linoleum floors instead of hardwood. You don't see a lot of expensive wood trim. Usually a guy will come in and build one studio and put a million dollars into it, but we had to build four to make our concept!"

"In the past you would go to a new studio that would try to cover every aspect of the production and you were always stuck with a multi-purpose room. That's sort of like a multi-purpose stadium; it may be great for football but it's a lousy place to watch baseball. But if you go to Wrigley Field to watch the Cubs, where they only play baseball, it's great."

RE/P

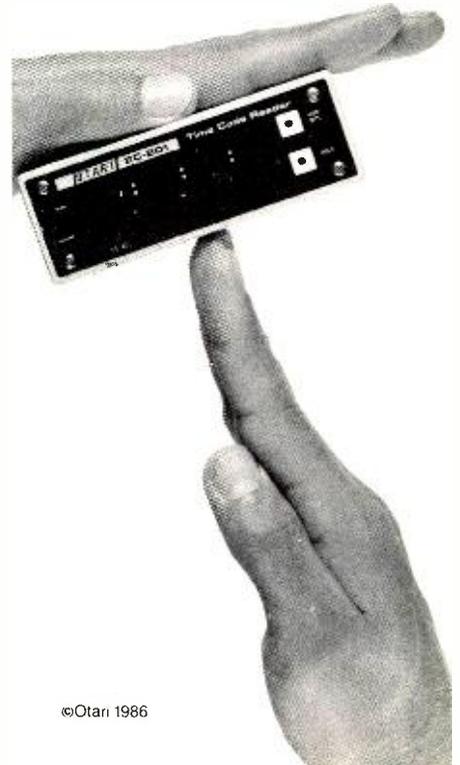
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Circle (48) on Rapid Facts Card

NBS Rejects Copycode

By Dan Torchia

**According to the National Bureau of Standards,
Copycode fails to achieve its stated purpose,
audibly degrades music and can be easily defeated.**

CBs's Copycode system for encoding R-DAT recorders is inadequate on three counts, according to a report issued by the National Bureau of Standards.

The 205-page NBS report cumulated five months of study in response to a congressional request.

The report answered three questions:

1. Does the copy prevention system

achieve its purpose to prevent R-DAT machines from recording?

2. Does the system diminish the quality of the prerecorded material into which the notch is inserted?

3. Can the system be bypassed, and if so, how easily?

The NBS concluded that the system does not achieve its stated purpose, that to some listeners there is a discernible difference between some notched and un-

notched copy and that the system can be bypassed easily.

Representatives on both sides of the controversy issued statements about the report. The Home Recording Rights Coalition (HRRC), which has lobbied against Copycode, said that the report effectively kills pending legislation before Congress. The pro-Copycode Recording Industry Association of America (RIAA) said it would seek another solution through "negotiation, legislation or litigation."

Jay Berman, president of the RIAA, said that the association would accept the verdict of the NBS study, adding that "any doubt about the sonic purity of our music requires that we go back to the drawing board." However, that conciliatory remark was tempered by a warning to R-DAT manufacturers.

"As I've stated in the past, it is our intent to sue any manufacturer that tries to bring R-DAT machines into the United States before this issue is resolved, and we have already established a legal fund for that purpose," Berman said.

Report details

The report is notable for its detailed technical descriptions and characteristics of the Copycode system—the first time such technical data had been presented. Previously, in congressional hearings, representatives from CBS demonstrated the system but had refused to release any technical data on the system.

According to the report, CBS provided to NBS two DAT recorders with decoding circuitry and two encoders. It also provided descriptions, specifications and circuit diagrams. NBS honored a CBS request that any proprietary information not be released. Consequently, all technical information presented in the report and cited in this article are from NBS measurements.

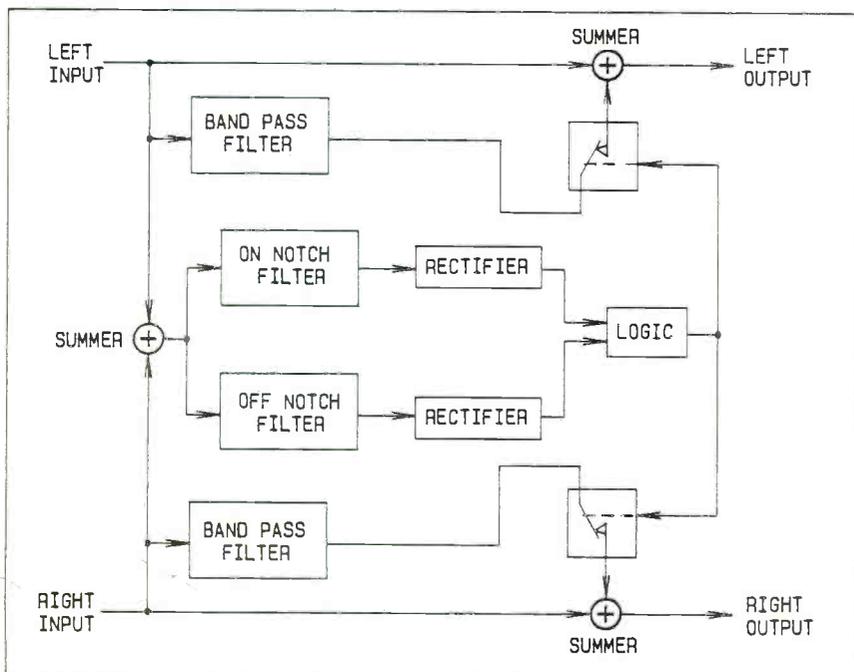


Figure 1. Conceptual block diagram of the major elements contained in the encoder, as determined by the NBS. Not shown are the unity gain amplifiers, through which the signal for each channel goes before it goes through a summing circuit (shown on the diagram as summer).

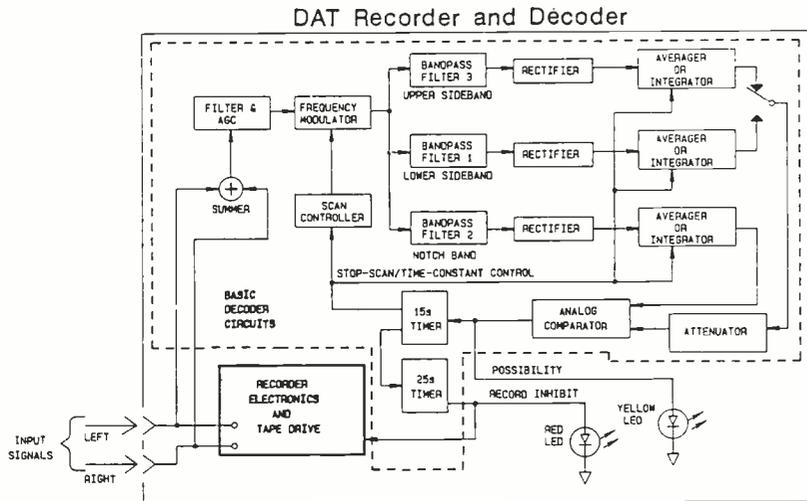


Figure 2. Block diagram of the major circuits contained in the decoder section (within the dashed outline) of the DAT recorder/decoder as supplied by CBS. These circuits were implemented on an ancillary circuit board separate from the other recorder electronics and tape drive.

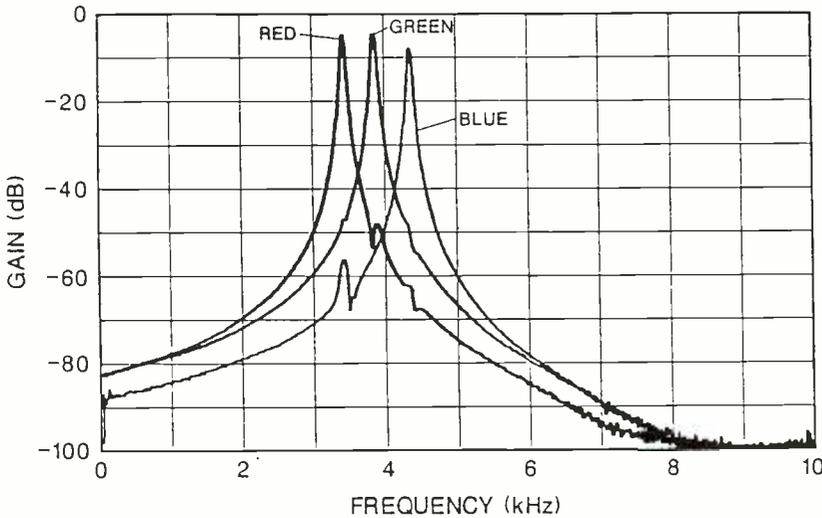


Figure 3. Amplitude response for the three band-pass filters in the decoder vs. the effective input signal frequency.

Encoder/decoder

The Copycode system, as proposed, consists of two parts, the encoder and decoder. The encoder consists of a narrow band-reject filter that removes signal components near a nominal center frequency of 3,840Hz. Logic circuitry determines whether to encode or not encode, depending on the signal levels near 3,840Hz and 2,715Hz.

Figure 1 shows a conceptual block diagram of the encoder as determined by NBS. The signal for each channel passes

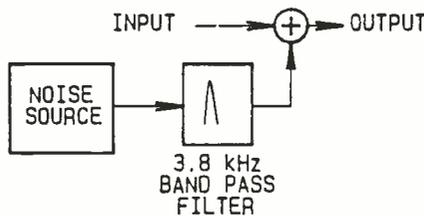


Figure 4. Defeat method No. 1 devised by NBS, which uses a band-pass-filtered noise source that is added to the encoded input signal. According to NBS, the method would be audible.

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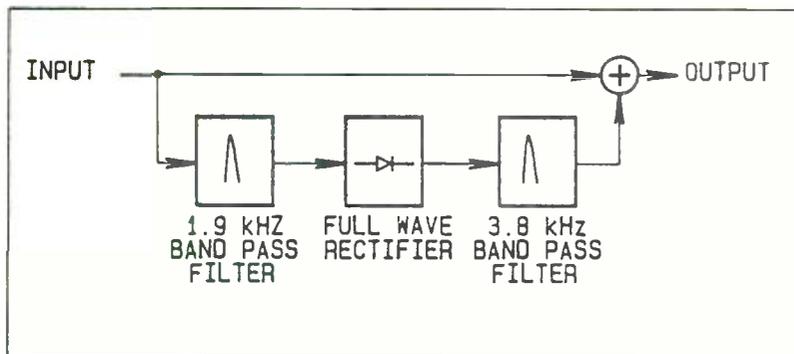


Figure 5. Defeat method No. 2, which uses a 1.9kHz component of the input signal doubled to match the 3.8kHz encoding notch frequency. According to NBS, this method would occasionally be audible.

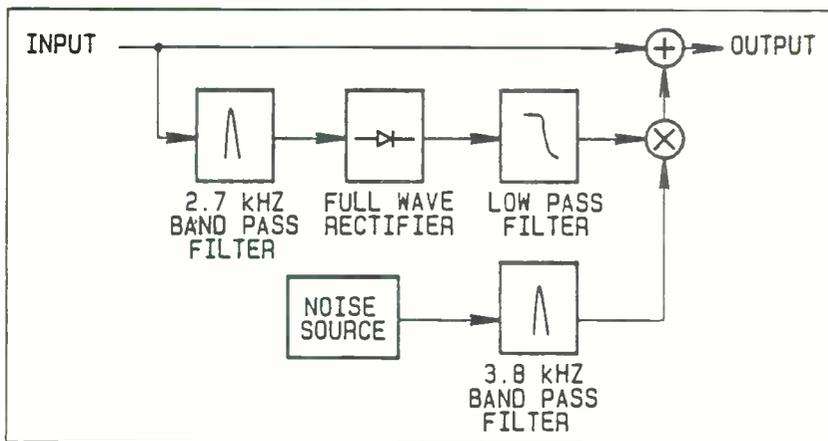


Figure 6. Defeat method No. 3, which uses the amplitude of the 2.7kHz component of the input signal to modulate the band-pass-filtered noise. Similar to method No. 1, this method would occasionally be audible, according to NBS.

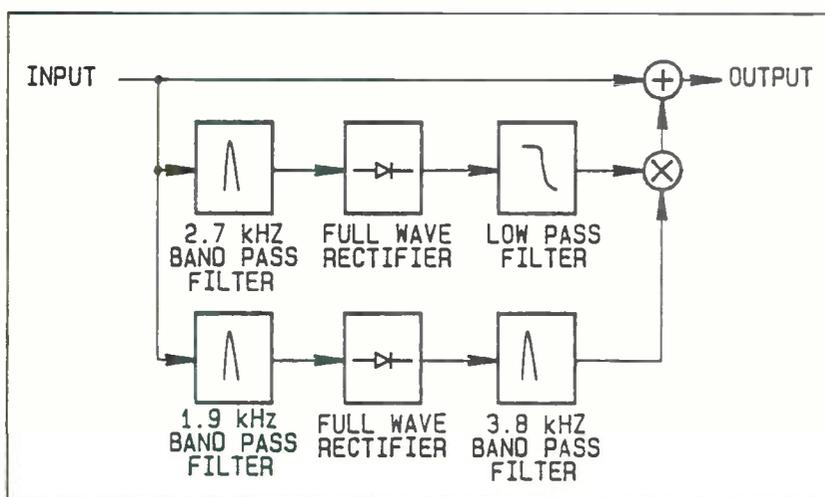


Figure 7. Defeat method No. 4, which uses the amplitude of the 2.7kHz component of the input signal to modulate the frequency-doubled 1.9kHz component of the input signal. Similar to method No. 2, this method would be inaudible, according to NBS.

through a unity gain amplifier to a summing circuit. In parallel, each signal passes through a narrow band-pass filter and then to a solid-state switch that can be enabled to direct the filtered signal for each channel to the corresponding summing circuit.

If the switch is closed, each summing circuit subtracts the band-pass-filtered signal from its corresponding direct signal and passes the resultant signal onto the output of the encoder. In this mode, the encoder acts as a dual-channel notch filter that removes the energy in the band covered by the band-pass filters, *notching* the material. When the switches are open, the signal passes through the encoder without being notched.

Selected equipment list

1. Recording playback material and listening tests

- Nagra IV-S recorder.
- Denon DCD-3300 compact disc player.

- Kurzweil 250 synthesizer.
- Sony PCM-3324 digital multitrack recorder with RM-3310 remote control.

- Sony disc mastering components: 2 Sony DMR-4000 digital master recorders; 2 Sony PCM-1630 digital audio processors, retrofitted with Apogee 944-G anti-aliasing low-pass filters; Sony DAE-1100A digital audio editor; 1 DABK-1630 PCB; 1 set of DABK-1631 digital I/O PCBs; D-3/4-75 cassettes.

2. Equipment calibration

- Copy of CBS CD-1 test disk for measuring CD player performance with EIA standard signals.

- Audio Precision System One-A audio test system with Zenith ZWX-248-62 computer.

- Fluke 8860 digital multimeter.
- Tektronix 7704 oscilloscope, with 7A18 dual trace amplifier and 7B70 time base.

3. Listening room

- STAX SR-Lambda Electrostatic Earspeakers.

- Energy 22 Pro Monitor loudspeakers.

- McIntosh MC7270 amplifier.

- RPG Diffusor Systems QRD diffusors used for acoustic treatment.

- Acoustic measurements: B&K calibrated microphone system (Type 4133 mic, 2645 pre-amp and 2607 measuring amplifier); Techtron TEF-10 audio spectrum analyzer.

The signal would then pass through the decoder circuit in the recorder (see Figure 2), which scans the input signal. If the notch is detected, a yellow LED mounted on the rear of the R-DAT recorder is activated, which indicates that a notch is possibly present.

The decoder monitors for the presence of the notch for approximately 13 to 15 seconds. If the notch in the signal persists, the decoder goes into the record inhibit status, lighting a red LED, based upon the output signal of the decoder.

Because the decoder must differentiate between a naturally occurring notch in the spectrum of the original signal and one generated by the the encoder, NBS noted several criteria exist in the process:

1. The notch must be sustained for at least 13 to 15 seconds.
2. A notch cannot be detected simply because the input signal level in the notch region is naturally too low.
3. The level of the audio signal in the notch frequency band is compared to the level of the signal in the sidebands, at 3,300Hz and 4,300Hz (Figure 3). For the decoder to detect and indicate the encoded signal, the notch band level must be lower than the level of the lower sideband.

False positives, false negatives

Critics of Copycode have cited that one of the problems with the system was that many musical instruments and musical passages have harmonics that fall into the notch. NBS specifically tested the decoder to see if this would happen, looking for what it called "false positives" and "false negatives."

The system worked correctly when encoded material tripped the record inhibit light and when unencoded material did not trip it. However, if unencoded material caused the record inhibit state, it was considered to be a *false positive*. And when encoded material did not cause the record inhibit, it was considered to be a *false negative*.

NBS cited two examples of naturally occurring notches in two pieces of music, the "Star Wars Proto" and Mendelssohn's "Wedding March."

NBS found that it was more difficult for the decoder to detect shorter notches than longer notches. In all, it tested 54 compact discs with a total of 502 tracks. After only a few passes, about 2% of the total number of tracks, or 9% of the CDs, had a false positive indication on at least one of the playings.

Additionally, during more extensive testing where NBS examined individual tracks or parts of tracks, it found a higher false positive rate. During this additional testing, false positives occurred on 16 different

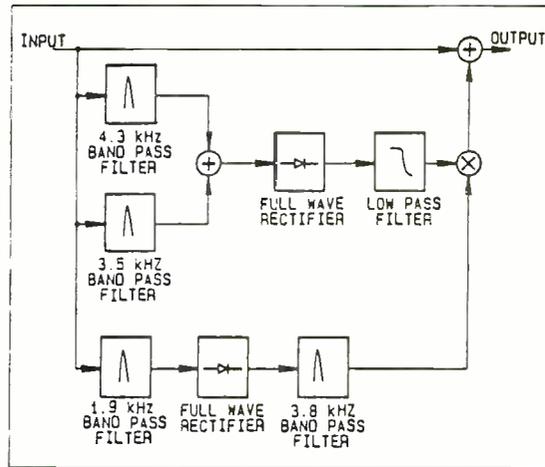
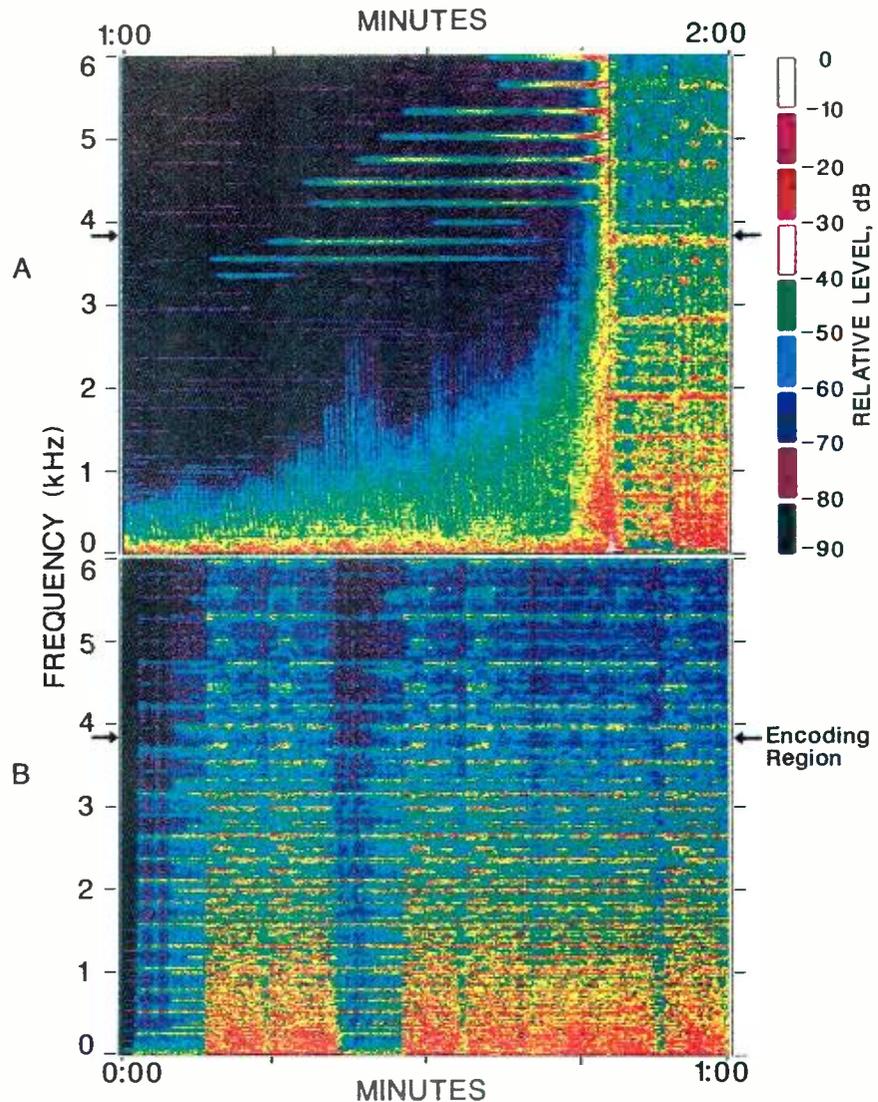


Figure 8. Defeat method No. 5, which uses the amplitude of the 3.5kHz and 4.3kHz components of the input signal to provide the modulation. Similar to method No. 4, this method would be inaudible, according to NBS.



Color spectrographs of music with naturally occurring notches that caused the decoder to go into the record inhibit mode even though the music was not encoded (a false positive). The top spectrograph shows one minute of music from the "Star Wars Proto," starting at 1:00; the second shows the first minute of Mendelssohn's "Wedding March."

tracks on 10 of the CDs. The final false positive rate was 3% for the total tracks, and 18% of the CDs.

NBS also tested for false negatives by recording whether the *possible* or *record inhibit* LEDs were on while it received encoded music. A track was labeled as a false negative if no record inhibit was observed during at least one of the runs. A disc was labeled as a disc false negative if no record inhibits were observed for the entire disc.

Listening tests

To determine if the system diminishes the quality of prerecorded material, two series of listening tests were carried out in which volunteers attempted to distinguish between encoded and unencoded music.

The first test was a serial listening test,

in which selections were presented as pairs that were repeated three times in close succession, with each member of the pair being either encoded or unencoded. Subjects were asked to decide whether the two members of the pair were the same or were different.

The second test, called parallel tests, consisted of 10 segments of music, each about one minute in length, presented one at a time over earphones. The selections were recorded on a 24-track digital machine, and consisted of three parallel pairs of tracks: a reference pair, and two encoded and unencoded pairs—with the encoded and unencoded pairs switching back and forth. Listening subjects could rewind the tape at will and switch among the three pairs of channels. The task for each was to identify the track pair containing

the encoded material.

Listening subjects for the most part were audio professionals in the Washington, DC area, according to NBS, who responded to a request through the local AES chapter. Music was selected to explore the audibility of specific effects upon specific passages. More subtle changes may occur, NBS stated, but would require more extensive listening in a familiar environment for evaluation. The results indicated that for some selections, the subjects detected audible differences between unencoded and encoded material.

Defeat methods

The third question involved whether the system could be defeated by electronic means. Because the units CBS supplied were prototypes, and individual components may not be representative of a final design, NBS considered only those methods that did not require physical modifications. A secondary consideration was the ease of implementation.

NBS devised five defeat methods, all of which could be constructed with off-the-shelf parts for about \$100, including power supplies (see Figures 4 through 8). The audibility of the methods ranged from noticeable to inaudible, correlating to the complexity of the design.

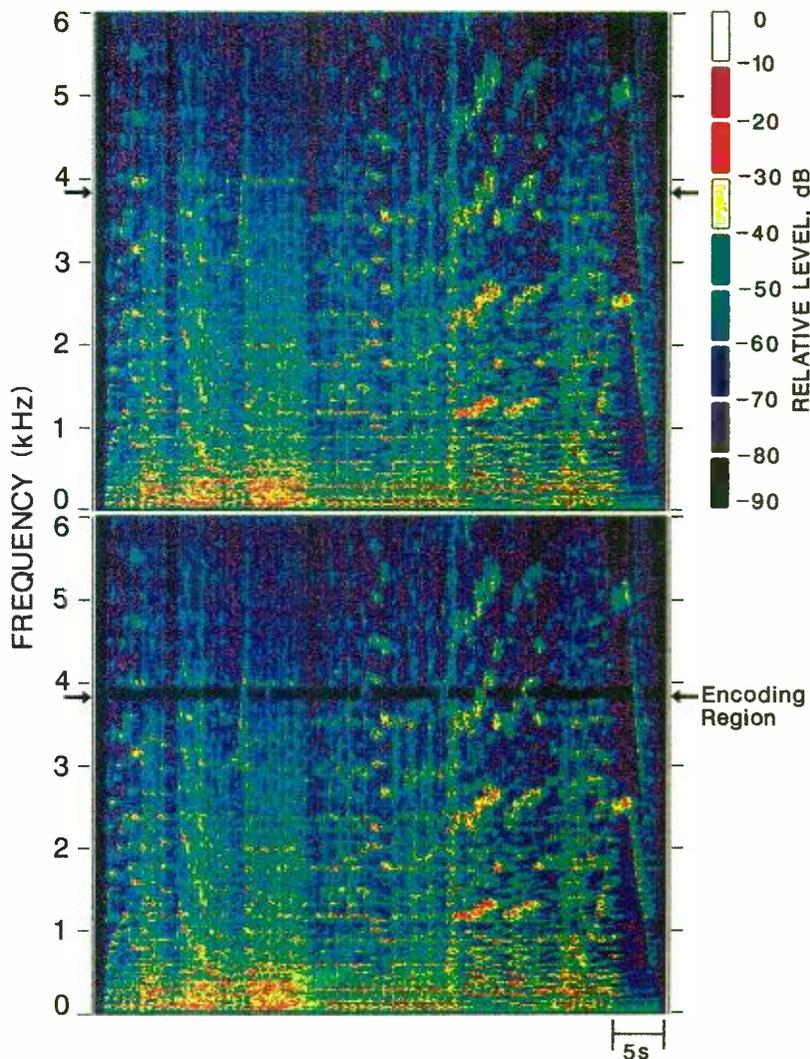
According to NBS, the degree of knowledge required to conceive, design and test a defeat circuit was that of a professionally trained electrical/electronics engineer. However, the knowledge level needed to construct one of the circuits was that of an electronics technician familiar with reading circuit schematics.

Test summaries

In its summary of results, NBS stated that the primary reason the system did not achieve its stated purpose was because of the decoder. For one decoder, the false negative rate was 7%, while it was 60% for the other.

Another consideration was the subjective listening tests. In the serial listening tests, 69 out of 84 listeners correctly identified encoded material more than 50% of the time. The odds of this happening by chance, NBS said, was three in 1 trillion, indicating that audible effects of the system are significant.

REP



Color spectrographs of unencoded (top) and encoded material. The passage is from Bizet's "Sarasate Carmen Fantasy Paganini Ensem"; the encoded passage shows the removal of a harmonic of a solo violin.

Editor's note: All figures in this article were taken from the National Bureau of Standards report, "Evaluation of a Copy Prevention Method for Digital Audio Tape Systems." Copies are available for \$24.95 from the National Technical Information Service, Springfield, VA 22161. Request document BP #88-169537. More information on the Home Recording Rights Coalition and Musicians for DAT, the pro-DAT organization for industry professionals, is available at Box 33576, 1145 19th St. NW, Washington, DC 20033; 800-282-8273.

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STUDIO UPDATE

Talkback

Remicing a Grand Piano For a Split Mix

By Rick Shaw

Anytime you try to provide a simultaneous television/recording and a PA mix on a live event, some compromises become inevitable. Typically, the PA mix tends to cause hollowness in the recording mix, and close micing is important for good rejection of the undesirable amplified sound.

Such was the case in an application where we were micing an 8-foot IBACH grand piano. The piano was amplified for performance at Mt. Paran Church of God, Atlanta, at the church's new 3,000-seat facility.

Complete television/recording capabilities at the new facility allow a simultaneous PA/recording mix, split between a 48-input DDA-D series console (loaded with theatrical-style inputs) for PA, and a Studer M-903 production console for the recording mix. Presently on the DDA, five foldback circuits provide monitoring for the platform, orchestra pit and chancel area. The main outputs drive a Turbo-sound speaker cluster and 32 Bose 102 ceiling speakers on three separate delay lines for a huge underbalcony area.

The director of music at Mt. Paran, Dwayne McLuhan, oversees a 40-piece orchestra and several choirs that combine for music specials.

We were having difficulty providing effective monitor levels on the piano for choir monitoring without creating feedback. Obviously, this also affected the integrity of the recording mix. We had been using two Neumann U-89s, mounted in suspensions inside the piano, with the lid at half-stick.

This provided a low- and high-end mic as well as offering some stereo separation for the recording/video mix. Unfortunately, keeping the piano lid open at half-stick

Rick Shaw is owner of Music and FX, Marietta, GA.

didn't give us the isolation we had hoped for, and it caused us to have to mic the piano too closely, which created hot spots across the keyboard. Using a piano pickup was discouraged because of the "electric" characteristic they impose on a good instrument.

We had also tried using PZMs taped to the underside of the lid, but the results lacked the presence we desired. The final solution proved to be a bit unusual, but did the trick.

We ended up using two TRAM TR-50 miniature mics. One was taped to the lid just behind the treble bridge and the other was taped over the bass strings. The lid could now remain closed, which kept our leakage problem to a minimum and also resulted in increased gain in our monitor speaker system.

The two TRAMS gave us the presence we needed for the recording and house mixes while at the same time offering a cleaner stage setup for the video cameras.

Have you encountered a problem or unusual request during a recent job that required a unique solution? We would like to share it with the industry. Send it to "Talkback"; if we use it, we'll pay you \$50. "Talkback" is a new forum for sharing your solutions to difficult production situations other engineers may encounter. In a continuing effort to educate, we feel this type of information is helpful and will display your professional abilities. This is not a tech tips column; rather, the focus is on solutions to problems—technical or non-technical.

Each month RE/P will select one or more "Talkback" pieces for publication.

To submit: In 1-2 pages, describe the job, what the problem was and what you did to solve the problem. Include any supporting documentation, such as diagrams or photos, that would help explain the situation. If we publish your entry, you and your company will be fully credited.

Send material or inquiries to Michael Fay, Editor, RE/P, 8885 Rio San Diego Drive, #107, San Diego, CA 92108.

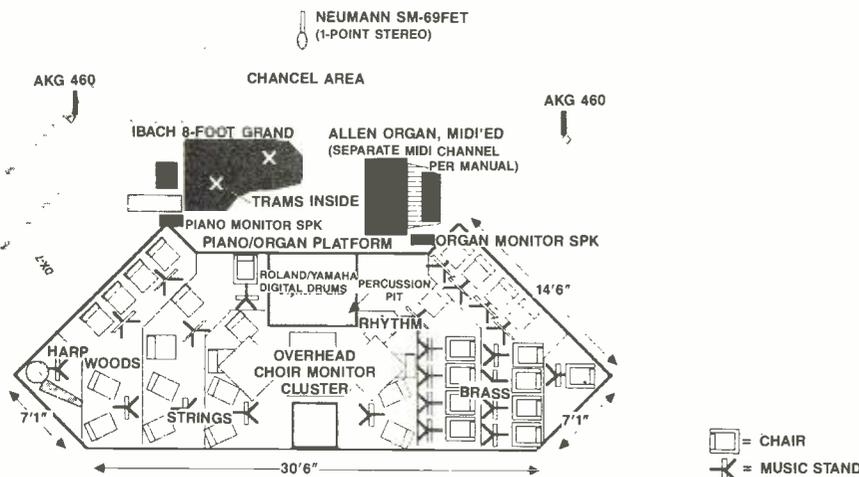


Figure 1. Orchestra pit and chancel area for Mount Paran North with instrument positions.

Studio News

Northeast

Effanel Music (New York) has installed Acoustic Science Corporation's Tube Traps in its 48-track mobile unit, said to be the first such use in a remote vehicle. 66 Crosby St., 4B, New York, NY 10012; 212-807-1100.

Southeast

Strawberry Skys Recording Studios (West Columbia, SC) has purchased two dbx 160Xs, a Panasonic television/monitor

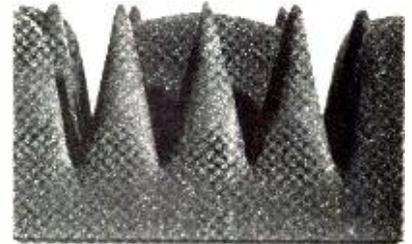
and an Atari 1040ST computer system. They have also installed oxygen-free cable. 1706 Platt Springs Road, West Columbia, SC 29169; 803-794-9300.

Southern California

Soundcastle Studios (Los Angeles) has remodeled and re-equipped its Studio 1 with a 72-channel Solid State Logic SL-4000 G series console with Total Recall automation. Other new equipment includes Studer A820 24-track machines with Dolby SR noise reduction and a Mitsubishi X-850 32-track digital recorder. Acoustic design and construction was by Vincent Van Haaff. 2840 Rowena Ave., Los Angeles, CA 90039; 213-655-5201.

RE/P

A pointed attack on noise.

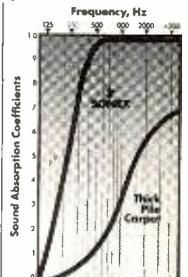


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SONEX is manufactured by Illbruck and distributed exclusively to the pro sound industry by Alpha Audio.

Circle (51) on Rapid Facts Card

NEW PRODUCTS

Soundcraft Digitor audio editing system

Digitor records up to six minutes of stereo into RAM memory, allowing all editing features. Audio recording is 16-bit, 44.1kHz and is analog in/out; digital interfaces will be introduced soon, according to the company.

Circle (155) on Rapid Facts Card

Yamaha NS40M close-field monitor

The studio monitor is a larger, upgraded version of the company's NS10M, and is designed for close-field monitoring. A 3-way system, the monitor incorporates dual "white-cone" woofers, a soft-dome midrange driver and a soft-dome tweeter. Frequency range is 30Hz to 20kHz, power capacity is 100W and SPL is 90dB at 1m.

Circle (157) on Rapid Facts Card



Soundcraft automation for TS12 console

The automation system uses a 68000 16-bit microprocessor and includes real time switching of three aux send on/off and channel cut and EQ in/out, in addition to fader and mute settings. Up to 10 mixes can be stored on a 3.5-inch disks; a high-resolution color monitor is optional.

Circle (158) on Rapid Facts Card

Kirk & Sons fiber-optic cleaver

Produced under license from British Telecom, the Optical Fiber Cleaver Tool allows optical fibers to be produced cleanly and squarely, which allows the maximum transfer of light across a joint when two ends are spliced together. The self-calibrating tool can be used to cut a variety of fiber widths.

Circle (150) on Rapid Facts Card

Akai SXM007 memory expander

For the company's X7000 and S700 digital samplers, the expander provides battery backup for 10 samples. This allows the easy access of frequently used samples, and also allows the units to continue operation if they are accidentally unplugged or if there is a power interruption.

Circle (151) on Rapid Facts Card

Perma Power Communications RTD410 surge suppressor

The unit, an outlet strip, is designed to protect facsimile equipment and computers with modems against surges on both the power line and the telephone line. It provides four power line outlets, one telephone line input and one telephone line output. The unit is also designed to shut down if the suppressor element wears out, protecting equipment from raw power.

Circle (152) on Rapid Facts Card

Upgrade, addition to Editron 500 audio production system

The system now features 10 softkeys, each holding up to 50 keystrokes, an auto-assembly of cue lists, enhanced cue list operation, varispeed sync and an optional 20-channel ADR card. Also new is the 100A system, a starter system configured to control one master and up to three slaves.

Circle (154) on Rapid Facts Card

Akai-Linn MPC60 MIDI production center

The first product from the Akai-Roger Linn collaboration, the MPC60 consists of a drum machine, a MIDI sequencer and a SMPTE-to-MIDI synchronizing system. It is designed to provide all necessary components in a production studio except for keyboard voices. The unit features a 320-character LCD display, and every data field has its own Help screens.

Circle (160) on Rapid Facts Card

WexTech Systems MPE software

Music Production Ensemble is a software package designed for various administrative tasks in the music production industry. The programs in the package allow users to write an AFM contract, an estimate with a cover letter, payroll checks directly from contracts, and calculate and print out 1099 and W2 forms. The software assumes no prior computer knowledge.

Circle (161) on Rapid Facts Card

Atlas/Soundolier Series V modular equipment enclosures

Manufactured to EIA standards, the enclosures are made of 14-gauge cold-rolled steel and have removable side panels for easy equipment installation and servicing. Available widths are 19, 24 and 30 inches; heights are available from 21 to 70 inches.

Circle (163) on Rapid Facts Card

Audio-Technica 40 series mics

The series features three models, two shotguns and a cardioid capacitor unit. The shotguns, models AT4073 and -4071, feature open circuit output voltages of 56mV/PA and 62mV/PA and have very fast rise times, according to the company. Model AT4031 has a maximum SPL of 140dB and a frequency response of 30Hz to 20kHz.

Circle (164) on Rapid Facts Card

Zildjian ZMC-1 cymbal micing system

A joint venture between Zildjian and Barcus-Berry, the system consists of electret mics and a power mixer, and is the first designed especially for cymbals, the company says. The mics attach to the cymbal stand above the tilter; the mixer contains an effects loop, allowing signal processing to be added to the cymbals.

Circle (159) on Rapid Facts Card



Additions to AMS AudioFile

New features to the digital recorder/editor include full bandwidth scrub editing, full cut and paste editing, internal digital level control, internal fades/cross-fades and internal panning, and up to seven hours of storage.

Circle (179) on Rapid Facts Card

Tascam 238 Syncaset 8-track cassette recorder

The first 8-track multitrack recorder using standard cassettes, the unit features 3¾ ips tape speed, remote control, auto punch in/out, auto rehearse, dbx II NR, MIDI FSK compatibility and SMPTE compatibility. A serial connector for external computer control is also included, and the unit has open architecture for future software development.

Circle (162) on Rapid Facts Card

Stewart Electronics HDA-4 distribution amp

The 4-channel unit is designed to be connected to the headphone output of a mixer. A master level control allows simultaneous control of all four outputs, and individual level controls allow independent control of each headphone. A stereo/mono switch allows the unit to send mono signals to both channels of a headset.

Circle (170) on Rapid Facts Card

LVW Electronics DPA 1624 programmable amplifier

The unit is a 24-channel amp that automatically adjusts the volume level of every loudspeaker in a distributed sound system, depending on the microphones in use. Different groups of loudspeakers may be programmed to have different output levels simultaneously and in any combination. The unit can be programmed, allowing users to define a specific loudspeaker dimming pattern for each microphone.

Circle (171) on Rapid Facts Card

Valentino library on CD

The company's production music and sound effects library is now on compact disc. Containing 22 volumes, each selection contains a 3-minute version, 15-, 30- and 60-second version, and a rhythm section only version. Each selection has its own track, eliminating sub-indexing.

Circle (172) on Rapid Facts Card

Paso 4000 series

The 4000 series is comprised of four integrated audio amps and three power amps, all with 25V-70V or 4Ω, 8Ω and 16Ω output. Power ranges from 40W to 200W. The line is also augmented by a 9-input mixer and a line of equipment racks.

Circle (173) on Rapid Facts Card

Shure W15HT wireless microphone transmitter

The transmitter is available in two versions. The W15HT/58 is equipped with an SM58; the W15HT/87 comes with the SM87. The transmitter operates at a single crystal-controlled frequency between 166MHz and 216MHz, with a total of 15 frequencies available. The SM58 and -87 heads may be interchanged with the transmitter.

Circle (175) on Rapid Facts Card



ART Master Blaster sound system

The powered speaker system consists of two 10-inch transducers with a 1-inch compression driver, coupled to a constant coverage horn. The system comes in two models; Supercompact eliminates high SPL distortion, and Impact I handles low-end frequencies. The dispersion angle is 90°.

Circle (174) on Rapid Facts Card

Sanken CMS-9 microphone

The CMS-9 is a MS-stereo field microphone that uses a mid-side sum and differencing producing circuit inside its body, and outputs normal stereo signals that can be inputted directly to a field recorder equipped with L-R and 48V phantom power. Dynamic range is 108dB and self noise of 19dB or less.

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Circle (52) on Rapid Facts Card

NEW PRODUCTS

Winsted rack slide kits

The kits are designed to attach to Sony BVU-950 VCRs for mounting in the company's EIA rack cabinet. Two models are available; model F8525 for slope cabinets, and model F8526 for vertical cabinets. The kits feature full extension ball bearing sides with positive lock at full extension for servicing and maintenance.

Circle (176) on Rapid Facts Card

FSR Inc. WCA-4 audio summer

The unit has up to four line-level bridging inputs and gives the user a 600Ω transformer balanced output. The WCA-4 is packaged in a metal enclosure that mounts on the rear rack rails and also features unity gain, no insertion loss and XL I/O connectors.

Circle (177) on Rapid Facts Card

Pulizzi Engineering PC 125 power controller/line conditioner

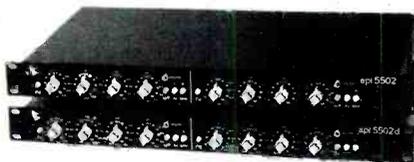
The PC 125 is available in 240V or 120V, and in 12A, 15A or 24A models. Each unit contains eight switched outlets and two unswitched outlets with three remote I/O ports that are activated with the local/remote switch. The models can be fitted in a 19-inch racks and are 3.5 inches high.

Circle (178) on Rapid Facts Card

API Audio Products 5502 equalizer

An all-discrete unit, the 5502 has all the features of the company's 550A, in addition to four frequency bands and 14 new frequency points, including 20Hz and 20kHz. The unit also has all-gold switches, four API 2520 op-amps and 36 discrete transistors, and many functions are selectable with special jumper plugs.

Circle (169) on Rapid Facts Card



Soundcraft 6000 console

Featuring a split bus architecture based on the company's 500 and 600 consoles, the 6000 accepts 68dB of continuously variable gain and a low noise floor. It is capable of up to 24 buses and can be expanded up to 32-track monitoring. Also featured is a 4-band semi-parametric EQ and a phase reverse switch to reverse polarity on individual input modules.

Circle (165) on Rapid Facts Card



Furman Sound HA-6 headphone/monitor amp

The unit is designed for overdubbing applications. In the headphone mode, it powers up to six pairs of headphones. In monitor amp mode, the unit powers a set of studio monitor speakers, allowing musicians to listen to the playback without headphones. Power is 20W per channel.

Circle (166) on Rapid Facts Card

DigiTech DSP 64 processor

The DSP 64 is a dual digital reverberation and effects signal processor that can be used as two independent reverb-effects units or as a single stereo unit. Each channel offers inputs and outputs on the rear of the unit in stereo; an effect defeat switch jack for each channel provides remote defeat of an effect in performance applications.

Circle (167) on Rapid Facts Card

TRF Music Tele Music library

The library is a collection of orchestral and instrumental recordings for a variety of applications. Categories include AV Industrial, Rock, Dramatic, Electronic, Children's, Opening, Closing, Classical and Period. In addition to full selections, 15-, 30- and 60-second versions immediately follow for many selections.

Circle (168) on Rapid Facts Card

Digidesign Sound Accelerator DSP card

The Sound Accelerator is a DSP card for the Macintosh II and SE and provides CD-quality sounds directly from the computer and makes most sound processing and synthesis functions in real time. The company's Sound Designer and Softsynth software have been updated to use the card. With Sound Designer, samples digitally loaded into the Macintosh can be played directly from the computer with 16-bit fidelity. With Softsynth, all synthesis functions are processed in real time.

Circle (180) on Rapid Facts Card

Microtran Co. HD-30 bulk de-gausser

The unit erases high energy, 1,200 oersted tapes to 60dB to 90dB below recorded level, and provides a 3,300-gauss field strength. The unit erases audio, video and digital reels up to 17"x2", cassettes (including U-matic) and cartridges through DI. Duty cycle is 10 minutes on, 10-20 minutes off, with a thermal overload cutoff that resets automatically.

Circle (182) on Rapid Facts Card

D&R Dayner SR console

Designed for sound reinforcement use, the console features the FSS (floating subgroup system), which allows any input module to become a full subgroup module with eight aux send buses, a 4-band EQ and routing to other subgroups as well as the stereo mix buses. The console can be configured from eight to 80 inputs in five different chassis sizes: 21, 31, 42, 59 and 84.

Circle (156) on Rapid Facts Card



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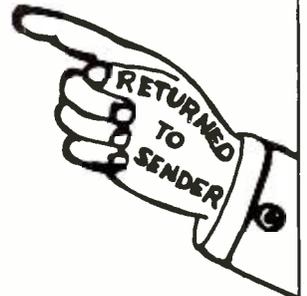
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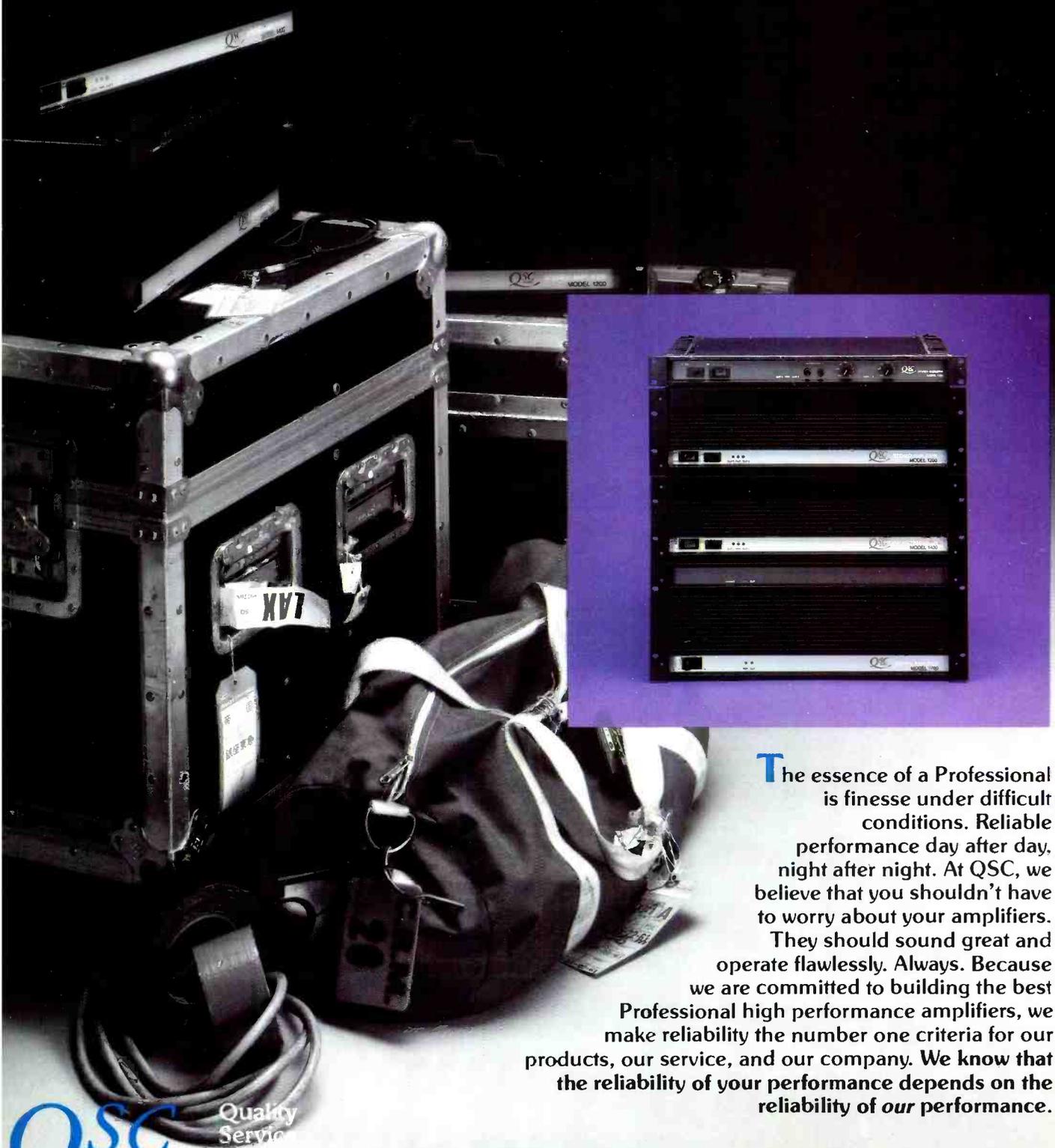
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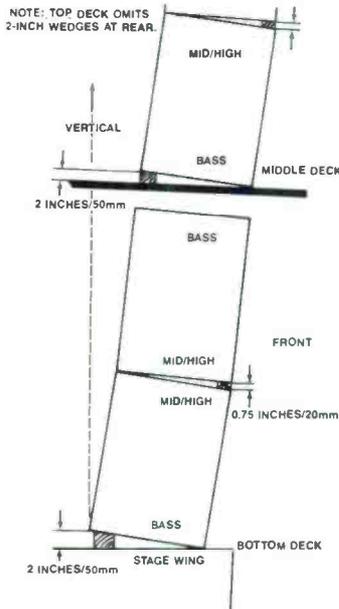
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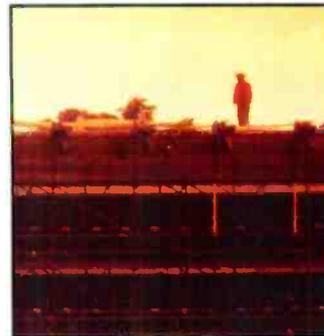
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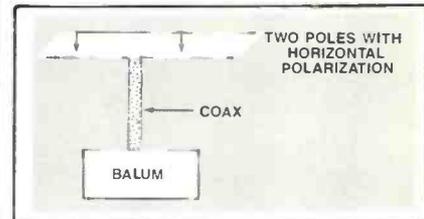
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Iron Maiden headlines a heavy metal festival, featuring a Turbosound-based system, said to be the biggest single-source PA ever constructed.

Volume 20, No. 1

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Artists' expectations, engineering limitations, and other myths of digital recording.

Digital. The word itself conjures up visions of a totally perfect recording process where anything is possible.

Nothing could be further from the truth.

For example, if you treat your digital tape as we've shown below you'll likely end up with exactly what you'd expect. Useless tape.

And how about *sound*: that nebulous, very subjective quality that is, for each one of us, the *raison d'être*? After all, even though we build what we believe to be the world's finest digital machine, the new 32-track DTR-900B, some audio engineers would stack our analog multi-track machines up against it in terms of sound quality any day.

So why did we build the digital DTR-900, and then follow it up with significant new features and improvements in the second generation DTR-900B? And why do we believe it

may be the single most important purchase you will ever make in your business? Simple. It will solve problems for you that no other system can solve. It can cut hours from session times. And it can make your life as a professional magnitudes easier and more rewarding. Here's how.

Just imagine a session where after only a few takes you can send the talent home. You got their best when they were fresh, and now you can do *your* best when you're fresh, and cre-

ative. You use the DTR-900B's session controller to *electronically* assemble the final master from the tracks with no—that's *zero*—sound degradation. (As one studio owner put it, "Often a record becomes what analog makes it—not so with digital.") And no matter how intense the mix-down, the PD format with its powerful Reed-Solomon error correction scheme means you could lose up

to 8 tracks of data and still record and play all 32 channels! So, if you were to lay a cigarette down... no, no, just kidding!

But there's a down side to digital, too. For one thing, there's no friendly tape noise to cover up mistakes, or to add that mysterious "something" to the mix. And the initial cost for a digital machine can be *scary*.

So what's the final mix, or the bottom line, if you prefer? The cost is high, and even though the Otari DTR-900B is a powerful client draw, it's important to consider your return on investment.

But then, a great sounding record is hard to put a price on, isn't it?

It's your decision, but we can help. After all, Otari can offer you the best in digital, *and* the best in analog. Call Otari at (800) 338-6077 X900, for more information. (And if you own a DTR-900, ask us about how the new features on the "B" can be added to your machine.)

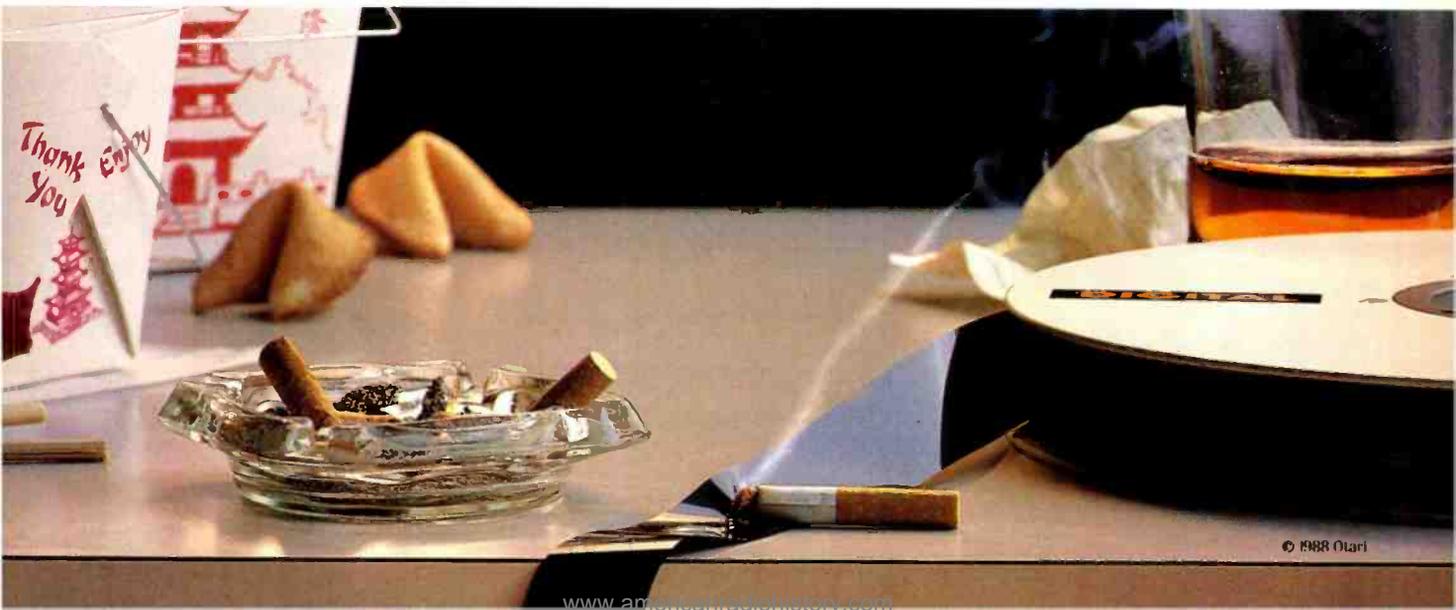


The DTR-900B's new Locator/Remote features improved hardware and software that make the machine faster and easier to operate.

OTARI

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EDITORIAL

We Are Too Loud!

For the last 30 years or so, sound systems have steadily increased their capacity to deliver higher sound pressure levels. This evolution has continued to the point that today SPLs are reaching 130dB at the stage, 110dB in the audience and similar levels in control rooms. These are not peak levels, they are continuous program levels. It is, no doubt, an impressive display of power and technology that allows this much sound pressure to be delivered in a reasonably clean and undistorted manner.

At issue here is not the system's capacity to reproduce crushing levels, but one of responsibility. Who is in charge of the awesome program levels that accompany many live concerts? Is it the house-sound engineer, the artists, the promoters or maybe even the artist's management? I suggest that *ultimately* the house mixers have the control—it's at their finger tips. Therefore, the health and well being of every person in the venue become an added responsibility to consider.

We've all got a thrill from turning up the volume and *immersing* ourselves in the sound of our favorite act. But, we have reached the point that we can now do serious, permanent, physical damage to ourselves and others. [See *AES Journal*, September 1988, Volume 36, No. 9, pp. 686-691.] Some of the larger touring systems are capable of *sustained* levels at or near the threshold of discomfort (120dB+).

There are a few well-established guidelines for evaluating loudness. The Fletcher-Munson or equal loudness curves are familiar to most. They show that average human hearing is flattest at 80dB to 90dB SPL. This is useful because it tells us that, at these levels, the sound system requires less EQ to deliver equal power across a given bandwidth. Another familiar table shows us that the threshold of pain is 140dB SPL, and that the range between 120dB and 140dB represents the sensation of feeling—the level at which sound waves physically move our bodies. In this region, which many systems seem to be reaching and sustaining, the potential exists for

short- and long-term damage.

To my knowledge, no long-term scientific studies have been done to examine the effects of exposure to the high SPL levels that are prevalent in the professional audio industry. Experts in acoustics and medicine can only speculate that continuous exposure to high-power sound systems will indeed cause detrimental effects. Certainly, in heavy construction and with airport ground crews, the conclusive evidence is that hearing impairment does occur after exposure to these industrial noises. Just because music is usually less random than industrial noise does not mean that it is any less dangerous.

Over the years we have conditioned audiences to expect, demand or maybe even crave more level. Is this the "bigger is better" syndrome all over again? Where does it all stop? If, indeed, we have slowly conditioned audiences to want louder concerts, is it not possible slowly to condition audiences, and ourselves, to want a little less painful—more listenable—levels in the future?

Another factor that works into this entire equation is that a very low-distortion program signal of 110dB SPL can be tolerated for longer periods, with less ear fatigue, than a highly distorted one. A distorted 110dB program signal may contain distortion peaks well into the threshold of pain. 130dB of pristine sound is not necessarily any less damaging, it is just more listenable. The acoustic-concussion lows and the dental-drill highs still exist.

I wrote this editorial to draw attention to the fact that something has to be done to temper the enthusiasm of house engineers who operate systems that can actually harm us. If our industry cannot recognize this as an emerging issue, then we may be faced with some unacceptable alternatives.

One solution, which no one wants to think about, is government intervention. It's already happening in the U.K. The U.S. government is imposing restrictions on the levels that can be generated—at outdoor concerts in particular. Individual states, cities and municipalities have considered or are considering restrictions on noise pollution that fall under OSHA's umbrella of health and safety standards. Using this scenario, a regulatory agent will be looking over your shoulder, monitoring the sound levels and personally controlling the output.

Another way the government could get

involved would be through manufacturing restrictions. That is, manufacturers may be forced to limit future developments in power handling. It would be pretty cumbersome to configure a 50kW system with 250W amps. You're probably thinking, "They would never limit us to only 250W amps." Oh, wouldn't they? Take a look at what the FCC did when they allocated a new group of frequencies for the wireless mic guys! [See "Coping with Wireless Microphone Systems" on page 52.]

What, then, is the solution? The answer is self-regulation. Stop and think—there is a real chance that OSHA or some other regulatory agency will step in and place restraints on the concert sound industry to control the levels.

Self-regulation is really quite simple. It starts with an acceptance and understanding of the problem. Of course, if you don't think a problem (or potential problem) exists, maybe a hearing test is in order. Next, get an SPL meter and check the levels at the stage, at the mix position and at various locations within the intended coverage area. Don't stop yet. Now go beyond the intended seating area 100 yards, a quarter mile, even a mile in all directions. Plot the energy dispersion coming from the PA. From this information, not only will you be able to monitor the levels at the mix position continuously, but you'll also be able to do a reasonably good job of predicting the levels elsewhere.

Redesigned coverage patterns can also help to concentrate higher levels throughout the intended listening area without having *all* the SPL coming from the stage. Because of the inverse square law (attenuation of SPL vs. distance from loudspeaker or sound source), levels at the stage are often pushed to the threshold of pain. This is necessary to deliver 100dB to 110dB levels only a few meters away. Maybe it makes sense to find ways of distributing the sound more evenly throughout the audience in order to maintain a lower, but more consistent, SPL?

If you have specific examples or experiences of government interference of this type, please write and share them with us. There's nothing like first-hand experience to convince the skeptics that this is a real threat to the live sound industry today—and possibly the studio tomorrow.

REP

Michael Fay
Editor

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LETTERS

Phase shift experience

From: J. Russell Lemon, Carlsbad, CA.

Last week, I received my first copy of RE/P, and the letters on phase shift brought back many memories of the studies I did on that subject in 1974 and 1975. The interesting thing is that there is some truth in both sides of the argument.

In the early 1970s, I had access to a digital audio system that was developed for the purpose of testing real-time, digital signal processing techniques (software). Input was a digitally recorded source, and output could be recorded or monitored through a speaker. A high-speed computer that could create any amplitude and phase characteristic desired (within the limitations of the FFT used for the complex convolution), was used for the tests.

The results of those tests indicated that the ear was relatively insensitive to phase, and that hundreds of degrees of phase shift in mid-band (500Hz to 2,000Hz) were required to be heard. Similar results were reported 12 years later by J.A. Deer and P.J. Bloom in AES preprint 2197, called "New Results for Perception of Phase Distortion," which was presented in March 1985 at the 77th AES Convention in Hamburg, West Germany.

Many years later, I made the comment that the ear was relatively insensitive to

phase, and Don Davis told me that I didn't know what I was talking about and to reverse my speaker terminals and listen to the difference. I did and found that with certain types of music, at a sufficient playback level, that there was a difference.

To further experiment, I connected my headphones to my computer and programmed several waveforms. As expected, symmetrical waveforms and their inverse sound the same. But a waveform with a large positive pulse is brighter than its inverse. Similarly, music from a solo instrument with a large unidirectional impulse is also the most sensitive to 180-degree phase shift. That is, human voice, brass and, of course, percussion sound brighter when the leading impulse is positive (speaker moves out) as it is originally created.

With a negative impulse, the edge is gone and the sound becomes dull. The effect is level-dependent and obvious at levels above 90dBA. The effect is masked if any component in the audio chain scrambles the phase. If the monitor system is phase linear, a microphone at the listening position will show a large positive impulse when human voice, brass or percussion are reproduced and the microphone output is displayed on an oscilloscope. If headphones are used, the oscilloscope will also show the same large positive peak at the phone jack.

An interesting test is to use two sine

wave audio generators and set one to 1,000Hz and one to 2,001Hz. If you put one frequency into one ear and the other frequency in to the other ear, no beats are heard. But listening to both at the same time produces beats. If the ear was not sensitive to the phase between a fundamental and its "second harmonic," you would not hear beats.

After these experiments, I realized that my first tests lost their validity because of the 8th order elliptic filters used on input and output. These filters had a rolloff of more than 500dB per octave and a large resulting relative phase shift. In other words, the test differential phase shift did not significantly change the peak to RMS ratio of the already scrambled signal. Before digital, commercial devices to add second harmonics to a mix were popular, to add the edge back into the music when recorded on systems that lost phase information.

Perhaps the best way to summarize the above observations is to say that the ear may be unable to sense phase, but it is sure sensitive to the phase relationships between tones and that this sensitivity is level-dependent. But then, this can be said of any other non-linear device.

RE/P

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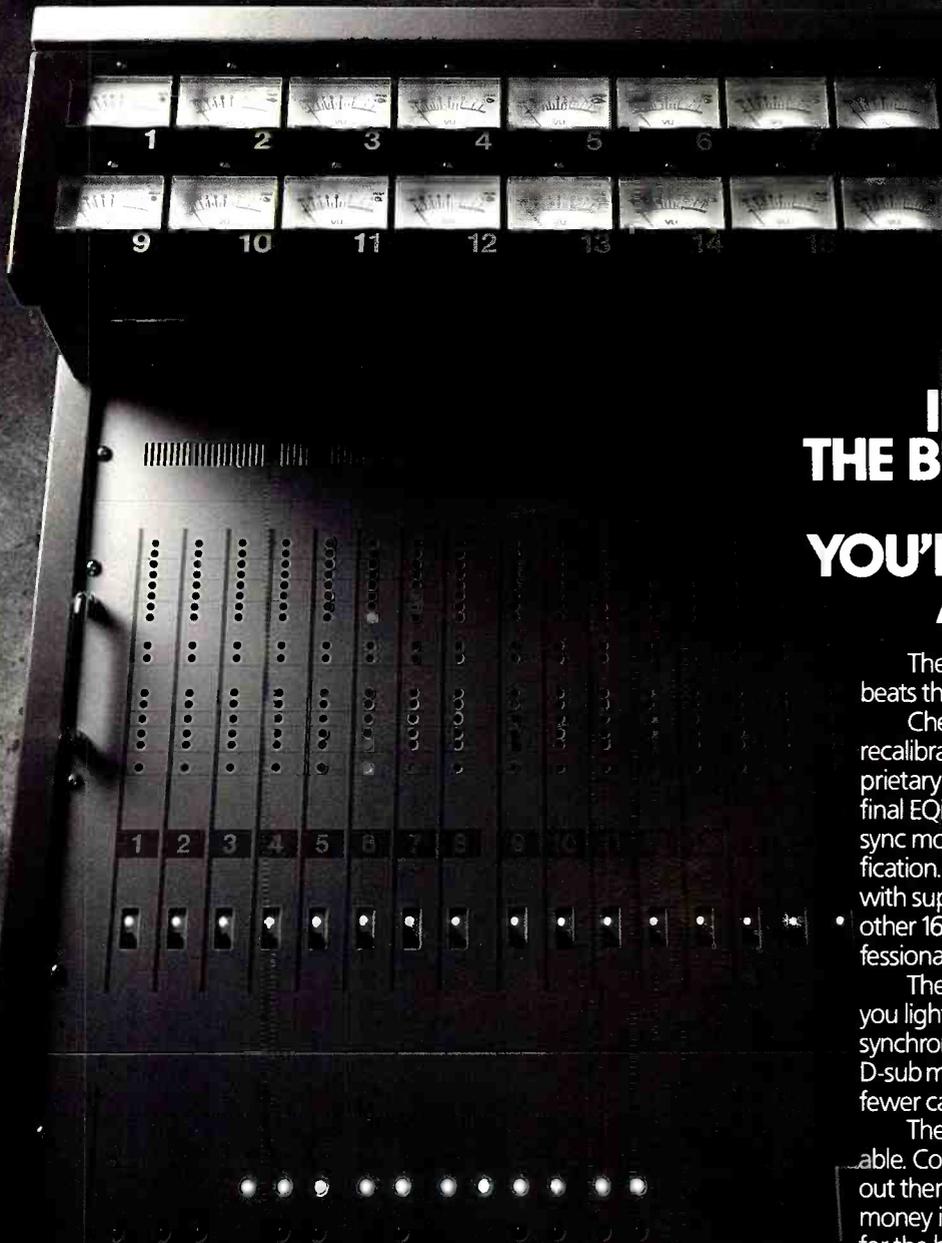
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RE/P to debut new departments

Beginning in the March issue, *RE/P* will debut two new departments: the "Engineer/Producer Index" and "Tracks."

The "Engineer/Producer Index" is a monthly listing of engineers/producers and the commercially released projects that they have worked on. It is open to engineers and producers involved in all types of audio production. Listings will contain the engineer's name, address and phone number, the five most recent projects performed and codes classifying the projects.

"Tracks" spotlights the studios and facilities dealing in commercial production and post-production, the two fastest-growing business categories in the industry. Listings will contain the facility's name, the five most recent projects that occurred at the facility, and the engineers who worked on the projects.

Beginning in this issue, postage-paid reply cards for the "Engineer/Producer Index" and "Tracks" will appear in the back of the issue. To be included in the departments, respondents should fill out the cards and return them to *RE/P*.

EV expands engineering facility

Electro-Voice has relocated its engineering department to a new 28,000-square-foot building, providing working facilities for more than 50 EV engineers, technical assistants and support staff. A large prototype fabrication, assembly and testing area is located at the facility, which includes a metal shop a wood shop, and a listening room. The building is about four blocks from EV's parent facility in Buchanan, MI.

Alan Watson, EV's director of engineering, said the facility would allow the company to expand its facilities and develop a greater number of products.

Ampex opens test facility

Ampex's Magnetic Tape Division has opened its new test facility in Opelika, AL, near the corporation's processing plant. The \$2 million, 9,600-square-foot test center has eight separate labs for 1-inch video, cassette video, audio, instrumentation, plastics, physicals and microscopy. State-of-the-art recorders and signal generating/processing equipment will be used to analyze tape performance for a wide variety of formats and user conditions.

The lab will be used to evaluate competitive products as part of Ampex's Audit Program to obtain information about every format and brand of audio, video and instrumentation tape.

Passport to support NeXT system

Passport Designs has announced plans to support Steve Job's NeXT computer system, which was unveiled in early October and is said to be the first computer capable of acting as an all-in-one music workstation.

In addition to porting over the software titles, the company will design new applications that take advantage of the NeXT's multitasking and digital signal processing capabilities.

The NeXT computer is capable of generating a stereo digital audio signal at the standard 44.1kHz playback rate. Also included is a built-in coprocessor, a digital signal processor and a Motorola 6830 microprocessor running at 25MHz.

Aurora Systems files petition

Aurora Systems has filed a voluntary petition in the U.S. Court in San Francisco, and a plan of reorganization that provided interim financing, settlement of all outstanding claims, warrants to purchase shares in Chyron and acquisition of 100% of the new Aurora capital stock by Chyron.

These events were the results of several months of negotiations with Chyron, in which Aurora was unable to secure shareholder approval of offers by Chyron to purchase a majority equity position in Aurora. This incorporation of Aurora into Chyron will provide the former with the necessary financial, technical and marketing opportunities to profitably exploit the opportunities open in the TV industry, as well as to market their newly developed products.

News notes

ROR Audio Research is a new company that manufactures close-field monitors. Recording engineer Shimon Ron is the company president; the company's address is 161-14 Union Turnpike, Flushing, NY 11366; 718-969-3660.

Audio Animation has relocated to 210 W. Magnolia Ave., Knoxville, TN 37917; 615-544-0458.

J.L. Cooper Electronics has moved its

warehouse and corporate offices to 13478 Beach Ave., Marina Del Ray, CA 90292; 213-306-4131; fax 213-822-2252.

Digital Audio Research has moved its offices from the San Francisco area to Hollywood. Lee Bartolomei has been promoted to regional sales manager. The company's new address is 6363 Sunset Blvd., Suite 802, Hollywood, CA 90028; 213-466-9151; fax 213-466-8793.

New England Digital has formed a new dedicated sales and marketing group within Harmon International, the European NED distributor. The group is intended to enhance the company's European market position as the continent moves to a unified marketplace in 1992. Mark Terry, NED's director of marketing, will lead the group.

MCA Records has purchased two Sonic Systems with NoNOISE software from **Sonic Solutions**, for use in mastering vintage recordings for CD reissue.

Imrex Financial Services, a financial services company geared toward the film, video, broadcast and audio industries, has opened a California office located at 5777 W. Century Blvd., Suite 265, Los Angeles, CA 90045; 213-670-1009.

At the AES convention, **BASF** honored two companies with its annual Inventors Award, given to companies deemed most instrumental in enhancing the quality, distribution and sale of music cassettes. Receiving the award were Concept Design, Burlington, NC, and WEA Manufacturing, Olyphant, WA.

Neve has signed a contract with Full Sail Center of the Recording Arts to teach applications on the V Series Console and the Flying Faders automation system.

Soundmaster USA has opened a New York technical support office, located at 120 W. 88th St., New York, NY 10024; 212-787-5832; fax 212-787-8888. On the West Coast, the company has installed a second Integrated Audio Editing System at ABC-TV in Hollywood.

Manny's Music has acquired Audiotechniques, a pro audio dealer based in New York.



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MANAGING MIDI

By Paul D. Lehrman

Gotta Resolution

I received an interesting and somewhat distressing letter from a reader recently about a problem he was wrestling with that seemed to have dire consequences for all MIDI users. He had approached various figures in the MIDI world with his problem and had yet to get a satisfactory response, so he tried me.

It seems that he was having difficulty getting his hardware sequencer, when running from an external sync source, to resolve events any closer together than 1/24th of a quarter note. Although the sequencer was capable of much higher resolution when it was using its internal clock, when it was clocked externally, any notes that were supposed to be spread out over 1/24th of a beat sounded simultaneously—in other words, fast runs became groups of chords.

His letter said he had experimented with various sequencers, both hardware and software, and had come up with the same results, and was therefore convinced that this was an inalterable fact of MIDI life—that because MIDI clock pulses are 1/24th of a quarter-note apart, no sequencer, no matter what its internal resolution, could hope to do better than that when it was being synced externally.

This came as something of a shock to me, as I've been using external sync for years, and never noticed my arpeggios clumping up or my carefully "lagged" snare drum hits catching up with the beat. If this reader's hypothesis were true, MIDI music would be even more mechanical-sounding than it already is, and whole portions of the music industry—like electronic film scoring—would probably not exist.

It is true that MIDI clocks are sent at the rate of 24 per quarter note, and they are the only timing reference an externally clocked sequencer has to work with. At

a tempo of 120 beats per minute, this results in a resolution of about 21ms, which is adequate for some music, but not nearly all. Some sequencers developed in the early days of MIDI did not resolve externally synced music any closer than this because nobody really thought about it (and the hardware sequencer on which the reader first encountered the problem dated from this period).

By the time the first professional software sequencers came out, however, many developers realized that this could be a serious limitation, and so they incorporated interpolation into their programs. Today, most sequencers use interpolation—including one of the programs my reader said he had tried the experiment with.

Interpolation means simply that the sequencer maintains its high internal resolution, and uses incoming timing pulses merely as a *reference* for its own timing circuits. So even though the incoming MIDI clocks are only running at 24ppq, the sequencer's (or computer's) internal timing circuits are still working at full speed.

Say, for example, a sequencer has a resolution of 240 "ticks" per quarter note (or 10 ticks per clock), and an event is supposed to take place seven ticks after a beat. The sequencer will measure the time interval between the *previous* MIDI clocks (call it *t*), and then output the event at a time equal to beat + $t \times 7/10$.

"Ah," you say, "what happens when the tempo is changing?" Well, yes, there could be a problem then, but it would be minimal. If the tempo of a sequence were to speed up drastically, and there were high-resolution events immediately following the tempo change, those events might well come out late. But the sequence itself would not slow down (any events on or after the next clock pulse would be right on the money), and so the effect would be a "bunching up" of events that would last only for 1/24th of a beat or so (depending on how tight the interpolation algorithm is).

When using MIDI Time Code, the approach is approximately the same, although it provides much finer resolution: There are 120 quarter-frame messages per second, which translates into about 8ms per message. (For standard MIDI clocks to achieve this resolution, the tempo would have to be 300bpm.) Despite the increased resolution, since MIDI Time Code timing messages don't normally fall on

musically significant times, interpolation is that much *more* important: If a sequencer couldn't interpolate, the music would sound very strange indeed, locked up to quarter-frame messages that bear no relationship to the actual tempo.

So, to my concerned reader, this message: relax. The walls of MIDI are not going to come tumbling down. Replace your old hardware sequencer with a newer one, and you'll find your arpeggios intact. If you already have a newer sequencer, it may have selectable resolution—make sure it's set as high as you need it.

Another item that arrived recently in the mail is a little book that I can recommend to anyone involved in computers and music, and that is "The Electronic Musician's Dictionary," by Craig Anderton, published by Music Sales Corporation. Craig is one of the best-known writers in the field of electronic musical instruments, and is editor-in-chief of *Electronic Musician*.

The dictionary is a handy volume containing brief, but informative, definitions of just about any term you are likely to encounter as you wade through the morass of jargon that is electronic music. In fact, one particularly apt definition is of the word "jargon," and includes the sentence, "Jargon is sometimes used to intimidate beginners." This is not an electronics dictionary, and you won't find definitions of Butterworth filters or half-wave rectifiers, but you will learn that "boot" (as in "boot the computer") is short for "bootstrap loading", and that "k" as an abbreviation usually means 1,000, while "K" means 1,024!

You'll also find precise and very useful definitions of synthesis terms like "hard sync" and "FM," as well as the origin of acronyms like DIP, DRAM and DIN. (But is "kluge" really German for "rat's nest"? It ain't in my German dictionary.) Best of all, Craig has described the book as one he intends to work on "the rest of my life," with additions and upgrades as the field develops and expands, and he invites readers to contribute their ideas. It's a great start for a great idea.

REP

Paul Lehrman is RE/P's electronic music consulting editor and is a Boston-based producer, electronic musician and free-lance writer.

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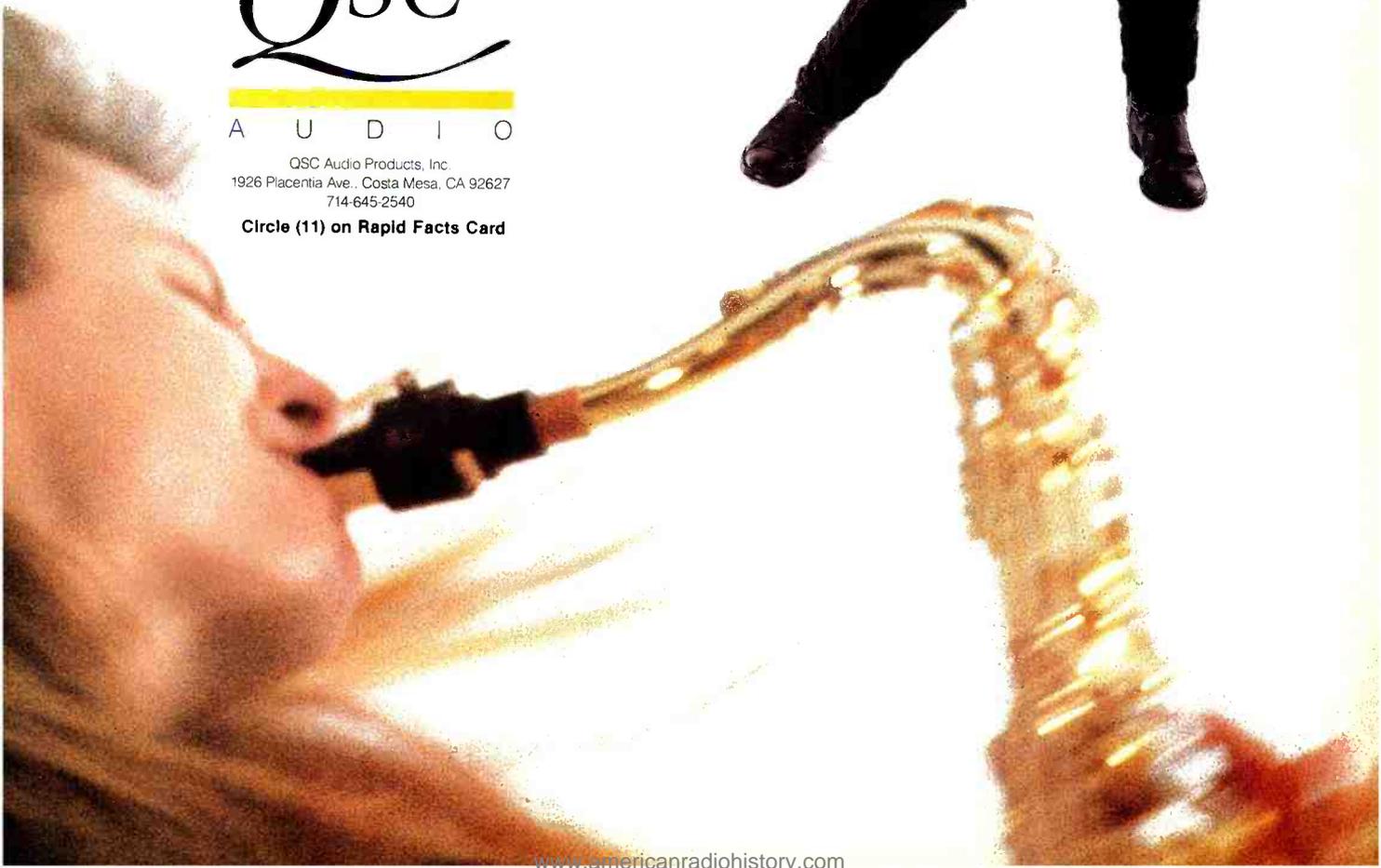


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SPARS ON-LINE

By Richard Trump

Exploring Sound Reinforcement Opportunities

There is a tendency among the general public to see the audio industry as a whole, without fully understanding the specialty fields that we, as professionals, have chosen to pursue. It's not unusual for someone on a church board to equate your recording business with being able to solve the audio problems of a highly reverberant sanctuary. Or, a friend with a bar might expect you to be able to solve his sound system problems. Perhaps your best ad agency client has a presentation to make and wants to rent the speakers from your studio. Sometimes I feel like a doctor whose friends are always mentioning their mysterious aches and pains.

When confronted with such queries, the most obvious and wisest response should be, "Sorry, can't help you. That's beyond my expertise." Unknown territory is best left unknown. On the other hand, such in-

If a more qualified organization is available, let that company handle the job.

quiries usually trigger a different response: "Sure, let's give it a try." There's nothing like the temptation of extra cash to shift the direction of a business. Somewhere in between lies the indecision—"Gee, I hadn't really thought of doing that before. Let me think about it."

Having given all three responses in the past, I intend here to relate some background and reasoning that has worked well for the long-range goals of my business.

Richard Trump is regional vice president/treasurer of SPARS and president of Triad Productions, Des Moines, IA.

Do you have the knowledge and experience necessary to tackle a sound reinforcement job? Even if you know studio acoustics quite well, the demands of a reverberant hall, noisy night club, outdoor setting or large arena offer considerably different challenges. Instead of facing the studio problems of proper imaging, extending bass response or providing the right outboard gear, you will need to deal with speech intelligibility, loudspeaker driver alignment, coverage and acoustic gain.

Designing and installing a good sound system is an increasingly complex process.

Sometimes, because of market size or the lack of other resources, you may be the only person for the job.

Increased industry knowledge, not changes in the laws of physics, has made system performance more predictable. Make sure that you have the knowledge necessary to meet the needs of your customer.

Do you have the appropriate equipment? One reason that system performance has become more predictable is the availability of new tools to measure and analyze the difference between "adequate" and "excellent." These tools are generally not part of a recording studio's facilities, so your results are left to be judged subjectively. If the results are truly good, you are probably safe, but when they are not, corrective measures can be difficult to prescribe.

Do you have something special to offer? You may have unique equipment or special qualifications for a particular job. If a system's primary function is in your area of expertise, you have good reason to be involved. For example, a sound reinforcement project may have some incidental need for recording.

Are you an "expert" by default? Sometimes, because of market size or the lack of other resources, you may be the only person for the job. If you are a courageous soul and really can't think of someone better suited for the task, you may be the best person for the job. But always weigh the customer's options carefully and try to make the decision that

is best for them. A wrong move could seriously damage your overall reputation.

Are you desperate? You just had a major session cancel, the equipment payment is due next week and a client wants to rent a playback system. Saving your own hide may be reason enough to cross over into a new business. Just remember that the precedent is likely to create more requests and will make refusals more difficult in the future.

Annually, we provide the on-air mix for a two-site, 21-hour regional Variety Club telethon. It is necessary to supply house and stage mixes at both sites. At one venue, the room is small enough for us to handle all audio. At the larger facility, we understand our limitations and subcontract reinforcement mixing and reproduction.

We keep a couple of small presentation playback systems available for client demos. The equipment requirements for the playback of tapes in a typical boardroom setting are similar to those found in a production studio. Since we often have future work at stake, it is appropriate that we provide a system that properly displays our product.

Saving your own hide may be reason enough to cross over into a new business.

My company certainly doesn't profess to fill the role of expert that some people assume we hold in our market, but we have found a niche that encourages us to continue expanding. Since our reputation is always at stake, we refer more sound reinforcement work than we accept. We also have learned the benefits of collaborating with other professionals when it is called for. A general guideline might be: If a more qualified organization is available, let that company handle the job. If not, be aware of your limitations, and consider carefully whether you are capable of satisfying the customer's needs. Then, proceed with cautious enthusiasm.

RE/P

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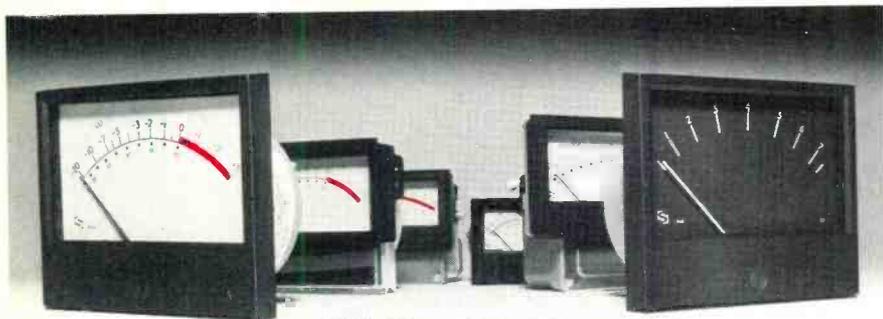
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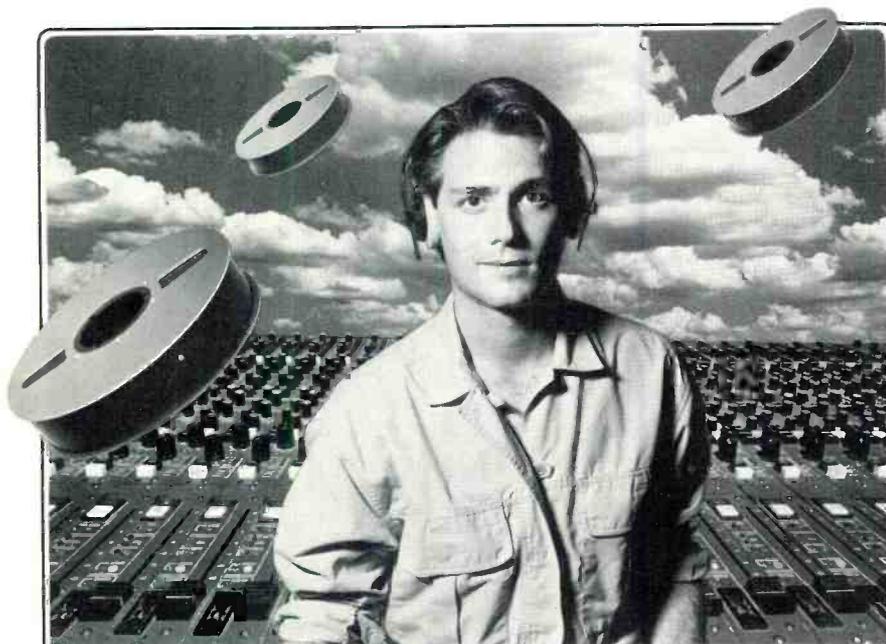
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ucts more friendly, intuitive, reliable and cost-effective to reach the broadest possible market.

When this column first started, I brought out a point that merits repeating. If you work in a recording studio, you're already dealing with computers every day. Most of today's consoles, tape decks, signal processors and instruments have some type of computer incorporated in the heart of the machine, if they aren't actually full-fledged computers. Trident's Di-An console and the Emulator III, to mention just two, are really computers with specialized interfaces and additional hardware. The Casio Data Bank on your wrist is a computer. Your programmable VCR and microwave...well, you get the picture.

The mouse-driven point-and-click approach has taken over as a more intuitive way of interfacing with computers.

Computer technology is giving us the ability to tap instantly into the entire wealth of man's accumulated knowledge. CD-ROM and on-line databases allow us to access encyclopedias, dictionaries, reference materials and, indeed, entire specialized libraries that contain staggering amounts of information.

Modern audio and video technologies have blurred, resembling computer technology more and more with each day. The computer has become teacher, entertainer, business partner, muse, artist and much more. An emerging technology, of special interest to many people, is interactive education with the computer providing audio-visual tutorials tailored to each user. With the wealth of information available and the almost limitless learning capabilities of the human brain, this technology is pushing us head-over-heels into our future, whether we like it or not.

If you still have a touch of unfounded computer fear, stay tuned to this column—it was developed to help demystify computer terminology and technology. Next month, we'll return to our discussion of telecommunications.

REP

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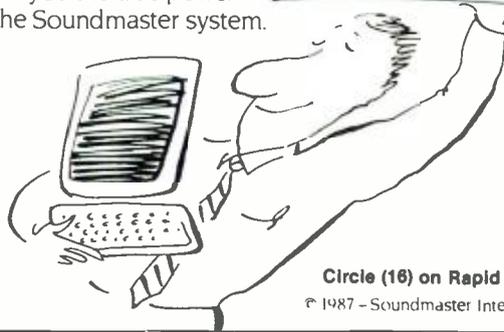
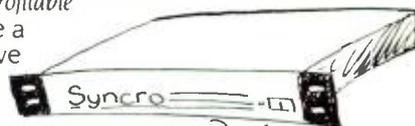
Our Syncro™ machine controller can synchronize, not just chase, from a fraction of, to several times play speed. With Smart Sync, *you* control literally thousands of varispeed

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Sound Thoughts on Live Performance

“Creating the subjective experience of live performance sound belongs to the artists and sound designers. Our creativity comes in building speakers and systems that give the truest possible account of what the performers produce.”

Meyer Sound has devoted itself to designing, manufacturing, and refining components that deliver superb sonic reproduction and expand the artistic possibilities of professional sound reinforcement.

Meyer strives for professional sound quality that is predictable and neutral over an extended lifetime and across an extended range. Even after extended use, Meyer Sound performance is never compromised.

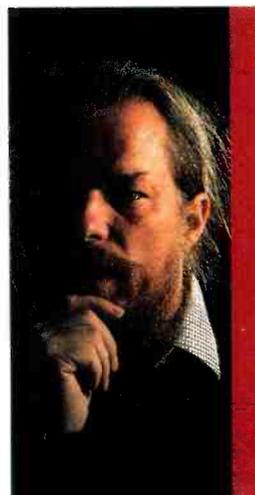
As a consequence, Meyer Sound products have earned a reputation for the highest reliability in the industry. All are guaranteed to meet or exceed specified performance levels when properly installed.

“The general public’s sophistication keeps growing. Soon, if we have our way, the audience will demand the same accuracy in live performance that they get from home recordings.”

Instead of second-guessing the tastes of the market, Meyer produces sound systems that most truly represent the character of the signal they receive, leaving artistic control where it belongs—with the artists and sound designers.

Meyer takes a conservative view of exotic loudspeaker materials, preferring to use proven materials in new, more elegant ways.

Every part of every component undergoes rigorous, comprehensive testing. Meyer Sound controls all aspects of the system design—if not by manufacturing, then by modification and refinement to Meyer’s stringent standards.



John Meyer, Founder and President, Meyer Sound Laboratories

“As expectations rise, our performance standards have to rise even higher. And the only way to increase performance is with increasingly sophisticated measurement.

“Which is how we found ourselves also in the measurement business.”

Meyer originally intended to be solely a manufacturer of high-quality, rugged and reliable loudspeakers, expecting others to pioneer and perfect testing equipment. But the need to accurately measure the performance of Meyer components individually and in arrays outgrew the quality and resolution limitations of available testing equipment.

To make sound work in spaces, Meyer Sound Laboratories developed by necessity its own testing technology and methods.

John Meyer, his engineers and his designers have authored several definitive works, and research remains an integral, driving force behind all production.

SIM™ Equalization

SIM™ equalization is the logical result of Meyer’s commitment to uncompromised sound quality through sophisticated measurement. The non-intrusive SIM technology uses real-world program material (either voice or music) as the test signal. Working interactively with the sound designer, a Meyer SIM engineer helps create superior clarity for every member of the audience.

Ask your distributor for more information on SIM equalization, or call Meyer Sound Laboratories direct.

John Meyer’s involvement in loudspeaker design began in 1967 when, as a technician for a Berkeley, California Hi-Fi supplier, he set out to discover why a leading manufacturer’s drivers kept tearing themselves to pieces. Further investigations convinced him that the market sorely needed a class of rugged professional speakers that would maintain their characteristics over time.

Research in Switzerland in the early seventies secured his knowledge base. In 1972, Meyer developed the JM3 all horn loaded tri-amp system with rigging, which was the standard for Broadway shows until the introduction of the UPA in 1980. From 1973 to 1979, Meyer sought out the best available parts and designed the first Ultra Series™ reinforcement speakers. In the decade since, John Meyer has established Meyer Sound Laboratories at the forefront of professional reinforcement technology.

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RE/P



Electronically Controlled Speaker Systems

By David Scheirman

A look at the assets and liabilities of integrated controller/speaker systems.

In less than 10 years, an ever-expanding group of commercially available electronically controlled loudspeaker systems has appeared in the sound reinforcement industry. Ranging from lightweight, compact audio-visual boxes to large-scale modular enclosures suitable for concert sound use, this new breed of speaker systems is marketed as a cure-all in some circles. The various system performance parameters and design philosophies differ significantly, as a closer investigation will reveal.

Ever since the development and introduction of commercial sound systems for use with public address, live music and playback reproduction in sound reinforcement applications, designers have searched for the optimum way to package the loudspeaker components.

An incredible array of loudspeaker systems is available today, in every conceivable size and shape. The desire for compact, high-performance systems has led many manufacturers to concentrate on the development of integrated systems that not only match the transducer components to the box enclosure, but also rely on dedicated electronic circuitry packages to wrestle every last ounce of "woof" and "tweet" from the loudspeakers.

These electronically controlled speaker systems vary somewhat in design

philosophy. What they all have in common is that the user has more than just speakers and a box to deal with; a proprietary active signal-processing device is usually needed to make the speaker system operational. (See Photo 1.)

Some manufacturers use the term *processor* to describe their dedicated electronics packages. While taken from the commonly used term *signal processing*, this nomenclature also provides a convenient marketing "hook." It links us with the technologically minded terms *data processing* and *microprocessor*, both of which have the subliminal implication that our all-knowing and error-free friend, the computer, is hard at work on our team, although this is not necessarily the case.

I will use the term *controller* to describe these dedicated electronics packages. Whatever a manufacturer's name for a special signal processor, its purpose is to control the performance of the speaker system, using predetermined operating parameters and/or real-time *sensed* information.

A number of functions are being assigned to these different black boxes, but a careful, logical study of a system manufacturer's literature should provide a reasonably good idea of the design philosophy behind a particular system. Some are merely electronic crossovers with equalization enhancement. Others incorporate group signal delay, phase inversion, bandpass limiting, component protection functions, and more. All, however,

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

are controllers in the sense that their purpose is to instruct the loudspeaker system how it is to perform. The controller takes an input signal, divides, filters, and cuts or boosts it, and offers the system user more optimum loudspeaker performance, in most situations, than could be otherwise obtained.

The controller's task is to integrate a set of carefully selected speaker components with a wide variety of input program material. A combination of loudspeakers that is optimum for reproducing just the human voice may have very different characteristics from those most suitable for certain types of instrumental music.

And, different loudspeakers respond in varying ways to different input signals. For example, drum rimshots, brass groups and cymbal clashes can lead to problems with loudspeakers whose impedance drops to low values at high frequencies (as will happen with low-frequency components operating in the upper end of their response region).¹

The controller is called upon to deliver the best possible sound under all conditions automatically, with any program input. Users of such systems often include rental companies that provide sound equipment temporarily for public speaking, acoustical and symphonic music, or rock concerts. The ideal control electronics package will take it all in stride.

Indeed, some of the systems detailed here are severely compromised in terms of design intention if the prescribed controller is not used. While it also may be possible to wire the output of any crossover to the power amplifier channels and drive a given speaker system in that manner, performance of the whole system will suffer. Understanding a controller's intended functions helps up to realize its true value.

Is there really a concerted effort on the part of system designers to find ways of making packaged loudspeaker systems smaller, more efficient and better-sounding—all in the interest of good audio? Or is a conspiracy under way to force end-users to purchase expensive, mysterious and unwanted black-box "controllers" with each different brand of speaker system, with the motive being higher profits for the manufacturers?

You be the judge. Don't rely on this feature, alone, however. And don't rely solely on the makers' specification sheets and advertisements. Listen to these systems and use them in professional situations. Compare them with traditional systems and with each other. Keep the laws of physics in mind as you listen to the marketing claims, and make certain that a specific system will meet your needs in the real working world.

Speaker systems are available today that definitely offer higher average sound pressure levels, less mass and more intelligence than traditional, uncontrolled speaker systems. You alone must decide whether or not the intelligence, or signal-shaping ability, that's built into these electronic controllers is compatible with your requirements.

History

Early speaker systems were unavoidably bulky. The large two-way, horn-loaded bins such as those developed at RCA and Western Electric in the 1930s (for

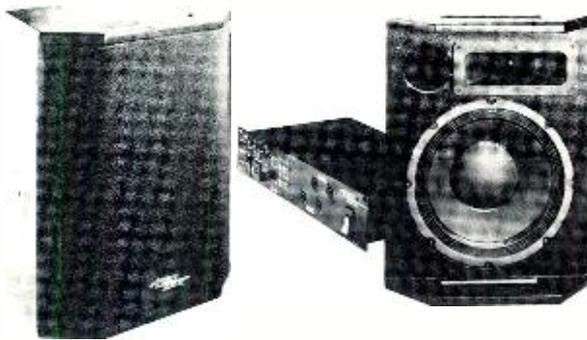


Photo 1. Electronically controlled speaker systems are currently available from many manufacturers. Shown here are a pair of compact enclosures with controller from Apogee Sound.

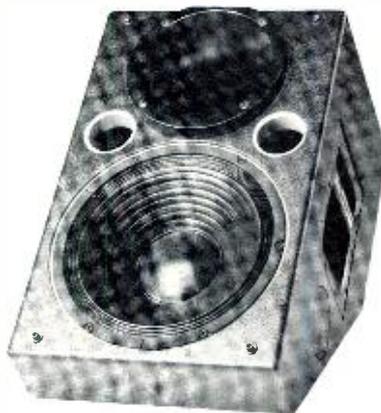


Photo 2. The UltraMonitor (now the UM-1A) is a compact two-way floor monitor developed by Meyer Sound Laboratories and first released in 1979. This product proved to the industry that electronically controlled speaker systems are marketable.

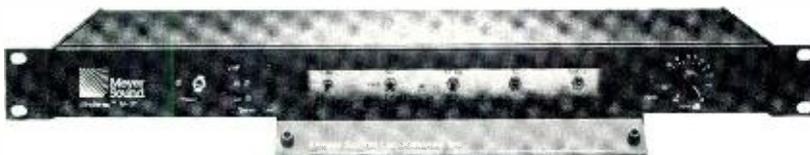


Photo 3. Some controllers, such as Meyer Sound Laboratories' M-3T, offer signal delay for optimum high-frequency phase alignment.

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use in movie theaters) are a startling contrast to some of the compact speaker systems available today. The system package for two cone-loudspeakers, a single compression driver mounted on a multicelled horn and a passive, internal crossover would fill an 8-foot truck bed—but was intended for permanent installa-

tion. While overly large, such speaker systems were relatively efficient, and were capable of producing respectable sound pressure levels with only a 10W or 20W power amplifier.

As packaged loudspeaker systems became smaller, speaker components became more durable and amplifiers

became more powerful, some designers began to focus on an integrated systems approach. With this design direction, the optimized crossover frequency and slope configuration for specific loudspeakers was sought out and provided. In the search for a good compromise between transducer reliability and sonic fidelity, the causes of loudspeaker failure under stress were examined. Means of offering both fail-safe system operation through speaker protection, and enhanced performance through variable, level-sensitive equalization were sought.

In the late 1960s, Bose Corporation pioneered the use of dedicated, active equalization for use with specific compact speaker enclosures. A low-frequency enhancement circuit packaged in a separate black box was available for use in conjunction with small-cone loudspeaker systems. Even in this electronically controlled age, Bose prefers to stick to this concept of active EQ with no motional-feedback or sensing circuits in its Acousti-Mass system.

In the mid-1970s, McCune Sound (San Francisco) fielded the proprietary JM-3 system, perhaps the first integrated, full-bandwidth controller/speaker system to see professional use in sound reinforcement. Developed by John Meyer for McCune, the innovative system saw much use in Broadway musical productions and live concert applications. Meyer began independent development projects on subwoofers and other speaker systems that required special signal processing.

In 1979 his company, Meyer Sound Laboratories, introduced the UltraMonitor with its own control electronics unit. (See Photo 2.) Combining crossover, equalization and loudspeaker system protection functions in a single package, this system created quite a stir upon its introduction. System users noticed that it offered relatively high average sound pressure levels for such a compact enclosure, along with seemingly extended low- and high-frequency response characteristics.

A new standard was being set in both the high average sound pressure level and the extended response available from compact speaker systems. A new standard was also being set in pricing for loudspeakers in a box. Accepted into the professional marketplace, the Meyer Sound products established a reference point that other manufacturers began to notice.

In the 10 years or so that have followed, the group of suppliers that offer electronically controlled speaker systems for sound reinforcement use has expanded dramatically. Let's take a look at the various operating functions attributed to



Photo 4. Celestion's SR speaker system includes compact enclosures housing two 8-inch wide-range cone loudspeakers. Optional subwoofers and unique mounting hardware round out the line.



Photo 5. This controller (Renkus-Heinz X31) has LED displays for temperature and excursion protection circuitry activation in low, mid and high bandpasses.

REAR PANEL:

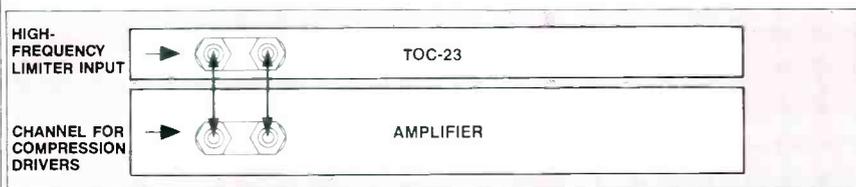
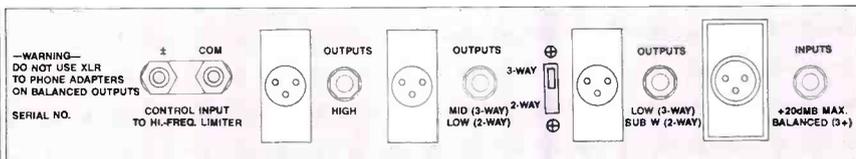
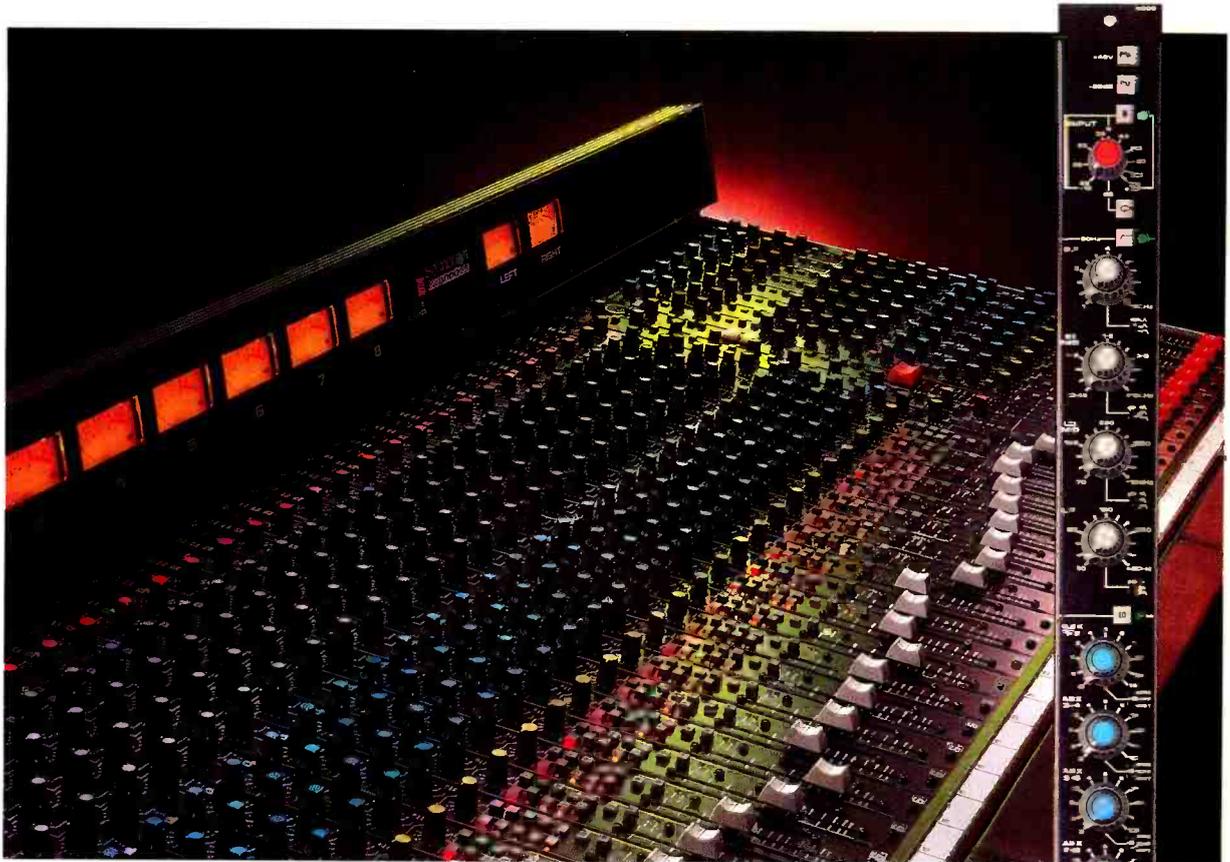


Figure 1. Controllers from PAS use sensing connections from the output of the high-frequency power amplifier channel to trigger limiting function.

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Designed to fit into almost any tour or live installation, the Series 8000 gives you a wide range of available features, such as VCA subgrouping, LED input metering, our unique three-way panning option and an input preamp similar in specifications to much more expensive recording consoles. Other choices include EQ options ranging from four band with mid sweeps to a full four-band parametric on each input channel, giving you the subtle control that yields dramatic results.

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The Series 8000 input module is available with LED metering or VCA subgrouping, and incorporates four-band parametric EQ giving you the subtle control needed to produce dramatic results.

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When you specify Soundcraft's Series 8000 for theater, touring or production, rest assured you're making a clearly superior decision.

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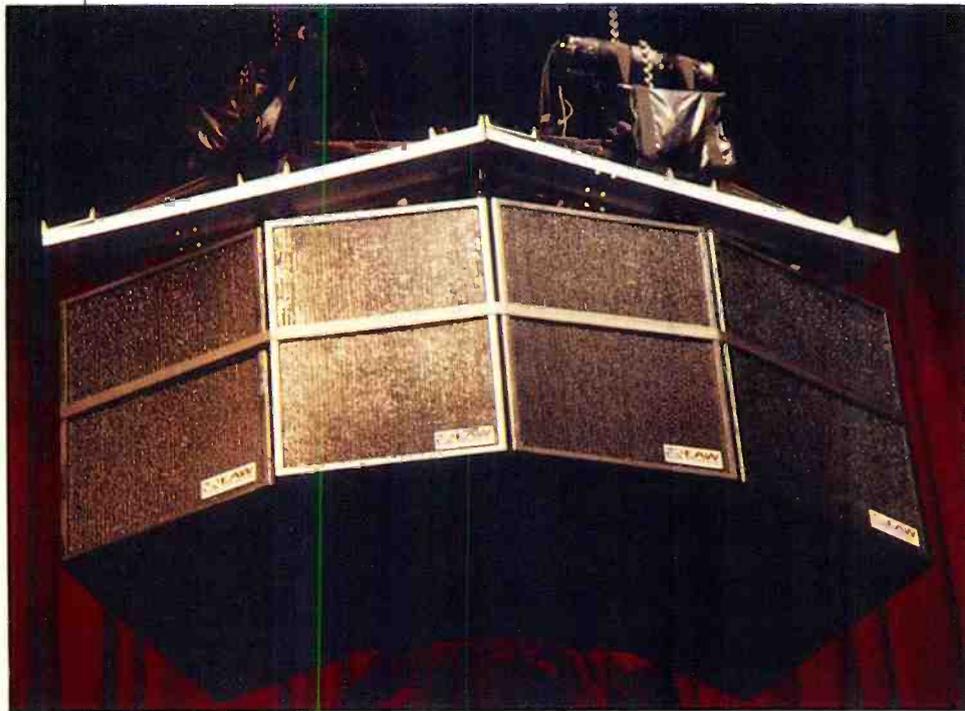


Photo 6. EAW's KF850 system, shown here in a central point-source array at Radio City Music Hall in New York.

some of these controllers. Then let's examine some of the more commonly used systems, along with a few new introductions.

The crossover/equalizer

As a system designer chooses the individual loudspeaker components, attention is paid to the crossover frequency, and the effect that a particular frequency choice will have on the power distribution to the various sections of the speaker system. A major purpose of the active filter is to confine the electrical energy that is sent to a given speaker to its own narrow operating bandwidth.

Most electronic controllers also incorporate specially tailored EQ "shaping," including narrowband filtering, to match the control circuitry to the specific speakers in use, and to offer an average frequency response that is nominally "flat." Narrowband filtering should be used carefully in such situations, however, since any filter/equalizer operating in the analog electrical domain can introduce aberrations to system phase response and transient-signal integrity.²

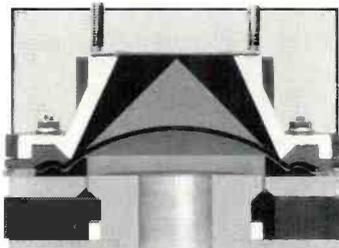
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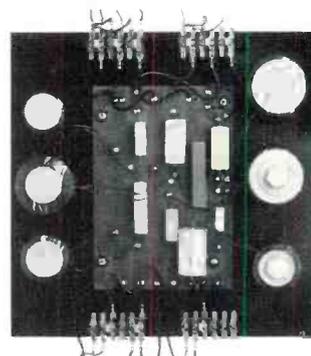


CVX/253

Cerwin-Vega! introduces the CVX Series. Using results of highly advanced measurement techniques, this line incorporates the configurations and componentry most requested by sound professionals.



The M-161 is a 150W (EIA) 2" throat compression driver with a 4.5" nodally driven phenolic dome. The unit features a moderate (4:1) transformation ratio, stepped pole piece and front plate, and aluminum flux stabilizing ring. This design results in high efficiency, wide bandwidth and 20dB lower distortion than any comparable 2" throat driver.



CVX employs our two most field-proven compression drivers, the M-161 and TI-1. Cerwin-Vega's high performance woofers complete the systems, utilizing the same midrange and high-frequency drivers. Each model varies only in maximum acoustic output.

Voice fundamentals and harmonics are seamlessly reproduced by the M-161 without crossovers in its bandpass.

crossover point selection and slope are critical. For this reason, some system designers choose to work with variable, or floating, crossover points. This type of processor allows a system to get louder by shifting a greater portion of the power response away from fragile compression drivers and down to more-rugged cone loudspeakers. This spectral shift can also affect both the system's coverage-angle characteristics and aural fidelity. For this reason, it may not be desirable in some situations. This is particularly critical in systems that use combinations of multiple loudspeakers, where the dispersion characteristics of the loudspeaker combination is different from that of the individual units.³ (See Photo 3.)

Level-sensitive, active equalization can be used to compensate for this phenomenon. It can also be used to take advantage of the Fletcher-Munson curves, based on the average human ear's sensitivity to particular frequencies, to provide more impressive bass and treble output when the system is working at lower levels. Thus, some listeners judge the system to be more "hi-fi" (or perhaps "disco?") even when it is receiving a very low input signal. This form of equalization

functions like the loudness contour button on a home stereo pre-amplifier.

Also incorporated in the crossover/equalizer section of some controllers is signal delay. (We can't delay time, but we can delay the audio signal.) High-level multiway systems often use signal delay to improve the system's coverage pattern at or near the crossover point by providing electronic compensation for driver phasing aberrations or mounting location. (See Photo 3.)

Loudspeaker protection circuitry

Different methods have been developed to protect loudspeaker components as packaged systems have become more compact. Loudspeakers fail either from thermal damage caused by excessive voltage input or from over-exursion, which can lead to rubbing and voice coil misalignment. Protection methods are based on sensing impending voltage overload or modeling voice coil temperature from the measured impedance readings of the speaker.

A single loud pulse, or transient, can tear a speaker apart, leaving us with mechanical failure. Heat damage tends to occur when a high average power level

is maintained at the mid to higher frequencies of the speakers' rated operating range.⁴

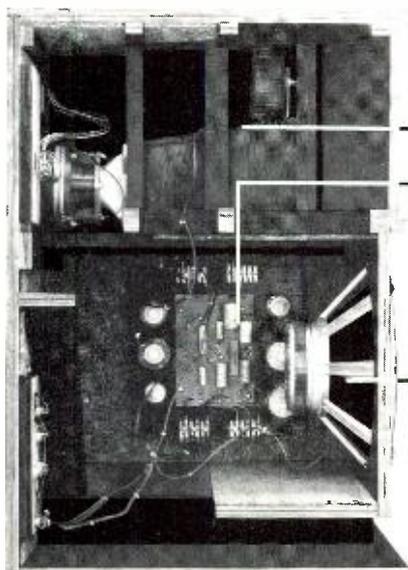
Controllers can use electronic circuitry to limit input voltage by gain reduction. To avoid voice coil failure caused by thermal overload, compression can be used to prevent the temperature from rising too quickly in the component. Some controllers use "sensor" circuits to monitor the power amplifier's output voltage, and to limit the signal going to the particular loudspeaker components that are beginning to show signs of thermal stress. (See Photo 5.)

Frequency-dependent limiting based on impedance-characteristic measurements is one of the more sophisticated methods used to guarantee the highest possible average sound levels without component failure. Both low-cut and high-cut filtering reduce the voltage input to the loudspeaker system. They also narrow the system's audible frequency bandwidth as the sound pressure level increases.

Most controllers rely on sensed information from the power amplifier's output terminals. Products from Apogee Sound, Celestion, Electro-Voice, Meyer Sound Laboratories, PAS (Professional Audio

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1" throat aluminum dome, titanium-former high frequency compression driver.

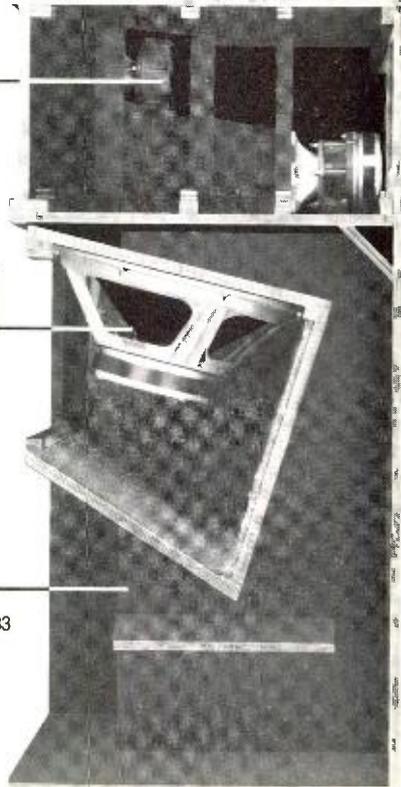
Mid/high frequency unit used in all models adds to structural integrity of enclosure.

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Photo 7. E/V's new Delto-Max two-way bi-amped speaker systems come in 12-inch and 15-inch versions. Dedicated controllers are available, and the manufacturer discourages its use with other crossovers and/or controllers.

Photo 8. The new KP-600 series system from Klipsch features modular loudspeaker enclosures with identical exterior cabinet measurements. The tri-amped four-way system has a controller with user-selectable crossover parameters.



Photo 10. A pair of Meyer UPA-1A enclosures are easily suspended from simple hanging hardware in a low-ceiling ballroom. The UPA-1A is typical of the single 12-inch box with horn/driver components in a trapezoidal package, available from several manufacturers.

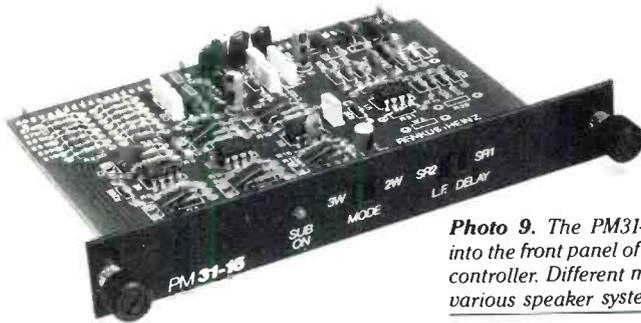


Photo 9. The PM31-15 circuit module plugs into the front panel of the Renkus-Heinz system controller. Different modules are available for various speaker systems.

Table 1. Size, weight and cost comparison of four compact trapezoidal speaker systems, each with one 12-inch speaker and a compression driver with horn.

	Apogee Sound AE-5	Electro-Voice DML-1122	Meyer Sound UPA-1A	Renkus-Heinz SR-121A
weight (pounds)	78	68	66	52
size (HxWxD) (inches)	23x14x14	23x14.6x14	22.5x14.5x14.5	24x15.5x14
peak output (decibels)*	130	129	132	132
retail price, single box (dollars)	2,045 (with fittings)	1,860 (add 348 for fittings)	2,390 (add 100 for fittings)	1,530 (add 296 for fittings)
retail price, controller (dollars)	1,850 (A-5)	1,632 (DMC 1122)	1,312 (M-1A)	1,650 (X22, add 250 for module)
total (dollars)	3,895	3,840	3,802	3,726

Notes: *The peak output is taken from manufacturers' specifications sheets. All figures are reportedly measured at a distance of 1 meter on axis. An independent verification under controlled conditions is recommended.

The total price for a single compact enclosure and necessary controller includes rigging fittings, top and bottom, which are optional on three of the examples. The Renkus-Heinz X22 controller requires the additional plug-in module for operation. Thus, the total cost in all four examples is for a single enclosure with rigging fitting and a functional controller.

The four examples are listed alphabetically by manufacturer. The fact that the total list prices are in descending order is coincidental. The model of the appropriate controller is in parentheses in the controller retail price column.

Systems), and Renkus-Heinz all include back-panel sensing-circuit line connectors, providing an information loop that enables the controller to function automatically in response to the amplifier/speaker system. Most units use both a low-frequency and high-frequency sensing signal path; the PAS unit senses high-frequency amplifier output level only. (See Figure 1.)

"We are not really big on limiters, because we feel that's a large part of what users refer to when they object to a 'processed' sound," explained John Kirkland, product manager for PAS. "However, when the limiter's input is tied directly to the output of the high-frequency power amp, you only get limiting when the drivers are in danger of being harmed. In effect, the limiter tracks the output of the HF power amplifier, using opto-couplers. So, the protect function is only engaged when high levels of certain types of program material are encountered."

In most protection systems, the goal is to maximize the output of the speaker components without destroying them. But, there is usually some sacrifice in signal integrity. Several different methods can be used to increase the average attainable sound pressure level of a speaker system.⁵

Controllers may use one or more of the following functions searching for the ideal combination of equalization and speaker protection:

- Loudness Compensation**—Low (and sometimes high) frequencies are boosted when the system is operated at lower levels, and the effect is removed as the level is increased.

- Variable High-Pass Filtering**—While often audible as a loss of bass response when the program material changes, this function will reduce both thermal and mechanical stress.

- Variable Crossover Frequency**—The floating crossover point can protect high-frequency components from damage at

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Photo 11. After proving that electronically controlled speaker systems could be accepted in the marketplace, system manufacturers worked on development projects for larger, arrayable components for sound reinforcement use. Shown here, the first outdoor field test of the Renkus-Heinz MR-1/LR-2M system (on stage at Irvine Meadows Amphitheater in Southern California).

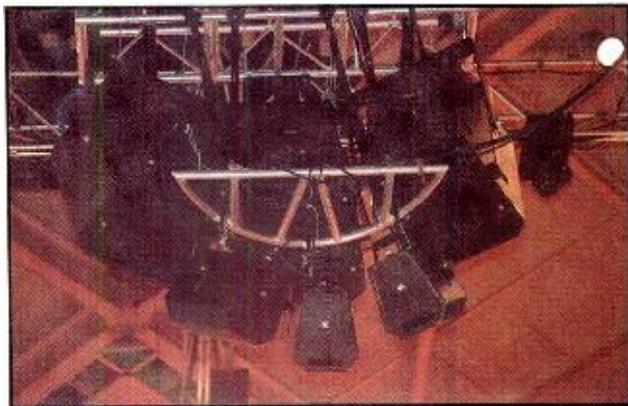


Photo 12. Manufacturers intent on producing large-scale systems for the professional market must offer enclosures that are easy to use in hanging situations. Shown here, Apogee Sound's 3X3 and AE-5 enclosures in an array by Burns Audio at the 1988 Democratic National Convention.

While electronically controlled speaker systems have been proliferating in recent years, some companies have decided not to jump aboard this trend too quickly. JBL Professional, for example, has offered a card-programmable electronic crossover since 1973. Models 5231 and 5232, introduced in 1973, evolved into the present model 5235, which features equalization for specific systems, including choice of crossover slope, power response correction and selectable high-pass filtering for low-frequency driver protection. Switchable monaural summing for subwoofers can be internally preset, along with

selectable equalization for low-frequency alignments. These features allow the crossover to operate in a fashion similar to more expensive "electronic controllers."

To this manufacturer, "smart" electronics means something entirely different from what has been brought to market so far. JBL system designers maintain that they would never use electronics as a substitute for a larger power class in transducer selection, or use them in a manner that grossly altered the artistic intent of the performing musicians. The following statements from JBL establish that company's position (page 34).

—David Scheirman

high operating levels, but off-axis frequency response can vary considerably.

- Soft Clipping**—Gently attenuates transient peaks, and allows protection from over-exursion without dropping the overall gain.

- Compression and Limiting**—Overall gain reduction is an effective way to prevent thermal overload that leads to speaker failure. Variable ratio compression circuits can work well. High compression ratios in limiting circuits can be very audible.

Controller comparisons

In searching for ways to evaluate one system controller in relation to another, the most obvious direction to take first is what each sounds like. If we have the luxury of auditioning several different systems in the same listening environment, with the same program material, it is relatively easy to develop a working knowledge of each system's characteristics.

Since this is often not possible, an awareness of the audible characteristics of the previously mentioned signal-processing methods can still give us valuable information. By recognizing the methods used in a particular controller, we can gain some general knowledge prior to hearing it in use.

- Apogee**—Apogee Sound's controller is a crossover/equalizer with fixed crossover points and corrective equalization. Apogee has labeled its sensing circuitry PAF (positive amplifier feedback). A limiter circuit is used for driver protection.

The models are keyed to different loudspeaker systems (such as the 3X3, the AE-1, the AE-5 and the AE-10 subwoofer). RV designates road version for touring; PV designates permanent version for installations. Thus, a model A-1 RV would be dedicated for use with the compact AE-1 enclosure, with input and output connectors on the front panel for easy access during setup and testing.

- Celestion**—Celestion offers a crossover/equalizer, with a fixed-point 150Hz second-order filter for low-frequency (subwoofer) enclosure operation. Low-frequency protection is through compression, adjusted by power amplifier output monitoring. A three-stage thermal protection circuit for the extended-range speakers is based on voice-coil temperature modeling.

This device is named the SRC1 Electronic Controller. It features a back-panel bi-amp switch and phase reversal for the subwoofer section. It is designed for use with the SR speaker system. (See Photo 4.)

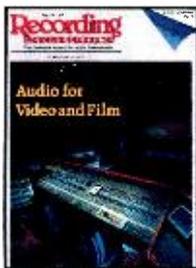
- Eastern Acoustic Works**—EAW offers a crossover/equalizer with signal delay and loudness compensation (low-

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Electronically Controlled Speaker Systems: A JBL Viewpoint

By John Eargle, Mark Gander and Garry Margolis

"Electronically controlled speaker systems" is one of the catch phrases in current pro sound. The term refers to electronics that are dedicated to a given loudspeaker and that process the sound for enhanced presentation over that loudspeaker.

One of the stated premises of such systems is that they are basically pre-adjusted and ready to work, with little in the way of user adjustment.

Often, the underlying premise is that these systems have marginal transducers, and the real purpose of the "smart" electronics is to prevent

overdriving or stressing those components.

Another premise is that the packaged speaker-plus-electronics should sound good at moderate to low levels when auditioned on the sales floor, since that is where sales are generally made. Therefore, a little loudness compensation (bass boost) may help make the system stand up better in competitive comparisons. Also, some degree of power response correction in the compression driver circuitry will help the overall balance.

As we see it, the big problem with these systems is that most of the correction, beneficial as it may be, is dynamic in nature, and it is gradually removed as the system is

driven at higher and higher output levels.

Examining one model

We have measured a number of such systems to determine how they typically operate. Figure 1 shows the electrical drive to a woofer in one such two-way system. The lowest curve in the graph shows a mild bass boost around 40Hz, providing some degree of loudness compensation for moderate acoustical levels. The curves were all incremented upward in 1dB steps.

Note that the boost is quickly removed, and in its place, at an input level 16dB higher, there is actually a bass roll-off below 80Hz. Note further that the actual drive level at 40Hz never varies more than 2dB over the entire input range of 16dB. Note also that the bass low-pass crossover frequency moves from about 1kHz up to 1.6kHz over the total input range shown.

Figure 2 shows what happens in the same system at high frequencies. Beginning at the bottom curve, note the generous 9dB power response boost above 4kHz. However, over a total input range incremented upward 16dB, the power response boost is reduced to only 3.5dB. Furthermore, the high-pass crossover point moves from 1kHz at low levels to about 1.6kHz at the highest input level shown here.

Benefits and liabilities

The system we have described exhibits three major departures from normal signal linearity:

1. The bass content at 40Hz remains virtually the same (1dB) over a program input range of 16dB. Music thus loses some important dynamic information.

2. The acoustical crossover point shifts from about 1kHz at low input levels up to 1.6kHz at the highest input levels shown here. There will

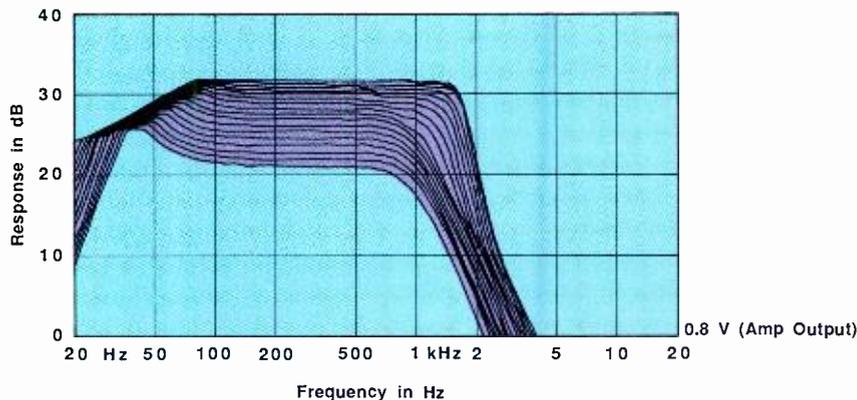


Figure 1. Low-frequency output of processor, 16dB input range.

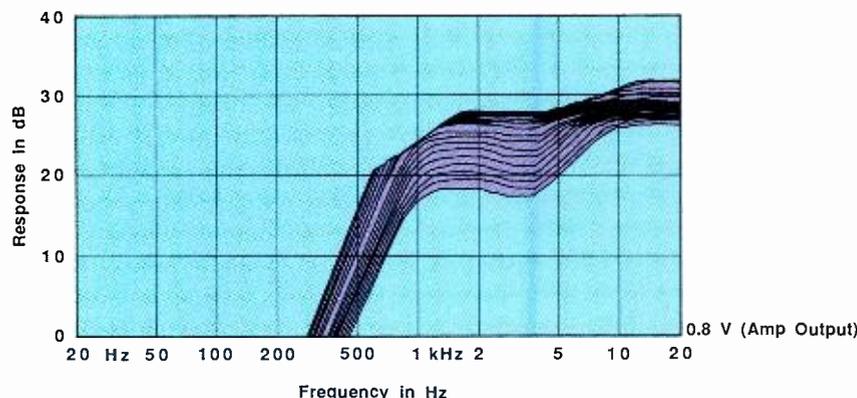


Figure 2. High-frequency output of processor, 16dB input range.



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be a change in system coloration and directional characteristics in the mid-band as the woofer is required to work at higher frequencies. In this case, the woofer is 380mm (15 inches) in diameter, and such drivers should normally not be crossed over higher than 1kHz because of the inevitable response and directional anomalies above 1kHz in drivers this size.

3. The overall change in high-frequency balance of about 5.5dB can rob music of brilliance as the program level goes from soft to loud.

In addition, there may be audible pumping in the dynamic control of the system as higher levels are reached.

On the positive side, such systems sound good at lower levels, and that is probably where they will be used much of the time. It is difficult for the system to go into gross distortion because of the overall compression activity in the processor. The systems are, in fact, relatively foolproof and quite easy for the novice user to set up. While some operators may prefer the relatively dense, compressed sound of the system, many musicians will not.

As an overall philosophy, if the system has inadequate high- and low-frequency drivers, the money spent on the processor might be better spent on more robust internal componentry. Transducers with smaller diaphragms and smaller voice coils than might otherwise be chosen will suffer from dynamic non-linearities such as increased harmonic and intermodulation distortion, and greater power compression. While electronics can try to prevent these overtaxed units from breaking and allow them to generate maximum output, the result is often not maximum fidelity.

frequency enhancement). The overload circuitry offers distortion protection through level-sensitive bandpass limiting. The unit is called the MX800 CCEP (closely coupled electronic processor).

A plug-in module allows the system to be used with the KF850 loudspeaker system, for which the MX800 is optimized. The KF850 system does not require this specific controller for operation. The manufacturer advises that electronic crossovers from other manufacturers can be used with excellent results. (See Photo 6.)

Electro-Voice—E/V's Delta-Max systems use a crossover/equalizer with signal delay circuitry. Crossover points are fixed, and high-pass filtering does not shift. Optimized equalization, including low-frequency enhancement (+8dB boost at 50Hz) is included. Speaker protection is achieved through an automatically ratio-variable compression circuit. Speaker modeling circuits provide thermal overload protection and amplifier anti-clip limiting. There is independent LF and HF soft-clipping for excursion protection.

Named the DMC-1122 and DMC-1152, the pair of controllers are intended for use with the respective 12-inch and 15-inch loaded Delta-Max systems. (See Photo 7.)

Klipsch—Klipsch has recently introduced a crossover/equalizer with user-selectable high-pass filtering in three bands. Automatic bandpass gain compensation is based on individual driver sensitivities. High-frequency equalization for flat response is incorporated.

Designated the KP-600-EC, the manufacturer calls this an electronic crossover/processor. Features include mute and phase reversal switches. It is designed for use with Klipsch's new KP-600 modular loudspeaker system. (See Photo 8.)

Meyer Sound Laboratories—Meyer Sound offers crossover/equalizers that use separate limiters for LF and HF outputs. The low-frequency component is protected by a variable high-pass filter. Meyer terms this circuitry Speaker Sense. Frequency and phase response alignment circuitry for specific loudspeaker systems is prealigned by the manufacturer.

Meyer calls its controllers "Control Electronics Units" and offers several versions, dependent on the loudspeaker system in use. These range from the P-1A (for the small UPM-1 column loudspeaker) to the M-1A (for UPA-1A and UM-1A UltraMonitor). The B-2A is designed for use with the USW-1 subwoofer. It features a Bass Extender circuit, offering transient-dependent "fuller bass" effects.

The M-3T incorporates logic circuits in the sense lines that monitor amplifier gain, providing automatic speaker output disconnect in case of overload or amplifier

failure. A time correction circuit is switchable in and out. This controller is for use with the MSL-3.

Professional Audio Systems—PAS features crossover/equalizers with EQ compensation for constant-directivity horn designs and a subsonic filter. TOC (time offset correction) is employed as signal delay.

High-frequency component protection is achieved through a fixed-ratio soft-knee-type limiter circuit. Sensing lines are connected from the HF amplifier output to the controller's limiter input.

The units are designed to be used with PAS's Modular Reinforcement Systems (MRS-1, MRS-2).

Renkus-Heinz—Renkus-Heinz's Smart Systems use crossover/equalizers with automatic loudness (bass-boost) compensation. This changes to bass-cut equalization at very high levels to maximize average sound pressure levels.

High-frequency protection relies on variable crossover frequency points and over-easy-type compressor circuitry. Low-frequency protection includes variable high-pass filters and overall level compression. The manufacturer calls this process SPT (spectrum power transfer).

The Model X-31 controller is adaptable to different speaker systems. It includes front-panel plug-in modules, so that a controller can be quickly changed for use with a compact two-way system to a large three-way system. Three different modules are available. The PM31-15 is for use with SR-1, W-1 and SR-2 speaker systems. (See Photo 9.)

Compact loudspeaker system packages

In examining the various controller/speaker systems that are available, accurate, fair A-B comparisons can be difficult since such a wide variety of loudspeaker enclosures is available for many different uses.

We can, however, focus on just one type of speaker enclosure and check out a few vital statistics. Let's look at four examples of a compact trapezoidal enclosure, each housing a single 12-inch speaker and a compression driver with horn.

This style of speaker system is extremely popular with a wide variety of users. It is compact enough to be placed on small stages and light enough to be tripod-mounted. It lends itself easily to use as a suspended speaker in low-ceiling meeting rooms and hotel ballrooms. Found in most rental sound company inventories today, this style of speaker is quite versatile. (See Photo 10.)

Since we'll need a controller even if we are only using a single speaker enclosure, we'll take a look at the current retail price

(as of October 1988) of one box and one controller. (See Table 1.)

Large-scale applications

With electronically controlled speaker systems firmly established in the smaller rental system market, manufacturers have begun to expand those technologies into larger system formats. Attempts have been made to develop integrated, full-scale speaker systems for use in bigger venues. (See Photo 11.)

Many of these same controllers are used with arena-sized, full-range systems for audio-visual, public address and live concert applications. To be successful in this part of the industry, speaker system manufacturers must offer products that are easy to use in arrays. (See Photo 12.)

Many companies include built-in hanging fittings and even loose rigging parts or flying bars with their product lines. Those manufacturers that are serious about the professional marketplace often work closely with touring sound firms in concert situations to develop products that are acceptable for this field, including high-powered subwoofer enclosures. (See Photo 13.)

A glimpse into the future

Power amplifiers can vary a great deal in their operating characteristics, and electronically controlled speaker systems can be very sensitive to the input signal as they work right up against their safe operating limits. Some speaker manufacturers prefer to see their customers use one design type, or specific brand of power amplifier with their controlled speaker systems. Other firms claim that any amplifier meeting the suggested power ratings will work fine.

Perhaps one of the next steps in the development of electronically controlled loudspeaker systems will be the integrated controller/amplifier/speaker system, in which the power amplifier is ideally matched to the signal processing and loudspeaker components. Perhaps the front-end circuitry will be included in plug-in modules that are easily inserted into power amplifier frames for use with different speaker systems. Several manufacturers are presently involved in development projects of this type.

Along this line, Yamaha Corporation of America has recently introduced the AST-P2602 power amplifier, which contains integral signal processing circuitry. A plug-in cartridge gives optimized equalization for the AST-S30 loudspeaker system, which comprises a trapezoidal enclosure housing a single 15-inch speaker, a compression/driver and horn mid-range section, and a ring-type compression tweeter. Yamaha says its Active Servo Technology

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allows the loudspeaker port to become an independently adjustable source of low-frequency energy, thus providing improved bass response from a compact loudspeaker system. The new AST amplifier does not require an extra sensing wire connection, however. (See Figure 2.) When the AST cartridge with its 800Hz crossover is removed, the AST-P2602 reverts to operation as a standard power amplifier.

We can also expect to see an ever-expanding range of tripods, clamps, flying fittings and portable accessories such as cable-sets to be developed and marketed by various speaker system

manufacturers in an attempt to become a one-stop supply source for professionals.

The value of dedicated signal processing in the form of electronic controllers will become more evident as technically advanced manufacturers begin to provide computer-interface data ports, coupling their signal processing to a central control microprocessor with VCA circuits. (A system manufactured in the Netherlands by Stage Accompany already offers two-way, full-range speaker enclosures with built-in microprocessor control and the ability to link these boxes to a master control computer.)

One day soon, perhaps even the

simplest setups will require more technical expertise to hook up—but once the system is wired and in place, many more operational variables will be in the hands of the sound system, not the human operator.

How does all of this affect the major touring sound industry? A few of the bigger concert-system rental companies stock large inventories of electronically controlled speaker systems, and these are in use with live musical concerts as much as they are for industrial and trade show events. Other firms refuse to bring this type of system into their operations. Most of the larger firms still field proprietary speaker systems.

Traditional electronic-crossover, multiway systems (with optimized equalization and bandpass limiting) will continue to be employed in applications where complete control of the operating parameters must be kept in the hands of musically oriented technicians at the mixing position. This is evidenced by most major touring concert systems in use today.

However, electronically controlled speaker systems have definitely found their niche in other parts of the professional sound reinforcement industry. Understanding the signal processing functions available within specific controllers is a must for any user needing to make certain that a given system will be appropriate for use in different applications.

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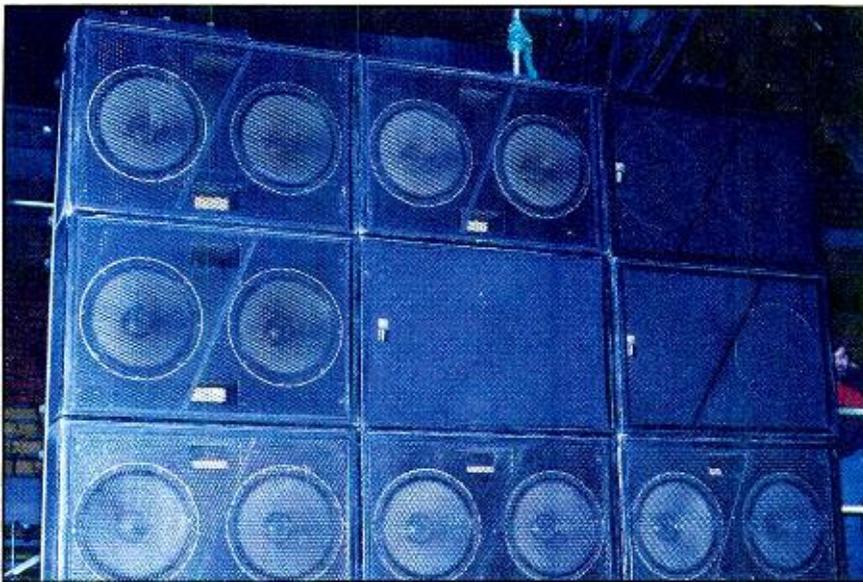


Photo 13. Large-scale concert use requires speaker system components that can fill huge performance spaces. Shown here, Meyer Sound Laboratories' 650-R2 subwoofers as fielded by Ultra-Sound at Meadowlands Arena in New Jersey.

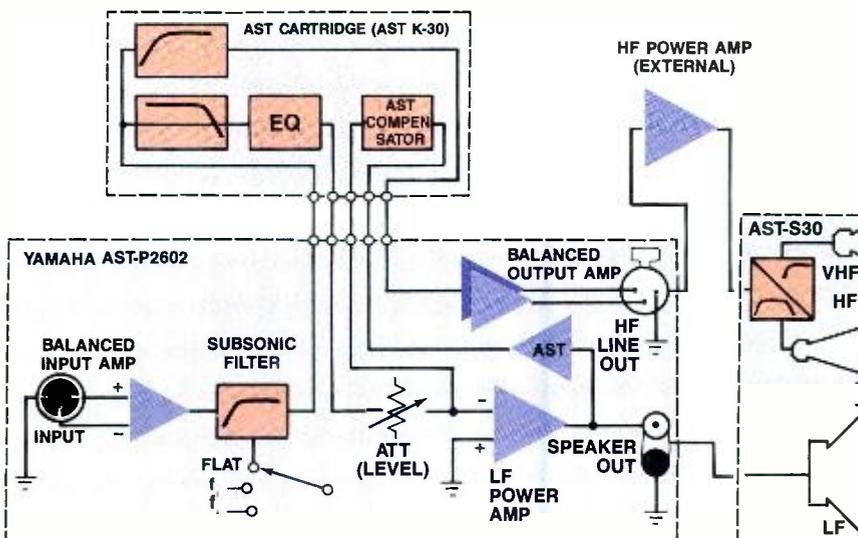


Figure 2. Detailed block diagram of an AST amplifier/loudspeaker system.

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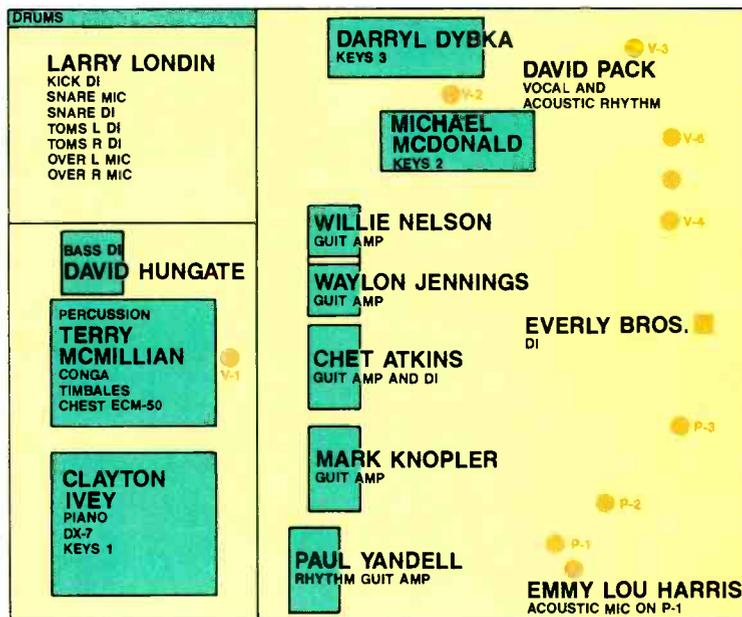
⁴Gary Davis & Associates, "White Paper: A Comparison of Several Manufacturers' Speaker Components for Sound Reinforcement Loudspeaker Systems," (Yamaha Corp., 1981).

⁵Michael Miles, "Electronic Control of Loudspeaker Systems," (Presented Paper, All Technical Engineered Sound Conference, 1988).

Vanderbilt University in Nashville. This small, black-box, multiform, theater was perfect. The acoustics were pretty dead and the audience could be placed anywhere. No seat was more than 30 feet from the stage. That meant the sound reinforcement system would not have to be large and that the sound of the instruments from the stage would seem a little more natural. The importance of the house and monitor mixer was paramount. The mobile audio facility was Fanta Professional Services, and the video facility was The Nashville Network, Unit 1. The main audio crew consisted of Mark Repp, Johnny Rosen, David Palmer, Robin Victor, Shipley Landis, Jamie Shankland and Billy Saurel. Here is how the jobs broke out.

Mark Repp: In charge of the audio inside the Nashville Network Unit 1 truck and the recording of all audience and effects microphones. Remember that the audience was very visible in this show, and their reactions were important to the overall feel. The live stereo mix had to be available for production personnel and had to be shipped to all the recorders.

All video machines had three audio tracks; the same mix was not sent to every



P-# = PRODUCTION MIC FOR DIALOGUE AND SINGING (USUALLY PLACED UP FRONT)

V-# = VOCAL SINGING MIC

BV-# = BACK-UP VOCAL

KEYS-# = KEYBOARD (USUALLY A DI)

A-# = AMP ON STAGE

PERC-# = PERCUSSION

DI-# = UTILITY DI (USUALLY MOVES)

Figure 1. Generic stage plot.

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Circle (22) on Rapid Facts Card

REC	PA	SNAKE	INPUT DESCRIPTION		SUB MSTR	TYPE
1	1		KICK	LARRY LONDIN		DI
2			SNARE	SUB-MIX C12-1		
3			LEFT DRUMS	SUB-MIX C12-2		
4			RIGHT DRUMS	SUB-MIX C12-3		
5	5		BASS	DAVID HUNGATE		DI
6			PERCUSSION	SUB-MIX C12-4		421
7	6		RHYTHM GUIT AMP	PAUL YANDELL		421
8	7		GUIT AMP	MARK KNOPLER		421
9	8		GUIT DIRECT	CHET ATKINS		DI
10	9		GUIT AMP	CHET ATKINS		421
11	10		V-1	TERRY MCMILLIAN		SM-58
12	11		V-2	M. MCDONALD		SM-58
13	12		V-3	DAVID PACK		SM-58
14	13		V-4	BACKGROUND VOCALS		SM-58
15	14		V-5	BACKGROUND VOCALS		SM-58
16	15		V-6	BACKGROUND VOCALS		SM-58
17	16		P-1	PRODUCTION 1		SM-58
18	17		P-2	PRODUCTION 2		SM-58
19	18		P-3	PRODUCTION 3		SM-58
20	19		ACOUSTIC RHYTHM	DAVID PACK		ECM-50
21	20		KEYS 1	CLAYTON IVEY DX-7		DI
22	21		KEYS 2	M. MCDONALD		DI
23	22		KEYS 3	DARRYL DYBKA		DI
24	23		ACOUSTIC MIC ON P-1			
25	24		GUIT AMP	WAYLON JENNINGS		421
26	25		GUIT AMP	WILLIE NELSON		421
27	26		ACOUSTIC GUIT	EVERLY		DI
28	27		PIANO MIC	CLAYTON IVEY		
29						
30	28		SPARE MIC	CAMERA LEFT		SM-57
31	29		SPARE MIC	CAMERA RIGHT		SM-57
32			tone generator			
33	2		SNARE MIC		421	C12-1
34	3		SNARE DI			C12-1
35			TOMS LEFT DI			C12-2/3
36			TOMS RIGHT DI			C12-2/3
37			OVERHEAD LEFT	SCHOEPS CAROIDID		C12-2/3
38			OVERHEAD RIGHT	SCHOEPS CAROIDID		C12-2/3
39			CONGA		421	C12-4
40			TIMBALES		421	C12-4
41			CHEST		ECM-50	C12-4
RETURN LINES TO THE STAGE						
#	PA	SNAKE				LEVEL
44	4		DRUM MIX FROM FANTA TO PA			+4
52			FANTA TALKBACK TO SPKR-AMP			

TIE LINES TO REAR PATCH PANEL								
JACK	SNAKE	JACK	SNAKE	JACK	SNAKE			
1	1	LEFT MAIN	3	3	LEFT SPARE	5	5	L. RETURN
2	2	RIGHT MAIN	4	4	RIGHT SPARE	6	6	R. RETURN

Figure 2. Track layout sheet.

machine. Some got stereo, some got just the audience or just the wireless lavalier on Chet. Mark also had to place and record all the audience microphones used for the original stereo mix onto the videotapes. A 24-track recorder was used, and each of the 18 microphones was recorded on a separate track for later use in post-production sweetening.

Johnny Rosen: The audio engineer in charge (A-1). Usually an A-1 develops the

audio plan, mixes the show and everyone follows along. The crew on this show didn't require that much supervision. Instead, the Fanta office became the communications hub for all the audio people. Having a single person for all other artistic and technical teams to communicate with simplified the process and avoided redundant and frustrating meetings. Johnny mixed the live recording, which was recorded on two analog 24-tracks, and

TRACKS

1	2
KICK	SNARE
3 LEFT DRUMS	4 RIGHT DRUMS
5 BASS	6 PERCUSSION
7 PAUL RHYTHM	8 KNOPLER GUIT
9 CHET DI	10 CHET AMP
11 V-1	12 V-2
13 V-3	14 V-4
15 V-5	16 V-6
17 P-1	18 P-2
19 P-3	20 RHYTHM DAVID
21 KEYS-1 & PIANO	22 KEYS-2 MCDONALD
23 KEYS-3 DARRYL	24 TIME CODE
25	26
27	28
29	30
31	32
FANTA PROFESSIONAL SERVICES 1213 18TH AVE. SOUTH NASHVILLE, TN 37212 615-327-1731 JOHNNY ROSEN	
PRODUCER	SUSAN HACKNEY
CINEMAX/HBO	
CHET ATKINS +	
5/2/87	# F#5308
MIXER	JOHNNY ROSEN
STAGE	BILLY SAUREL
A-25	TAPES ROBIN VICTOR
TAPE TYPE AND LEVEL	456 250nWb +3
NR	# OF REELS
AS OF DATE AND TIME BY 6/1/87 7:47 AM	

sent a simultaneous stereo mix to the video truck for distribution to all the various video machines. That ranged from 1-inch C format to 8mm home video format.

David Palmer: David had previously mixed an Earl Klugh-Chet Atkins album that was very successful. After listening to the album, we were sure that he would be an asset, but didn't know how to use him. Though he was not a regular with the Nashville crew, he was willing to fit in and find a special niche that wasn't being covered by other members of the audio team.

When he arrived in Nashville, he started by hanging out with the musicians. He had worked with several of them but never all of them as one group. Starting with the rehearsals at Studio Instrument Rentals, Nashville, David made elaborate notes about every song, from both the technical and artistic angles. Eventually these notes would help the rest of the audio crew the time to concentrate on their individual chores for the show.

The job that David finally filled was remixing the show for the final post-production audio mix. He had followed the music from the beginning of the rehearsals through to the live recording. Because he did not have to mix the live recording, he could concentrate on the artistic, rather than technical, qualities, which allowed him to emulate the live feel better in the final remix. Sometimes, when you are tracking a live show, it is very difficult to listen for the musicality of the songs or the overall energy in the performance. David could concentrate on the music while Johnny made sure that the tracks and the live mix were cut correctly.

Robin Victor: Coordinated all of the audio people and made sure that all of the logistical efforts flowed in the same direction. She kept track of all the audio team members by phone, pager, mobile phone and walkie-talkie. She documented every audio change in the production. As each deviation from the original plan occurred, she evaluated who needed to know and what people were affected.

The notes that Johnny, David and Robin made were all compiled into an Audio-Master notebook. Various parts of the notebook would later be distributed to the crew for use during the show and in post-production. If a problem affected other parts of the crew, Johnny would work out the details with each group. That way, other production members only had to deal with one person if problems arose. We wanted to avoid multiple answers to the same questions.

As the show was being rehearsed, Robin called cues to the audio team via a headset intercom system. Usually changes during

the show happened out on the stage and were supervised by the audio stage manager. All changes were logged in her book. Having a person at the location, dedicated to handling communications and documentation proved essential. Working from her updated Audio-Master notebook, she called the audio cues during the live recording. The version of the Audio-Master that was used on the set was called "a short run-down."

Billy Saurel: As audio stage manager, he interfaced directly with the talent, sound reinforcement team, TV stage manager, stage hands and camera team. He handled all the microphone placement and audio-cable patching in the stage area. A major part of his job was to communicate the needs of the musicians to the monitor mixer. The audio stage manager needs to understand exactly how the multitrack audio and video recording is done. Sometimes he has to make fast changes in the plan; unless he understands every monitor mix and audio track, his decisions could prove disastrous. Improper tracking makes remixing nearly impossible, improper microphone technique yields horrible quality, and choosing an improper input could make mixing the monitors very difficult. Basically, if the talent doesn't have to think about the audio system, the overall quality of the show will be much greater.

Jamie Shankland: Jamie's job was to supply and mix the stage monitors and the house system. He was picked because of his experience doing TV shows and his ability to cope with an unusual mixing environment. His monitor system was small, loud and of high quality. He would listen to requests carefully and only act when necessary. Before he made any changes, he checked in with Johnny or Robin so that everybody was aware of any critical changes.

ShIPLEY Landis: Past experience has shown that on every complex job, it is essential to have a utility person who can fill in for any of the other workers. His skills need to include mixing, production knowledge, electronics, audio/video interfacing and, especially, the ability to think fast and maintain a sense of humor while under fire. While interfacing three different communication systems, Shipley had to spot problems and alert the correct people. This job requires a jack of all trades, a master of all trades and a candidate for the "Henry Kissinger" diplomat award. We follow the philosophy that all of the equipment will break at a critical moment, and when that moment arrives, we need to know how to get around the problem rather than through it.

Equipment setup

The audio technical setup consisted of three main parts: the sound reinforcement system, the audio recording system and the audio tracks on the video recorders. The only part of the video system that matters here is how the sound was shipped to the video truck for its recorders and monitors.

All music and dialogue mixing was done in the Fanta truck and sent to the TV truck via a single audio snake. Coming back

from the Nashville Network truck was a stereo signal that could come from many sources in the Nashville Network truck. The controller for that crossbar switcher, that fed audio and video back to the audio truck, was located in the Fanta truck and the signal that controlled it traveled on a single coax cable into the Nashville Network truck. The following were the channel assignments for the Fanta to TNN Snake A:

Line 1—left stereo mix

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Circle (23) on Rapid Facts Card

January 1989 *Recording Engineer/Producer* 43

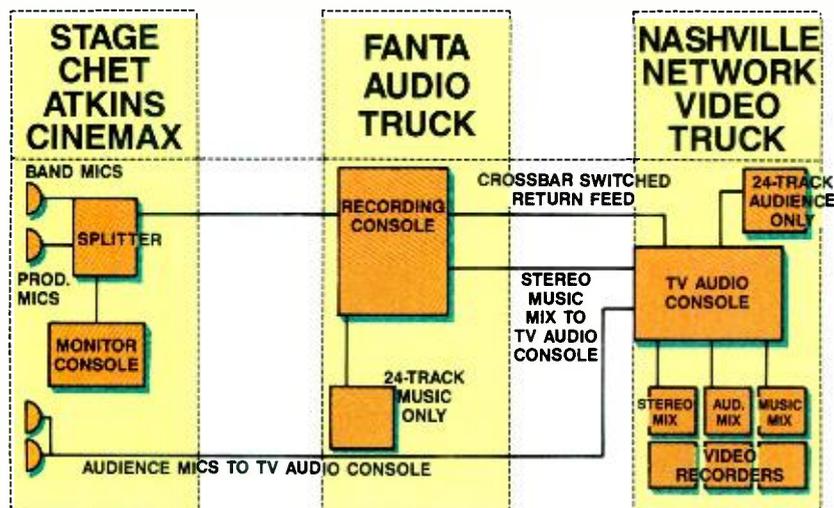


Figure 3. Audio path for live recording.

Line 2—right stereo mix

Line 3—left back up

Line 4—right back up

Line 5—TNN RTS intercom channel 1 = director, channel 2 = audio

Line 6—spare

Channel assignments for TNN to Fanta snake B (made of individual cables) were:

Line 1—left return line

Line 2—right return line

Line 3—spare

Line 4—output from return line crossbar switch controlled in Fanta truck (BNC coax)

Line 5—crossbar switcher control signal (BNC coax)

Line 6—SMPTE timecode

The stereo mix from audio truck was sent to the TV truck where it was distributed, along with the audience mix, to the various audio tracks of the video recorders. Obviously, we had to rely on Mark in the TNN truck to sweeten the live mix flawlessly because it really affected the way we cut the 24-track tapes. In this case, sweetening means the adding of audience and effects sounds to the stereo mix. We could hear the dry mix without audience by listening to the feed that left the audio truck, the sweetened mix by listening to the return feed from the TNN truck, or we could listen to just the audience mix by selecting the correct signal on the crossbar switcher. The advantage of doing it this way is that we had 23 tracks for audience alone that were recorded on a third 24-track and a mixer who could babysit them. Also, Mark was the engineer who would later have to sweeten the show when the final remixed tracks were laid back to the final edited video tapes.

Splitting the signal between the PA and

the audio truck was accomplished with our 54-input × 108-output splitter box. We use Jensen transformers with three windings, one microphone input and two microphone level outputs. It is built into a steel Haliburton suitcase. Individual grounding switches are available for the recording and PA feeds. 48Vdc phantom power is always "on" for each input. There is also a master ground switch that can lift the entire splitter system from the truck's audio ground.

Microphone technique

Good microphone technique is the most important part of multi-track recording. Microphone technique for recording and PA is almost always slightly different. For this particular show, we leaned toward the needs of the recording, but usually the PA company has the most say when it comes to selecting and positioning microphones. The attitude of the audio crew was that we needed to cater to the talent. If we did that, all the other pieces would fall in line. It turned out that there were only a few special or unusual problems.

The microphone and monitor system for drums was a hybrid of Larry Londin's equipment and the regular system. Larry's drum kit can be configured in many ways. He used D-drums for kick and toms while using his Drum Workshop 7"×15" snare to trigger an electronic snare, as well as other sounds. All this is mixed through a 8×4×2 Soundcraft mixer. He sent us, via Countryman direct boxes, a stereo drum mix minus kick and snare, a separate kick mix and a separate snare mix.

The cymbals were miced using two Schoeps SKM-541 cardioids overhead. We rarely use large diaphragm condensers in live recording because we get better phase

response with the smaller diaphragms. Larry mixed his own headphones and did not want a big loudspeaker for his drum monitor. He takes a special mix-minus from the PA system that has everything, including talkback from the audio and video trucks, minus his own drums. That special mix-minus from the PA was brought into an input of his mixer where he could balance his drums and a click track.

We were lucky that he would do it this way. Because there was no monitor speaker near him, we could vary the level of the over-heads without changing the apparent stereo perspective. It also helped us on snare because we could use a Sennheiser 421 instead of a Shure SM-58. The 421 is not as directional as the SM-58, and we could get by without a hi-hat mic. He played a lot of very soft cymbal parts and used his brushes often—the choice of microphones really made a difference.

Terry McMillian, the percussionist, is an enthusiastic and animated player. It was apparent from the rehearsal notes that he plays lots of different instruments with lots of different physical contortions. His congas and timbales were covered by 421s. All his other hand percussion toys were covered by an ECM-50 that he wore on his shirt. Having him wear the microphone provided us with one in-phase input that was always in the correct place. Terry understood how this worked and used good playing technique for the microphone he was wearing. His harmonica and vocals were handled with one SM-58 on a boom. All the percussion microphones were mixed to one track. The vocal microphone was assigned to a separate track. Again, the rehearsal notes came in handy.

Limiters and compressors

Limiters and compressors can ruin a live recording fast. We were careful not to do anything in the live recording that would prevent us from remixing and sweetening the audio. All the main production microphones were recorded through ADR Vocal Stressors. They have been our standard lead-vocal processor for years. This was necessary to avoid harshness from the monitors and provide good presence in the TV mix. Using the vocal stressor as a frequency-dependent limiter makes it appear to roll off the high end when it has to limit a lot. The monitors usually get harsh during loud passages; the vocal stressor makes that less apparent. All other vocals were compressed with dbx-160 units set at just under 1.5:1 ratio and almost no meter movement.

The bass was not limited in the recording chain. David Hungate has an elaborate rig, and we figured that he would set it up

better than we could.

All the electronic keyboards were peak-limited with a Valley International Dynamites just to protect against runaway short-term levels. The acoustic piano mic was limited with a UREI 1176-LN set at 4:1 with the meter swinging 6dB to 12dB. The meters are very slow, so we weren't real sure how much limiting that was. If it sounded OK, we left it alone.

The kick DI was also limited using a 1176-LN. Larry often plays very fast parts on kick drum, so it was set for peak limiting with the fastest release time. It helped contain the beginning of a note without losing its body or the next note. The left and right toms from the on stage drum mixer were compressed with dbx-160s set on 6:1, with about 3dB to 5dB of meter movement.

Effects

The effects used during the live recording were very simple: two Yamaha SPX-901s, one for vocals, set on preset program 1, and the other for drums, set on a preset random gate. An Eventide Harmonizer was used to get a delay of 30ms and 70ms from a mono electric guitar signal, just to open up the sound.

After the audience and effects microphones were added in the TV truck, the overall mix was run through an Aphex Compeller to smooth out the mix and make it easier for the editors. This allowed them to hear the soft passages more easily without having to deal with distorted loud passages.

The live mix

The mix of the live recording was the first thing that anyone would hear in post-production. If it was bad, it would affect the judgment of the editors. There was no time to remix before the editing began. After the show was edited, the new, remixed sound was added to the final video tape. Smoothness and simplicity got us through the original live mix. To avoid burning out the musicians, we decided to forego a long sound check. There was enough time during camera blocking to hear everybody play.

Each musician had played on stage many times, but they had never played together as a complete band. To encourage that mood, we felt strongly that all technical services should be as invisible as possible. The stage monitor system had to be physically unobtrusive, loud and very full-fidelity. The monitor mixing system would also have to supply the house sound. That worked fine in this case because the stage was at least as big as the audience area. (See Figure 1.) Finding a place to put the large monitor console, amps and peripheral equipment was hard

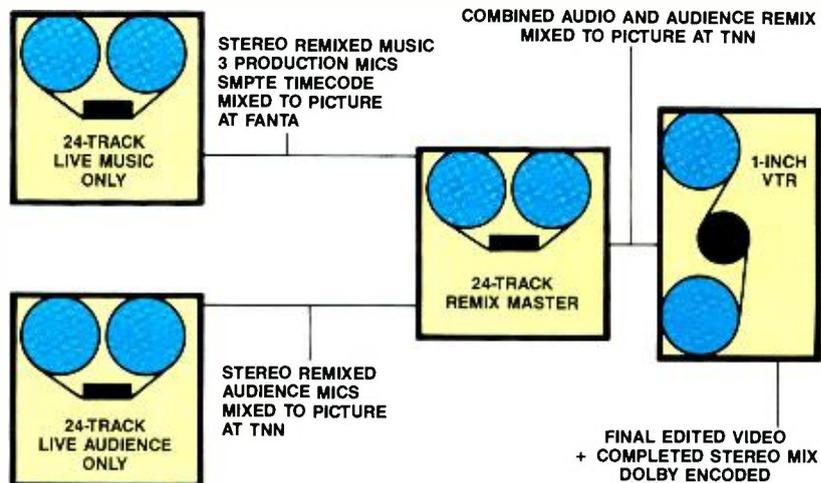


Figure 4. Remix audio path.

because we didn't want to see the engineer and his setup on camera, yet he had to be able to hear both the house and the monitor mixes. He got stashed at the edge of the stage area with a direct view of everything.

He was a little too far away for quick access to the microphones, so that put extra pressure on the audio stage manager. We tried to keep motion around the stage and chatter on the intercom minimal. We were trying to use the KISS method, which stands for "keep it simple, stupid." We never identified microphones by their input channel numbers. This avoided any confusion about what we were working on because the monitor and recording inputs were different. On the track layout sheet (see Figure 2), notice that Chet's guitar amp mic is input #10 in the recording system and #9 in the monitor system. By just calling it "Guit Amp Chet," we all knew what we had to find. We used Fanta's layout sheets for everyone because it has spaces for both system inputs and the on-stage sub-snake assignments. If the PA people understood what we were doing, we could all work together better. A group of us that have worked together over the years have settled into this system of labeling because it is simple and functional.

The site

The show was taped about one week before graduation, at Vanderbilt University, which caused some special logistical problems. The university is particularly proud of its grounds and buildings. The theater we were using is in the middle of the campus, surrounded by large magnolia trees and manicured lawns. The problem was how to pull the audio cables into the building without damaging the landscaping. The audio cable run was about 500 feet and needed to cross walkways, flower beds and lawns. After considering our

various options, we finally decided to run a guy wire through the trees that the cables could be attached to.

The power capacity of the building was quickly used up by the band's equipment, the lights and sound system. The audio truck, the main video truck and the extra tape machine truck were powered from a silenced, portable generator supplied by Citation Film/Tape Support. The unit had an operator with it whenever it was on and was available to us 24 hours a day if necessary. It had a crystal-controlled governor and very high current capacity, so the small changes in the load didn't affect the voltage or frequency. The only problem that we might have run into was a possible ground difference between the generator-powered equipment and the house-powered equipment. Fortunately, all the power and technical systems were wired correctly, so we had no problems. We attributed the lack of interfacing problems to detailed pre-production work and good communication.

Remixing this show was different from most TV shows. The show was planned around the audio, so there were great expectations. Chet lives in Nashville and would drop in from time to time to check on the progress. The producers and directors were editing all day, every day at Post Masters with Terry Climer. The producers/directors rarely made visual edits that would have an adverse affect on the music. The remix method was simple and easy to use. The original live tracks were cut on an Ampex 1200 24-track, and they were remixed to another Ampex 1200. The original tracks were locked to a Sony 5400 3/4-inch VCR with a BTX 4600 synchronizer, so that David Palmer could mix the show while looking at the pictures.

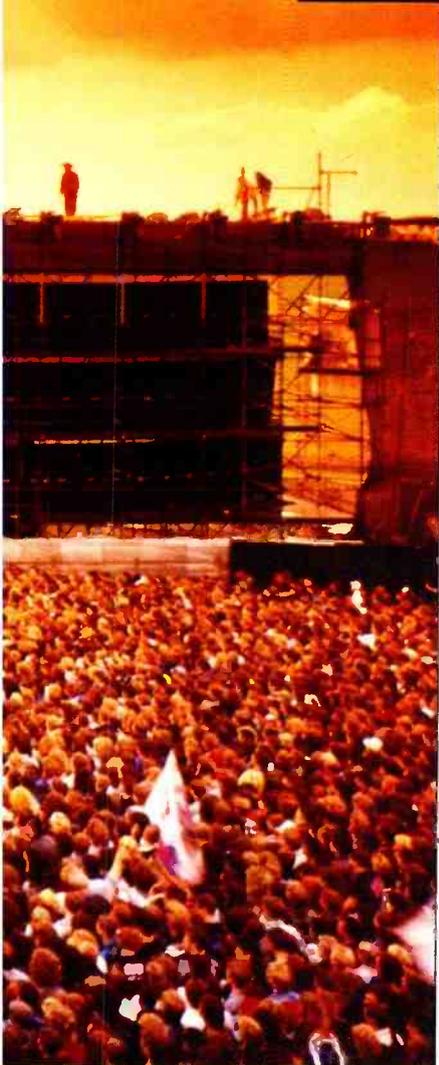
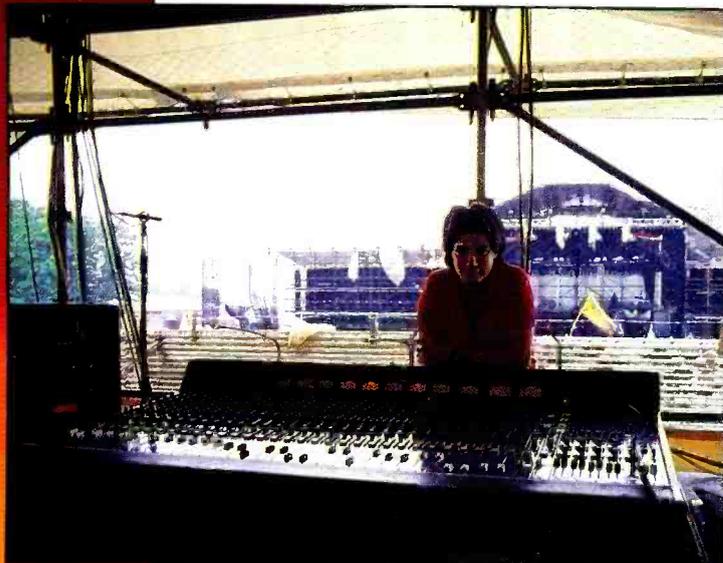
Continued on page 70

Building the Biggest PA in the Galaxy

By Ben Duncan



The biggest single source PA system ever constructed, rated
in excess of 500kW, has won an entry in this year's
Guinness Book of Records.



The motorsport racetrack at Castle Donington in the English midlands has been the location for a nine-year series of annual Heavy Metal, one-day outdoor concerts. At the last festival, on August 20th, 1988, eight bands played, headlined by Iron Maiden. They used the biggest single-source PA system ever constructed. Rated in excess of 500kW, it's won an entry in this year's "Guinness Book of Records."

Throughout the 1980s, UK touring bands have had to weather (for diverse reasons) tough economics. Most of the 1970s headline bands that owned their own PA systems have long ago "realized their assets."

Iron Maiden is an exception. In 1983, after two years of successful touring with Turbosound's own PA rental company, they decided to buy the rig they were then using, including 24 TMS-3 boxes for the main house system.

At the time, the idea of investing in such a huge (and partly "second owner") PA would have looked outrageous to record company accountants. However, Iron Maiden's management had some massive world tours planned. Their decision has

paid off because over the past six years, the band has averaged seven out of every 12 months on the road—including one two-year period in which they toured continuously for 13 months! Since then, the original PA has grown substantially to include 100 TMS-3s and 24 TSW-124 subwoofers.

Doug Hall, Iron Maiden's house engineer, saw at Donington the opportunity to use their PA for the whole event, but knew they'd need more equipment. "I like a system to exhibit comfortable headroom, about three-quarters of its total output," he explained.

Logistics

Dick Bell, Iron Maiden's production manager, then contacted Mike Low at Britannia Row, a major UK PA rental company, who was coordinating the sound at Donington. Mike set about arranging a further 400kW array with the help of Samuelson's Concert Productions (another UK rental company that, incidentally, was bought by Britannia Row the week after the concert).

Extra cabinets and amplifier racks were brought in from Regiscène in France and Ampco in Holland. Meanwhile, in Brit Row's warehouse, a section of the planned

Ben Duncan is a London-based free-lance writer.

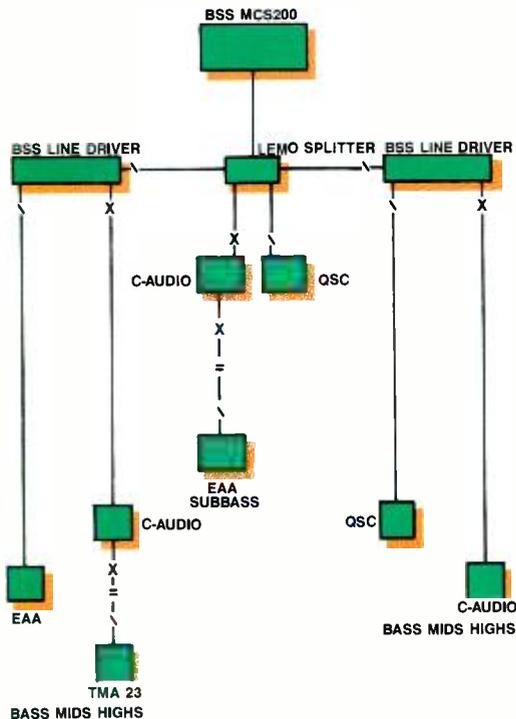


Figure 1. General interconnect scheme.

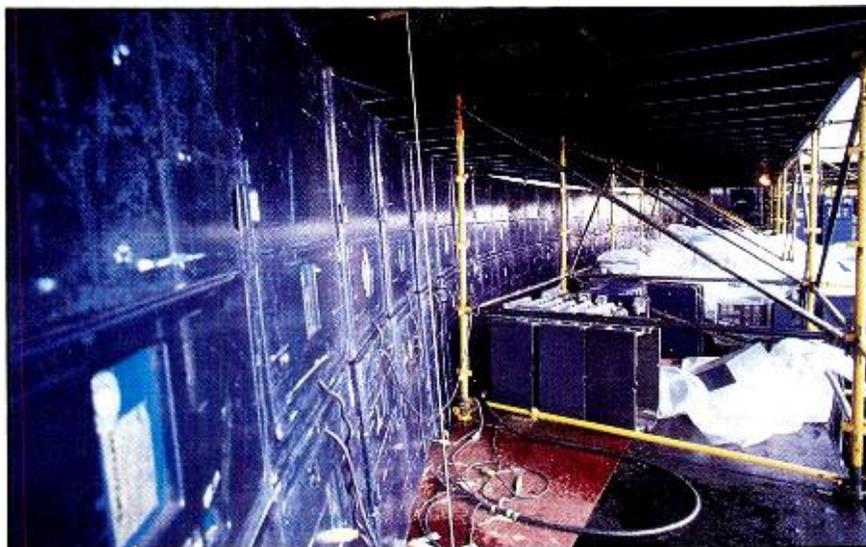
Table 1. House PA amplifier checklist.

Quantity	Model	Source
72	QSC 3800	Iron Maiden
102	C-Audio SR-707*	Samuelsons
6	Turbosound TMA-23**	Samuelsons
18	EAA 1000	Regiscène
6	Carver PM 1.5	Regiscène
14	HH V800	Ampco

* modified SR-606s

** Each tri-amp has three stereo channels.

total = 460 amplifier channels in use.



array was evaluated to simulate the intended coverage two months beforehand.

The setup

On-site setup took five days. Monday was spent erecting the main speaker array. The consoles, stage monitoring and the single delay tower array were rigged on Tuesday. Wednesday was "tech day": The system was fired up, hums were excoriated, the polarity of more than 1,700 individual components was checked, and the cabinet array in the PA wings' upper tier was adjusted. Thursday and Friday were reserved for sound checking for Iron Maiden and the support bands. This meant the instrument backline could be left in place backstage, positioned in order of appearance.

Before 1986, Iron Maiden guitarists Dave Murray and Adrian Smith had used the traditional Marshall amp stack (the kind that's permanently cranked up to maximum), together with daisy-chained FX pedals. The sound was neither clean nor quiet enough, so Hall generally used a clean feed from the two guitarists by mic'ing individual cabinets backstage. At Donington, however, stacks of Galen-Kruger amps and a rack of TC Electronics guitar FX cleaned up the sound sufficiently for Hall to mic the on-stage amps directly.

The mics used for Iron Maiden included a Shure SM-7 on bass guitar; Shure SM57s for the two guitars; an SM98 on the hi-hat and another on the tubular bells; AKG D-112 on the kick drum; Neumann KM-84s over the kit, together with SM57s for the remainder. Sony diversity wireless mics were used for the vocals. All the mic lines were fed into BSS MSR-604 active microphone splitters, which provided buffered feeds for the three house and three stage monitor consoles, as well as an isolated feed for the BBC's recording truck. A multicore was also laid from the house consoles to provide effects returns for the BBC. From the splitter rack to the house consoles were three snakes apiece, each with 19 pairs, supplying 3x40-channel capability, with spare pairs.

As usual, the aim was to keep show downtime to a minimum, with each successive instrument/amp array and drummer fully miced and waiting behind the backdrop. In theory, it was only necessary to plug the multicore into the next stage box. The actual duration of the seven changeovers ranged between 20 and 25 minutes. To achieve the rapid changeover while keeping Iron Maiden's own control gear intact required an on-stage crew of 24 and three substantially independent monitor systems. The support bands' monitoring alternated between a pair of Soundcraft series 4, 40/16 consoles (supplied by Samuelson's), with their own EQ

racks, and Carver PM 1.5 amps. John Shearman and Ed Wilson from Samuelson's did alternate duty on the support bands' monitor mix. Alongside was Maiden's own 24/10 Midas, with a third set of control gear. Their power amplifiers were mostly QSC 3800 (lo + mid), with a few Turbosound Fan Amps, and Turner B502s (on the vocal wedges).

The support bands "A-B"ed between two sets of custom stage monitor wedges, designed by Pete Brotzman when he worked at Turbosound Rentals. They all used TMS-3 side- and drum-fills from Iron Maiden's 40kW monitor system. Maiden used their Turbosound vocal wedges along with vocalist Bruce Dickinson's sidefills, which are rated at 12kW and combine a pair of Martin 215 bins with Community's M4, a 4-inch exit mid-range compression driver noted for its ability to deliver crushing SPL.

House control

In the mix tower were three consoles, each with its own outboard gear. Iron Maiden used its own Amek M1000 48/8/2, with custom modifications and a 16-input add-on stretch-frame for the drums. The support bands alternated between two Soundcraft Series 4, 40/16/2.

Hall's extensive FX racks contained Eventide SP2016 (used on the drums); Eventide 949 and 969 Harmonizers; dbx 160 and 165 compressors; two ADR Compex limiters; two UREI 1176 compressor-limiters; an Aphex C; an AMS RMX-16; an ADR vocal-stresser and dbx's RTA-1 analyzer. Hall also used a Revox B-77 for intro sequences and voice-overs. He says, "Iron Maiden does a lot of their own FX on stage. Being outdoors, I'm using some general 'small room' ambience to add some lushness. I compress the bass with the UREI 1176, followed by the ADR Compex and some post EQ. I compress the vocals by 5dB, so when Bruce goes berserk on the mic, he doesn't kill people!" After 8 years of touring, Hall knows the cues well enough to mute all the unused mics in a flash. This minimizes spillage from Maiden's enormous monitoring and on-stage amp array.

The main PA

Table 1 shows the 460 amplifier channels in use on the two PA wings. These were divided between stereo amps, half-used stereo amps and some TMA-23 stereo tri-amp units (a Turbosound market research product no longer manufactured). Owing to their diverse sourcing and the differences in rental companies' interconnect standards, the amplifiers needed very, very careful matching.

John Newsham from Turbosound and Julian Tether (then Samuelson's chief

technician, now working for Britannia Row) spent an evening religiously checking and correcting sensitivities and normalizing the polarity of every amplifier. Five more hours were spent adjusting the system grounding to reducing residual hum and noise.

At Hall's insistence, the 400+ amplifier inputs were driven from a single BSS MCS series crossover. This approach promised sonic coherence, but with incompatible amplifier racks supplied by four different

companies, it also presented a nasty interface headache. Any failure would silence the main PA outright. Stress on the crossover outputs was avoided by using BSS's newly developed high-power line-driver "booster" package, made to plug into the MCS series crossover "mainframe." It was first devised in association with Concert Sound to drive 200+ amplifier channels at the Nelson Mandela birthday bash. At Donington, two line drivers were placed at the stage-end of the returns

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Circle (24) on Rapid Facts Card

January 1989 *Recording Engineer/Producer* 49

NOTE: TOP DECK OMITTS
2-INCH WEDGES AT REAR

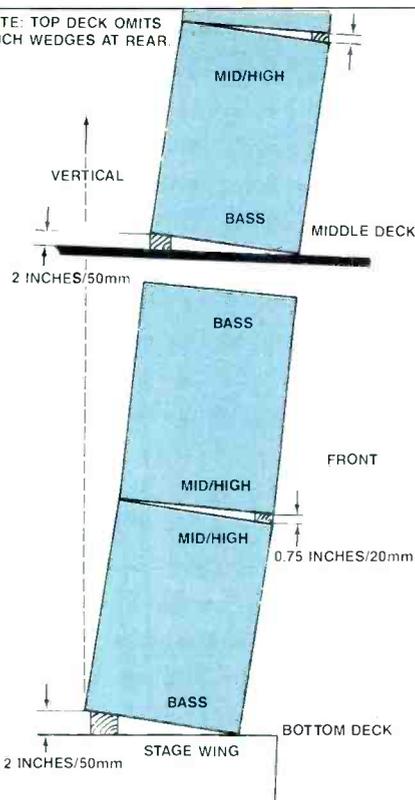


Figure 2. Every cabinet was tilted down with lumber wedges acting as "acoustic compensators." Smaller wedges at the front fine-tuned the coupling between adjacent mids.

multicore. Figure 1 shows the general interconnect scheme.

The main house system comprised 304 Turbosound TMS-3 enclosures. Each 72-foot stage wing (see Photo 1) had three tiers. Each tier had 48 Turbosound enclosures in two rows, totaling 144 per wing. The remaining 16 were placed under the front of the stage, with a separate vocals-only mix to balance the plainly audible instrument amp stacks and side- and drum-fill monitors. In the wings, the enclosures were stacked vertically: first, to gain the longest throw from the HF horn, and second, to close-couple the 10-inch mid drivers. On the ground, under the wings, were 60 Turbosound TSW-124 subwoofers, all arrayed for maximum coupling.

The TMS-3 is normally configured as a medium-throw "box." Nevertheless, the capabilities of this giant array were clearly proved during initial sound-checking—by the PA's clarity in a large village 2 miles away.

At Donington, flying the PA with the available scaffolding would have left a big gap between cabinets, ruining the coupling. Instead, every cabinet was tilted down (see Figure 2), with lumber wedges

acting as "acoustic compensators." Smaller wedges at the front fine-tuned the coupling between adjacent mids.

Each tier was built 18 inches in front of the next, while the enclosures in each wing were arrayed in an arc, receding 10 feet at each "wing tip," to produce a pair of phase-coherent, virtual point sources. This allowed a relatively constant SPL to

"With this system, you can get the level without distortion that hurts your ears."

be maintained over the 60-meter span, back to the mix tower.

The small delayed PA, located immediately behind the mix tower, was rated at 50kW. Because of the more-than-adequate bass and subbass projection from the main system, the delay tower's low end was barely "ticking over," and it needed no subwoofers. John Newsham said, "It's not there to start the sound all over again, it's there to give a gentle lift to the mids and highs." A look at the site

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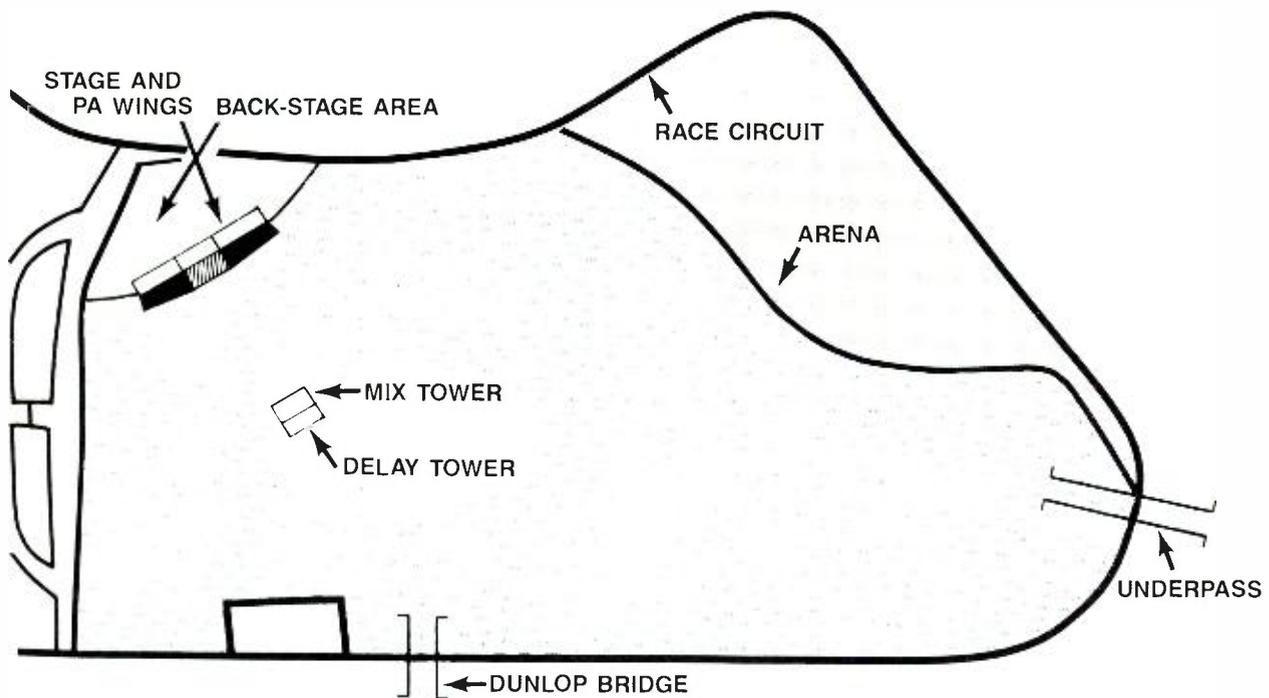


Figure 3. Castle Donington site plan.

plan (see Figure 3) shows why: The land rolls away from stage-left into bowl-shaped dip, then up into the main bowl and goes on rising gently to a ridge well beyond. Donington's infamous "dog leg" means a major part of the audience is well off-axis from the stage.

The delay array consisted of 30 MSI (Maryland Sound Industries) HF and 16 LF cabinets. In addition, eight JBL long-throw horns at the top of the tower provided additional reach for the HF material. This array used the stage-right mix to "fill out" the stereo image 500 yards from stage-left.

During Iron Maiden's set, the instantaneous peaks reached 124dB at the mix tower.

This is a technique first applied by Malcolm Hill, a UK PA rental company that has provided sound at Donington on past occasions. The delay tower components were driven off their own crossover, through a Klark-Teknik DN-700

delay line. Stereo balance, horn aiming and delay times were first estimated; the delay was 1ms/foot. With sound crew radioing back the results from a golf-cart, the fine-tuning was accomplished fairly easily.

The event

The warm day began with hazy skies and later clouded over. By 6 p.m., attendance had reached 107,000. The inevitable rain came after nightfall, during Maiden's set. The good side is that wet air seems to improve the transmission of sound.

After six years on the road and literally thousands of gigs to his credit, Doug Hall (who's been with Iron Maiden since their first UK tour in 1980) said, "I don't want a PA to sound like 'big speakers.' For me, it should be an extension of the stage, like it's coming straight from the band. The PA should disappear. With this system, you can get the level without distortion that hurts your ears." The philosophy is also supported by Motley Crue's engineer Mark Dowdell.

Since the support bands' sound peaked at between 6dB and 8dB below full output (which means one-fourth power, or

about 125kW), the remaining headroom was more than comfortable. Under these conditions, the short-term rms SPL, monitored at the mixing tower, was 118dB. It was 108dB at the Dunlop bridge, 185 meters (more than 606 feet) from the stage, and nearly on-axis.

During Iron Maiden's set, the short-term rms SPL was similar, but there were more dynamics; instantaneous peaks reached 124dB at the mix tower, compared to 119dB during KISS's set. 124dB corresponds to the system being driven briefly to within 0.5dB of full power.

What was achieved? Well, I believe that this is the first time a group of Heavy Metal bands have played (in the open air to more than 100,000 people) with enough power headroom to play *loud* and develop a peak-to-mean ratio (i.e. "dynamics") of more than 10dB—without stunning the limiters.

REP

Coping with Wireless Microphone Systems

By Bill Mayhew

Help is here for those who need to overcome wireless system difficulties.

Wireless microphones have become a convenient way to deal with many difficult production situations involving actors and musical artists who need freedom of movement or concealed audio pickup. Setting up a single wireless mic is a relatively easy task. Yet behind the scenes, in the invisible world of RF, trouble lurks.

Outside interference

To start, let's talk about interference that is not caused by other wireless microphones being used at the same time and at the same location. For lack of a better term, let's call it *outside* interference.

Most people would say that you should have no trouble if you have taken the usual precautions in selecting the transmit frequencies (for the given area) of the wireless mics you are going to use. The real trouble is in the use of the word *usual*. Most manufacturers design their wireless mics to operate in the frequency group between 154MHz and 216MHz. Of course, within this group of frequencies are walkie-talkies and the VHF TV channels. Table 1 shows the relationship between VHF channels 7A through 13F and the licensable wireless frequencies.

The unused TV channels within a given area are a good place to start when selecting a workable frequency. I've found that if you are further than 75 air-miles from a station's transmitter, you should be able to use wireless within that channel's bandwidth. For instance, in the Los Angeles area, the unused channels are 8, 10 and 12. The problem is that if you were to take

a wireless that was set on the frequencies for 8, 10 or 12 to San Diego, it would not work because the San Diego area has TV stations broadcasting on channels 8, 10 and 12.

Happily, the FCC has opened up a new group of frequencies that allow for clear-channel operation within the United States: 169.445MHz, 169.505MHz, 170.245MHz, 170.305MHz, 171.045MHz, 171.105MHz, 171.845MHz and 171.905MHz. (At least, I think that was the intent of the group that convinced the FCC that clear-channel wireless mic frequencies are really needed. The truth is that the FCC took the list of frequencies submitted and changed it a bit.)

At first glance, we all yelled, "Eureka!" But then we realized that the spacing was far too close. In fact, you can only get three of the above frequencies to work together at the same time. The rest of the bad news is that in some areas of the country, wireless users have had interference on these frequencies, too. So far, in the Los Angeles area, I have found the following frequencies to be free from interference: 169.445MHz and 171.105MHz. The reason these new frequencies don't work out as planned is that the spacing between frequencies is for walkie-talkies, which is too close. The FCC didn't take the wireless mic deviation of 15kHz into account.

What interference sounds like

If you are lucky, the interference will be "understandable" in your wireless receiver; you will be able to hear the station clearly enough to get the call letters of the station that is causing the in-

terference. With this information, you or your service tech will be able to tell if you have a fault with the wireless or if you are, in fact, on the wrong frequency for a given area.

Most of the "outside" interference in wireless microphone receivers comes blasting through—usually causing the receiver's VU meter to peg. It's usually a commercial TV broadcast station (and to a lesser degree, FM radio) that causes the interference. Since they use FM modulation with a deviation of 75kHz, you can see why the wireless mic receiver (12 to 15kHz of deviation) really grabs on to the outside signal.

To see if you are going to have outside interference, a good on-site test is to turn on the receivers and open the squelch. Now listen to the spectrum noise. If you hear a carrier, you might want to drag out that old mic cable.

If you suspect outside interference and you enjoy playing Russian Roulette with a shotgun, there are a couple of things you can do. First look at the receiver's meter. If the interfering carrier signal reads higher than -20VU in the RF position (meters vary by brand), you will probably have trouble. Even though FM wireless mics work on the capture principle, (the strongest signal is captured by the receiver), this is not much help because your transmitter is moving about the performance area. How strong must a signal be to interfere? I found that a 2 μ V (0.00002V) can cause interference problems.

With movement, the transmitted signal is being received at varying strengths by

Bill Mayhew is owner of Mayhew & Company, Burbank, CA.

the receiver. During a period of weak reception, the interfering signal could be received by the wireless receiver. If the interference is -20VU or less at the receiver, try moving the receiver a few feet. This could make all the difference in the world.

The idea is to find a location with the least amount interference. You might also try different antenna polarizations. Sometimes a 45° angle works well. Remember that the wireless transmitter and receiver antennas must be polarized in the same direction (See Figure 1.) There are no hard and fast rules for polarizing a hand-held system. Typically, a 45° angle works best. With an antenna diversity system or a switching diversity system, one antenna should be aligned vertically, and the other at 45° .

Antenna diversity and switching diversity systems

To capture the strongest possible signal and minimize dropouts, two reception techniques are available: antenna diversity and switching diversity. In an antenna diversity system, typically two or three antennas are employed, each positioned in a different dimensional plane. The antennas are combined in an antenna combining box/splitter, which feeds up to four standard radio receivers. As the transmitter moves about the performance area, the presence of a strong signal is reasonably assured at one of the antennas.

With antenna diversity, the signals are not switched as they are in a switching diversity system. With careful antenna placement, the antenna diversity method can provide the best possible coverage, no switching noise and a minimum of potential phase cancellation problems. Also, this method generally has no limitations as to brand type and is less expensive than a

switching diversity system.

In a single switching diversity receiver there are two antennas. The RF signal-strength from each IF amp controls the audio switch. When one side of the switching diversity receiver senses a weak RF signal, the switch switches to the other side of the receiver, which, at that instant in time, should be receiving a strong signal. (See Figure 2.) While switching diversity systems have a convenience advantage, they are generally more costly and slightly less reliable than a carefully installed antenna diversity system.

Another obvious (but often overlooked) key to successful wireless operation is maintaining a minimum distance between the transmitter and receiver. This helps ensure that the receiver will only "capture" its transmitter and not the interference. One last point, make sure that the receiver is turned off when the transmitter is not in use. This prevents unwanted transmissions from getting into the receiver and possibly leaking through the audio system.

A very common misuse of wireless systems occurs when the engineer thinks the receiver should be at the console. While this may or may not be important, the antenna most certainly should not be there. The antenna and receiver should be as close to the performing area as possible. This means there will be either antenna coaxial cable or audio lines going to the stage. "Wireless" refers only to the signal path between the performer's mic and the antenna.

Frequency selection

If all you need is one or two wireless microphones, then the frequency selection process is quite simple. The problem becomes more complex when you require many more channels. They have some

Table 1. VHF TV channels and the corresponding wireless frequencies.

Channel	Frequency (MHz)
7A	174.8
7B	175.4
7C	175.8
7D	177.0
7E	178.0
7F	178.8
8A	80.8
8B	181.4
8C	181.8
8D	183.4
8E	184.0
8F	184.8
9A	186.8
9B	187.4
9C	187.8
9D	189.4
9E	190.0
9F	190.8
10A	192.8
10B	193.4
10C	193.8
10D	195.4
10E	196.0
10F	196.8
11A	198.8
11B	199.4
11C	199.8
11D	201.4
11E	202.0
11F	202.8
12A	204.8
12B	205.4
12C	205.8
12D	207.4
12E	208.0
12F	208.8
13A	210.8
13B	211.4
13C	211.8
13D	213.4
13E	214.0
13F	214.8

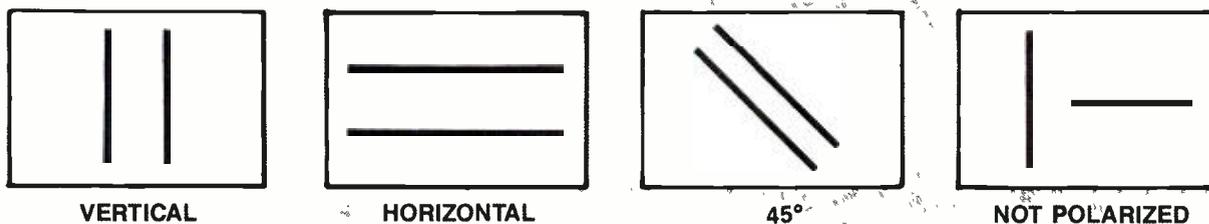


Figure 1. Antenna polarization options.

degree of compatibility, but as I said before, the problem can get quite complex.

If you consider IF frequencies, bandwidth, spurious emissions and image rejection, the problem becomes apparent. The sidebar "Frequency Selection" on page xx shows a formula that allows up to 22 wireless channels to be used at the same time, in the same location. To do the math long hand involves thousands of addition, subtraction and comparison problems. Of course, with the aid of a computer, the tedious task is reduced to just entering the parameters you wish to use. If you are not up to doing the calculations yourself, the answer for frequency selection can usually be found with the dealers and/or manufacturers of the wireless equipment.

Squelch

The purpose of squelch is to mute the spectrum noise received when no carrier is present (similar to a noise gate). There seem to be two schools of thought when it comes to squelch controls in wireless microphones. The first is that the end-user should not have to adjust the squelch on the receiver, so the manufacturer puts the squelch control inside the receiver, away from the end-user. This is done to avoid potential misadjustment of the squelch by the end-user.

The second approach suggests that a preset squelch control does not allow for

the most sensitive receiver. Further, it does not allow the operator to disable the squelch, if desired. Also, with a user-adjustable squelch, it is possible, at times, to squelch out faint interference.

One additional thought. You should readjust the squelch whenever a unit has been in for repair, when you change locations or if you have reason to believe that someone has changed the control.

The antenna system

Quite often I am asked how much coaxial cable can be connected to a wireless receiver to extend the antenna. I usually suggest 50 feet of RG-58 or 100 feet of RG-8. It seems that everybody has a different idea when it comes to coax length. In trying to find the standard losses of coax cable, I looked into a number of publications, all of which gave me a different answer. So I decided to try some measurements in the *real world*. I set up the test at home, in an open area to help reduce reflections.

First I connected a dipole antenna to our spectrum monitor using 3 feet of RG-58, which allowed me to polarize the antenna for maximum reception. Figure 3 shows a typical dipole antenna. The signal was measured with the transmitter placed 40 feet from the spectrum monitor. Then we disconnected the 3-foot section and in its place connected 25 feet of RG-58. To compensate for the loss in the additional cable, we had to move our dipole anten-

na 10 feet closer to the signal source. Figure 4 shows a graph of what we found.

This exercise shows that coax cable is better than air. It was quite surprising how little loss the cable had in this real-world application. What this means is that if you can improve your line of sight by relocating your receiver antenna, by all means, do it! As the data points out, a small distance closer to the transmitter more than makes up for the loss caused by additional cable length.

Now before everybody goes out to install coax, there are some finer points to consider. First, you should use a *dipole* antenna. If you just try to remote the whip antenna that came with your receiver, it usually will not have a good ground plane. Next, be very careful about the type of coax used. Appearance itself is no guarantee. Using an ohmmeter, you should read close to 0Ω center to center and also from shield to shield. You should read an open between the shield and the center conductor.

If you suspect trouble with the antenna, check the coax first—it may save time and money at the repair bench. Last but not least, select the location of your remote antenna carefully. It should be at least 2 feet from any metal, the coax leading to the dipole should not parallel the antenna elements and finally the antenna should be polarized to match the polarization of the transmitter antenna.

With all of the wireless microphone ad-

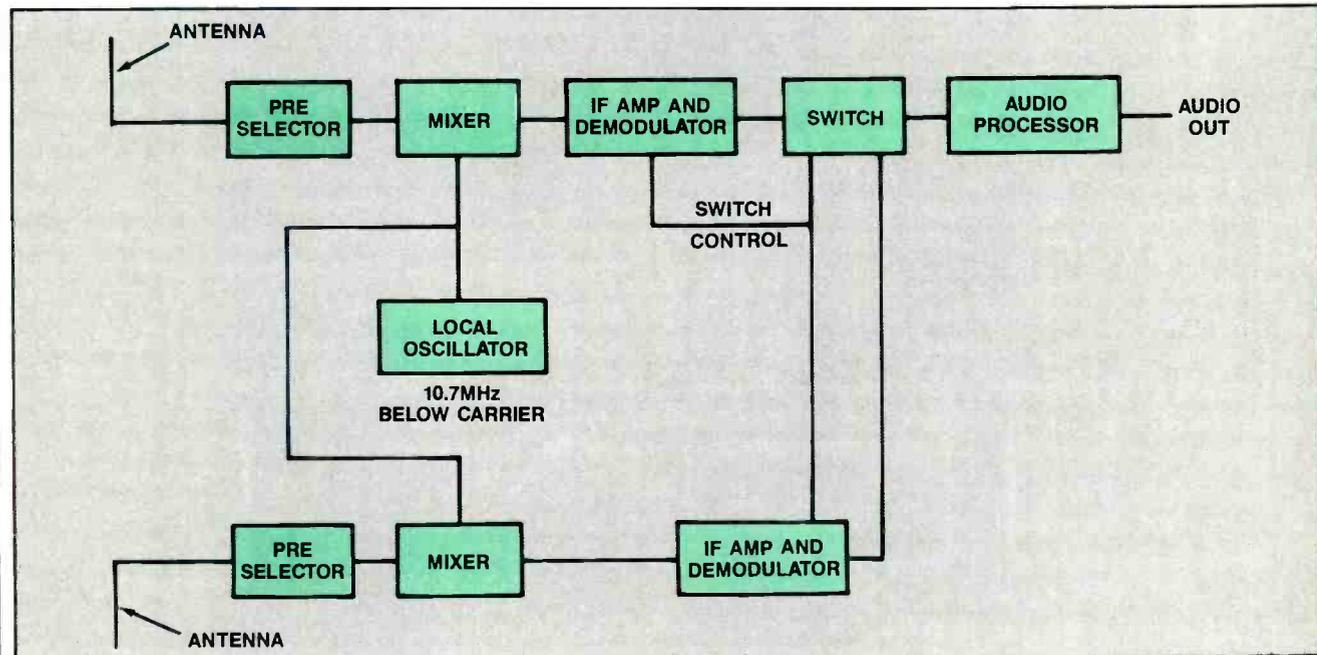


Figure 2. A typical switching diversity receiver.

Frequency Selection

The frequency selection formula reads as follows: IFI (intermediate frequency interference) equals f (frequency of the wireless) minus the IF, which is 10.7MHz, and products of the IF. Subtract multiples of the IF from the original frequency a total of four times ($-1F$, $-2IF$, $-3IF$ and $-4IF$). Then add multiples of the IF (10.7MHz) to the original frequency four times ($1F$, $2IF$, $3IF$ and $4IF$).

I have chosen three regular frequencies to work with in this example. There is a column for each frequency (174.0MHz, 184.8MHz and 195.4MHz). (See Table A.)

The next step is to see if any of the frequencies in any of the columns are within 0.7MHz of each other. If they are, then you will have some problems. The wireless system will be unusable if the numbers fall within 0.7MHz of the frequencies marked with an asterisk (*), that are below the carrier frequencies marked with an f . In other words, the numbers in the three columns will all generate some interference; those marked with an asterisk definitely will.

It is important to understand that this formula only tells you the frequencies that will cause problems; it doesn't calculate the workable choices. As you can see, 174.0MHz and 195.4MHz will not work together, nor will 184.8MHz and 174.0MHz or 184.8MHz and 195.4MHz. The only variable in this is the amount of tolerance you are willing to put up with. 0.7MHz seems to work very well—however, I'm sure you can lower that number and still have some degree of success. In this example, it will be necessary to change two of the carrier frequencies to have three usable mics.

This method only takes into account the local oscillator and the harmonics of same. In doing so, though, it happens to take into account other methods and computations. This is a very conservative method of frequency selection—that happens to work.

Table B shows three frequencies that *will* work well together, providing there aren't other outside interferences to contend with. Notice that more than 0.7MHz separates the frequencies in this example.

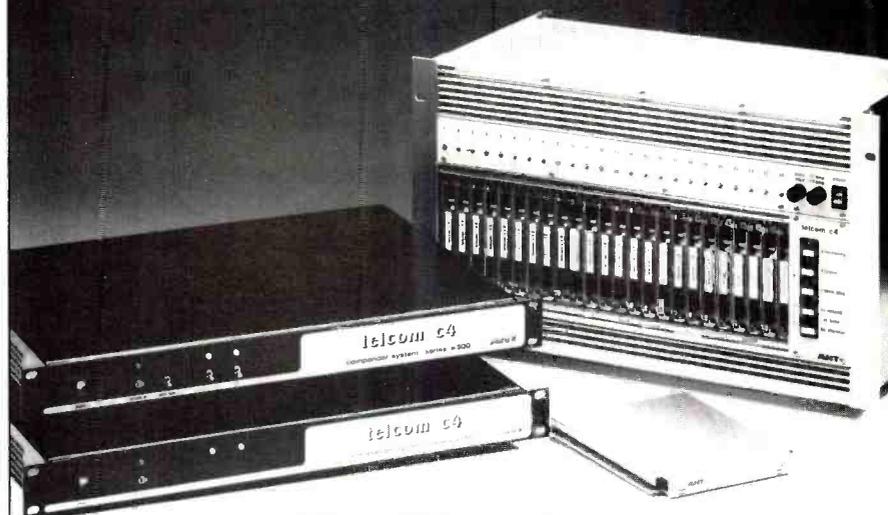
Table A. 174.0MHz, 184.8MHz and 195.4MHz should not be used together in a wireless microphone system. A calculation of their harmonics shows there will be interference.

216.8	227.6	238.2	($f + 4IF$)
206.1	216.9	227.5	($f + 3IF$)
*195.4	*206.2	*216.8	($f + 2IF$)
184.7	195.5	206.1	($f + IF$)
174.0	184.8	195.4	(f)
163.3	174.1	184.7	($f - IF$)
*152.6	*163.4	*174.0	($f - 2IF$)
141.9	152.7	163.3	($f - 3IF$)
131.2	142.0	152.6	($f - 4IF$)

Table B. Harmonics show that 180.8MHz, 181.8MHz and 184.8MHz are compatible in a three-mic wireless system.

223.6	224.6	227.6	($f + 4IF$)
212.9	213.9	216.9	($f + 3IF$)
202.2	203.2	206.2	($f + 2IF$)
191.5	192.5	195.5	($f + IF$)
180.8	181.8	184.8	(f)
170.1	171.1	174.1	($f - IF$)
159.4	160.4	163.4	($f - 2IF$)
148.7	149.7	152.7	($f - 3IF$)
138.0	139.0	142.0	($f - 4IF$)

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January 1989 *Recording Engineer/Producer* 55

The Ten Commandments of Wireless Microphone Use

1. Keep the distance between the transmitter and the receiver's antenna to a minimum.
2. Maintain a direct, unobstructed line of sight between the transmitter and the receiver's antenna.
3. Keep the antenna and mic cables on the transmitter separated.
4. Place the transmitter body pack as high up on the body as possible.
5. Have both the receiver and transmitter antennas polarized.
6. Place the receiver at least 4 feet high.
7. Keep the receiver a minimum of 3 feet from any other receiver or antenna.
8. Keep the receiver a minimum of 3 feet from any metal.
9. Use only fresh batteries.
10. Contact the manufacturer or a qualified consultant if you have trouble determining the usable frequencies.

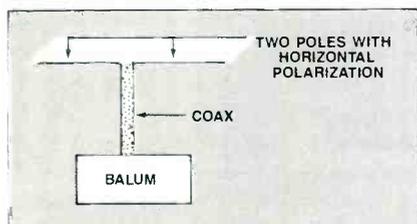


Figure 3. A typical dipole antenna.

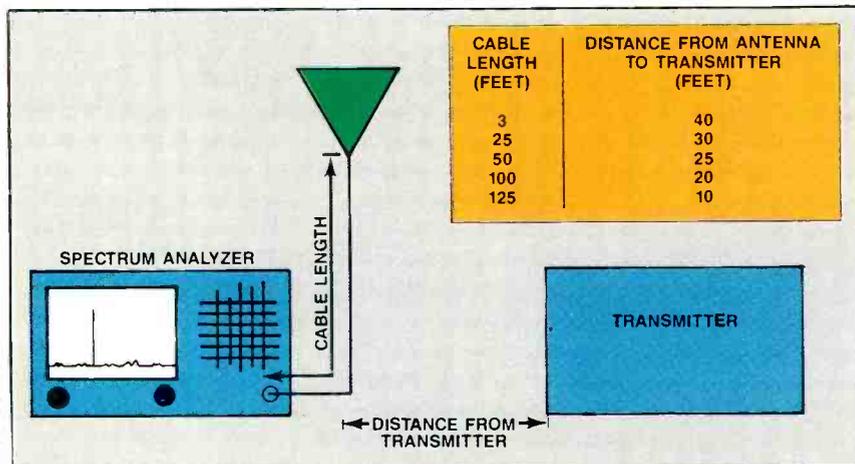


Figure 4. Relative distance from transmitter to antenna vs. coax cable length/feet.

vances that have taken place in recent times, a new gremlin has raised its ugly head to unforeseen heights. The gremlin is *clothing noise*. Yes, that's right; finally a gremlin that is not the wireless microphone's fault. A number of times I have had A-1s (video's terminology for first audio engineers) call me and complain about drop-outs. After investigation, more times than not, the problem proved to be clothing noise.

Ironically, with clothing noise, the prime cause is usually inexperience. Consider this: In both TV and motion picture production, the A-1 and/or the mixer is usually the most experienced person on the audio crew. In most production situations, the A-1 is removed from actually "dressing" the talent, i.e. attaching, placing and concealing the mic, transmitter and antenna in the talent's clothing. Therefore, dressing the talent is usually left to the A-2—or even a wardrobe person, whose main concerns are not audio quality. Figure 5 shows a "dressing" mic placement method that has proved quite useful.

Take a 2"×1/2" piece of gaffer's tape. This forms the cross arm of the T. A second 1/2"×3" or 1/2"×4" piece of gaffer's tape forms the staff of the T. Then place the omnidirectional mic onto the sticky side of the tape. The top of the mic should reach just to the top of the cross arm and align with the staff of the T. The mic cable should be dressed down the staff of the T. Then slip this entire mass of tape, microphone and cable under the talent's shirt, blouse or coat with the sticky side of the tape toward the outside of the garment.

To help eliminate microphonics caused

by the cable, tie the cable in a snug half hitch. In effect, this dressing system causes the microphone to become part of the clothing. The clothing noise is greatly reduced because the microphone now moves with the clothing instead of against it. One word of caution: This system works well on most fabrics, but is a real mess on some synthetics. Use a small piece of gaffer's tape and test delicate or expensive fabrics *before* installing this system. The adhesive may cause irreparable damage.

Neon interference

I really can't think of any other word that strikes as much fear into the hearts of knowledgeable users as *neon*.

Some time ago, I was given the task of making a number of wireless mics work in the presence of lots of neon. The neon person managed to enclose the proscenium of a theater completely with neon tubes. As I remember, there may have been three different-colored tubes running in parallel around the proscenium arch.

As I recall, I didn't find out about the neon until I arrived at the theater. I suppose the producers didn't want to hear any negative thoughts, especially since they had spent *big* bucks on the set.

Originally, I set the receivers up on the stage and connected them to an antenna diversity system using three dipoles, located stage left, stage right and center stage in the footlights. I arranged them thus to take advantage of the capture effect of using three receivers. If properly placed, using three antennas removes 97% of the dropouts.

Remember the old Frankenstein movies with the "Jacobs ladders" in the background? They consist of two wires set vertically. At the bottom of the ladder, the space between the two wires is rather close, and at the top, the space between them is wider. When a high voltage is applied to the two wires, the arc starts at the bottom and "climbs" to the top, where it finally cannot bridge the gap any longer, and the spark is squelched, usually with a loud snapping sound. The whole process continues to repeat itself until the power is turned off.

This adequately describes the sound that was coming from the wireless channels. It was not in the background; when it was heard it usually competed with the program audio. Also, the interference seemed to be greater when the wireless transmitter was close to the neon. Of course, the director wanted the MC to be next to the neon when he introduced the

various acts or when going to a commercial.

I went back to the office to pick up the spectrum analyzer. When I hooked it up on site I found that the effective radiated RF signal from the neon tubes was, at worst, equal to the wireless transmitters'. In addition, the neon bandwidth was full-spectrum. It went from 5MHz to 1GHz. I am sure it was wider, but the spectrum analyzer only measures from 5MHz through 1GHz. By connecting a whip antenna to the spectrum analyzer with a piece of coaxial cable, I could see standing waves along the tubes, as well as from the high-voltage leads feeding the tubes. The strength of the neon signal was equal throughout the entire bandwidth.

I also found that the neon RF field was also being coupled to the entire building's 110Vac line through the primary side of the high-voltage neon transformers. I noticed that the high-voltage wires were just twisted together at the splice points and not shielded. Nor were the transformers grounded. In effect, I had a very broadband (dc to light), high-level signal source that was using the entire proscenium arch, plus the high-voltage feed lines, as an antenna. Couple this with the fact that the broadband signal was also traveling down the ac lines, and I think you can see why I started to think about moving to Hawaii and opening a reef-shoe rental business.

First I met with the neon person. He seemed to become quite excited and happy when I showed him the field that his neon was radiating. He started to jump up and down and mutter utterances similar to "Far out, man!" I suggested that he ground the cases of his eight transformers. This actually started to have a positive effect because he started the grounding process by screwing a rather long sheet-metal screw into the transformer case and then connecting it to the ac ground. After a very short time, the transformers would short out, the neon would stop working and the wireless system would start working.

It took him a couple of hours to replace the transformers. When he grounded the new ones, he attached the ground by using a nut and a bolt through the transformers' mounting ears. I also suggested that he solder all splices.

With all the interference, I decided to move the receivers and the antenna diversity off of the stage and into the balcony. I also hoped that I would be on a different ac circuit.

Almost all professional wireless

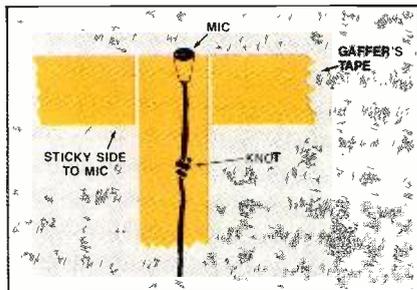


Figure 5. Attaching a wireless mic to the talent's clothing.

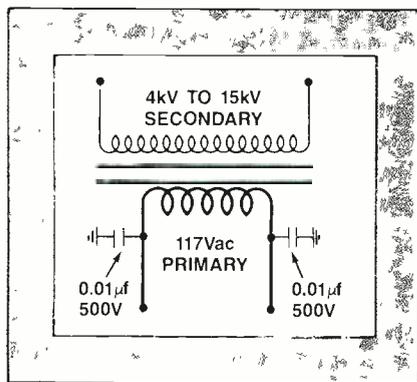


Figure 6. By-pass capacitors used to buffer the ac line.

receivers have good RFI (radio frequency interference) filters built into the ac circuit. However, the filters only remove the RFI from the ac line after it gains entrance to the receiver. If the RFI is strong enough, it can radiate from the ac line just outside the receiver and cause big problems.

So, I decided to run the receivers on dc. This seemed to help. I have since learned that a high-quality power strip/surge protector/noise filter (the type you would use to protect your PC) helps clean up the ac line radiation. Come to think of it, if you added these surge protectors on the primary side of the neon transformers, you could reduce the ac line radiation even further. If the surge protectors are not available, you can use some bypass capacitors hooked up as shown in Figure 6. The value of the caps should be 0.01µf and at least 500V. This is not as good as the off-the-shelf protectors, as they also have in-line inductors to shunt the RF.

I also ended up removing the antenna diversity system and running the receivers off of their own whip antennas. With all of the RFI in the air from the neon, I was sure that it was mixing with the transmitter carriers and producing products that

would be impossible to calculate. Sure enough, it did improve the operation. Up to this point, I was just working on removing the RFI that was getting into the receivers. What about the transmitters?

The transmitters had metal cases with additional metal shields built into the cases. The interference was most apparent when the MC stood right next to the vertical neon tubes located stage right and left. The RFI was getting into the transmitter through the transmitter's antenna. I found that by having the transmit antenna positioned horizontally on the MC, I again could gain marked improvement. I also rotated the receiver antennas to horizontal, but found that they worked best at a 45° angle.

One additional point of RFI intrusion would be the microphone cable and head that plugs into the transmitter. Since I was using ECM-30 mics and the transmitters had all of the latest RFI modifications on the microphone jack, there was not much I could do. However, I would like to have tried a dynamic mic. I have a feeling that the electret has some real problems rejecting a strong RFI field.

Did it work? Most of the time, it did. On a scale of 10, we started with dismal 3, and after all the troubleshooting and tweaking, it was a tolerable 7.

Here are two more suggestions that should help in the event you are faced with a neon nightmare:

1. Enclose all of the neon tubes in 1/4-inch hardware cloth. (Nobody said it would be easy or cheap.) Ground the hardware cloth to an *earth* ground, not the U-ground of the ac system.

2. Use shielded high-voltage cables to feed the tubes. Since the highest voltage used for most neon is 15kV, this can also be difficult. The cable's shield should also be connected to the *earth* ground.

It wasn't that long ago that most wireless users were more concerned about the system "just working at all," even if it did have a very limited dynamic range, frequency response, noise floor and so on. In recent years, however, this has all changed. Now, some systems have over 115dB of dynamic range and an overall frequency response ± 3 dB from 100Hz to 15kHz. This is more than the dynamic range of the input channels on most mixers. And, the use of switching diversity receivers has eliminated all but the most stubborn drop-out problems.

RE/P

Acknowledgments: Special thanks to Ken McLaughlin for his tips on dressing the talent.

Where Have All The Sprockets Gone?

By Scott Gershin

As alternative methods of sound editing become accepted, more and more production companies are starting to integrate tape and electronic editing with that of 35mm mag strip film.

Alternative methods of sound editing are becoming accepted in the broadcast and film community, with more and more companies starting to integrate tape style and electronic editing with that of 35mm mag strip film. In television, the trend seems to be that a majority of broadcast shows are having their post-production done on tape, while film companies/studios (theatrical releases) are still reluctant to embrace tape-based editing because of a lack of standardized hardware, operating procedures and terminology.

The two most common editing devices for 35mm mag strip film are the moviola and the flatbed (see Photo 1), while editing

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on tape consists of myriad synchronizers, tape machines, mixers and other peripheral devices that have their own operating languages and user interfaces. Fortunately, those in the electronic editing industry are trying to work with manufacturers to create some kind of consistency and standardization, but this can be difficult when technology advances faster than the industry.

So far three styles of non-film editing have evolved: 1/4-inch or 1/2-inch tape, RAM-based sampling, and editing using direct-to-hard-disk recording.

All three styles compile the edited information on a multitrack format that can easily be brought to the dubbing or mixing stage. While most mixing facilities use a 2-inch 24-track format, other formats include 1/2-inch 8- or 16-track, 1-inch

16-track, 32-track digital or 2-inch 32-track analog.

As for synchronizers, it's still an open market, with everyone either praising their own units or creating a new vocabulary to describe them.

Tape editing

Tape editing was the first of the three non-film styles to evolve. Borrowing technology from the record and video industries, editing rooms were created, using a multitrack machine locked to picture with the use of a synchronizer reading time code from a video playback unit. For video playback, a 3/4-inch cassette machine, with time code printed on its audio track (Channel 2) and a window burn (character insertion displaying time code information) corresponding with the printed time code is considered the video master unit. The 24-track is thought of as the audio master and is slaved to the video master with a zero offset. At this point, all other machines are slaved to the video master with an offset corresponding to the particular effect and the desired edit point.

Many facilities have their effects library stored on 1/4-inch tape. A technique for laying in effects from that format (as well as other formats, such as 1/2-inch 4-track, carts, CD) is to use trigger starts to "fly-in" the effect. When doing this, the 1/4-inch machine is not in hard lock. For short effects, most tape machines will stay in sync. When an effect is triggered, the editor can also take advantage of the VSO and half-speed tricks that are not possible without transferring between several tape machines—creating considerable generation loss.

It is advisable to trigger the 1/4-inch source machine slightly before the desired cut-time, enabling the machine to get up to speed. The effect, as well, has to be off-

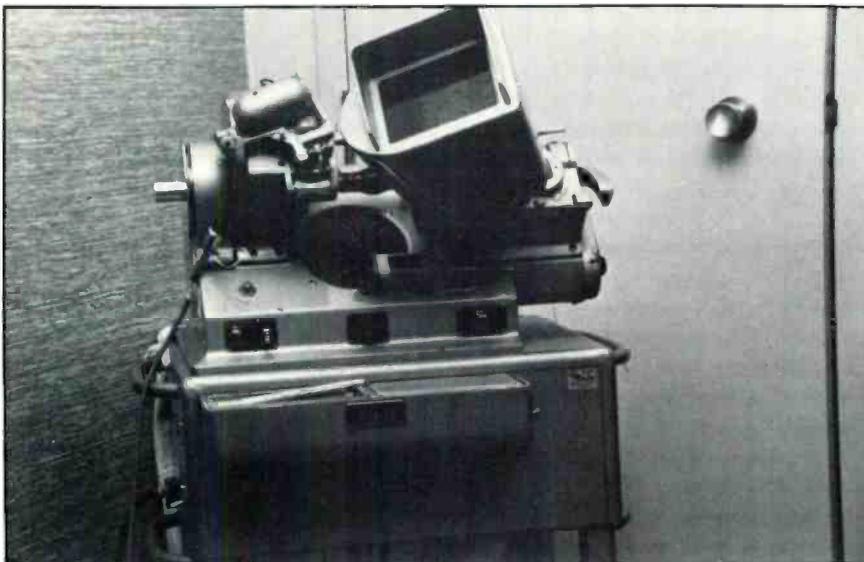


Photo 1. A moviola, used for editing 35mm mag strip film.

set (back-timed) one second. The best way to do this is to mark the 1/4-inch machine permanently at a one-second preroll point for both 7.5ips and 15ips, as this is the speed of most SFX libraries. Then just line up the one-second mark on the machine with the mark on the tape and fly the effect in.

Another technique is to store the SFX library on 1/2-inch 4-track (or 1/4-inch 2-track with center track time code) with effects on Tracks 1 through 3, leaving Track 4 for time code. Note: To avoid time code crosstalk onto Track 3, print the time code at -7VU instead of 0VU. With this technique, the effect will always be in hard lock.

A drawback to this is that your synchronizer setup may take time to accomplish a three-machine, or more, lock-up, and with deadlines, the trigger method might be quicker. An advantage to the 4-track lock-up method is that BGs (background SFX) can be laid in three at a time instead of one or two at a time with 1/4-inch machine. For example: You may have to lay in typewriters, phones and off-stage traffic effects that can't be married to each other. By using the 4-track technique, the BGs for a whole scene can be laid in simultaneously.

Many facilities are currently taking advantage of both methods of laying in effects and use multiple 1/4-inch and 1/2-inch machines. For quick access to effects, time code can be printed on the entire SFX tape, which then can be shuttled to a desired location within seconds. This is done by placing a *go-to* command into the synchronizer for that machine.

A common procedure, which is becoming a standard, is to print time code on Track 24 and a 60Hz neo-pilot tone on 23. This leaves Track 22 as a buffer track, which can also contain the production audio—for reference only. Printing a 60Hz tone on Track 23 aids the synchronizer in resolving to other formats and, in case of time code drop outs, it is available as a secondary guide track for jam sync.

One of the major complaints that film editors have with tape is that each of the 24 tracks are physically joined in sync with each other, creating a problem when only a single track needs to be offset. One solution for this problem would be to strip off the effects to a precoded roll of tape, then synchronize the tape with the desired offset and lay it into the mix as another slave unit.

Another solution is to use cart machines. Carts can be timed to be just about any length before the loop occurs. Common cart lengths are 40 seconds, 1.5 minutes and 2.5 minutes. Carts use a 1/4-inch lubed stock and can be recorded in stereo or mono. For effects that have a natural

repetition, such as crickets, a shorter cart will work sufficiently. Many electronic editing rooms have three cart machines and a wall full of carts.

Dialogue editing

In the realm of dialogue editing, it is common for the editor to use the 1/4-inch dailies and trigger in sections of dialogue that are phased against a 0 track. The 0 track is usually the audio track that has been edited by the picture editor. If the production audio sounds OK, then the editor will bounce the track to a dialogue track, but when a problem occurs, such as a transfer problem or a piece of dialogue clipped at the end of a sentence, the sound editor must go back to the dailies and recut the line into the dialogue track.

It is important for the dialogue editor to smooth out the dynamics and background ambiances in the production audio. This can be very time-consuming if each of the actors' lines were shot at different times of the day and/or in different environments. Since it is impossible to get rid of the environmental sounds entirely, the editor may well steal a couple seconds of background fill and lay it under the other actors' lines, creating a smooth transition between the dialogue of each actor.

Samplers

The second form of non-film editing involves the use of samplers. They can either be used by themselves as an on-line system or in conjunction with a tape setup. An editor using a sampler still needs to transfer his edits to a 24-track machine, so the product can be brought to the dubbing stage for final mixing.

A disadvantage of using RAM-based samplers is the lack of sampling time. A 4Mbyte sampler holds approximately 40 seconds of effects (10 seconds to each meg of RAM). That works great on effects that are short, such as door slams and gun shots, but when accessing effects that are long, such as several car moves, the sampler can dedicate itself only to one sound or subject matter at a time. In those instances, using a sampler in conjunction with tape machines makes a more versatile and powerful editing setup.

Patches

A patch consists of numerous sampled effects. A patch named Mustang might consist of samples such as: car-in, idle, car-out, car-start, brakes and doors. This is used to build the sound elements of the vehicle. The editor can audition any combination of sounds by downloading the necessary patch.

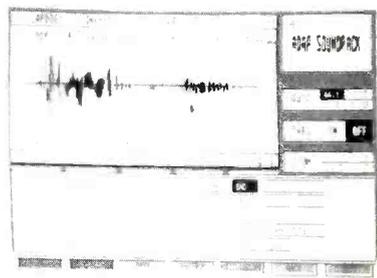
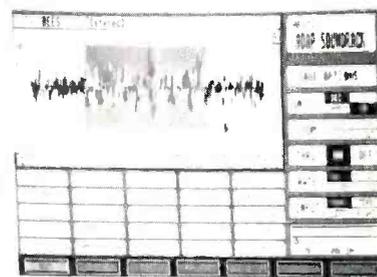
When using a musical keyboard, the editor can manually play-in the sound. For

example, to lay-in a car effect, the editor only has to press three keys: one key to start the car, a second to activate the idle (which is looped and can last indefinitely) and the third that starts the car-out. Each sample can be programmed to fade in and out, creating smooth transitions between each effect. Compared to editing tape, this process can take just a few minutes. With tape, you have to lay-in each effect on a separate track and smooth out the transitions, creating loops by checkerboarding from one track to another.

Sequencers

Sequencers in conjunction with samplers make a very powerful combination. Sequencing-in effects is the "Foley by fingers" style of laying in effects. The editor triggers the effect manually, using a keyboard (or any other MIDI controller). That information is recorded in a computer, including the volume and pitch changes that might have occurred during the "playing-in" of that effect. When you roll back the video, the computer will exactly reproduce all the moves that you played. (Note: The sequencer must be able to read all the formats of time code and to edit using those time code numbers.)

When sequencing-in the sample, the editor can contour the effect by using the pitch wheel, velocity control or other MIDI data controllers. An example is in the TV series "Beauty and the Beast," with pipe



Photos 2 and 3. Many samplers are offering visual representations of the samples to make cut-and-paste techniques more accurate.

effects constantly occurring in the background during the underground scenes. Velocity control aided the editor when laying-in sampled pipe effects, taking care not to clash with the dialogue. This was done by sequencing-in the effects to play louder in between lines and lower during the actual dialogue.

Changing the pitch also helped when trying not to clash with the tone of an actor's voice. For example, if the voice has a low-end presence with a whisper, the editor uses a high-end bright sound to cut through between lines, and then a duller, lower-pitched version of the effect when concerned about clashing with the actor's lines.

The same technique holds true for Foley-style effects, such as grabs, falls, and punches. The editor can often play-in the Foley effects manually, creating a more natural feel to the scene while saving time—relative to entering the time code locations for each effect. When laying-in Foley effects, it is important to have a large variety of sounds to simulate the random nature in which living creatures make real-time sounds. Samplers are helpful because of their audition features. A 4Mbyte patch can store 20 punches, a dozen body falls and various grabs on dirt, cement, wood, and grass. All of a sudden, you have a full fight-sequence library at your finger tips. To audition these effects, the download time can be 20 to 40 seconds. This is considerably faster than calling the transfer department or shuttling through several reels of effects—before finding the right combination of sounds.

Once the effect is played into the sequencer, the editor can fine-tune the sync of the performance by adjusting the start time of each effect, as each effect has its

own sequenced track. Most sequencers have between 40 and 200 tracks that the editor can preassemble and then transfer the edits to one track on the 24 track machine. This saves track space; instead of car moves taking up three tracks (car-in, idle, car-out,) they can be preassembled to one track. The dubbing mixer will thank you.

EDL

Another way of preassembling edits on a sampler is to use an edit decision list (EDL). This list lets the editor choose the desired effect and type into the computer terminal a start, end, duration and pitch number. The list can read time code from the video master, and at the specified mark, the EDL will trigger the sampled effect.

At that point, the assembled edits from the sequencer or EDL can be recorded on tape. This frees the RAM in the sampler to load another patch or EDL, such as tire squeaks and skids. This is the process of off-line editing. Each element or preassembled element has to be laid on to a different tape track.

When designing an effect for picture, the use of envelopes are handy to create wipe-ins and wipe-outs. A neat trick is to lay-in a gun sound at the natural pitch and then, on another track, lay-in the same sample pitched down—with a slower attack time. Then tell the mixer to pan the slower track slightly to one side. The effect is a gun with more bottom end that rings out and decays off to one side.

In the realm of sound designing, the sampler offers a quick, easy way to cut, paste, reverse and loop effects. Many samplers offer visual representations of the samples to make cut-and-paste tech-

niques more accurate. (See Photos 2 and 3.) This technique allows the creation of new, unique sounds that, if done on tape alone, would be quite time-consuming. The editor can listen to the completed edits while laying down new ones. When the editor can monitor all his tracks, it enables the effects to be balanced to the picture, as well as to each other.

Hard disk

The third form of non-film sound editing uses hard disk recording systems. This technology is still in its infancy, but it holds great promise for the future. Hard disk recording is the most similar to the moviola style of editing in that you place a prerecorded effect to time code.

Some of the newest workstation systems have DSP cards that allow reel-rocking techniques. Since effects are stored on hard disk and play directly from the disk, the editor has easy access to the full memory capacity of the hard disk.

Many systems can store between two and four hours of linear recording time. This gives the editor a well containing thousands of effects that can be called upon within seconds. (See Photo 4.) Each effect is assigned to a track output. For example, on the AMS AudioFile, when an effect is recorded, it is called a *cue*, and when it is placed on one of the 8-track outputs, it turns into an *event*. An event can be digitally manipulated without effecting the cue. Currently, most hard disk recording systems come with eight tracks, so for most projects, the hard disk recorder will have to download its edits to a multitrack to allow for more tracks to be digitally edited. The style of editing that most hard disk systems are supporting is the EDL.

Unfortunately, the hard disk recording system is not designed with as much precision for audio manipulation as the sampler. However, when both systems are combined, the editor can have a full, on-line editing system with all the versatility of digital editing.

Transferring from R-DAT or Sony F-1 to a hard disk system (through the use of the AES/EBU ports) is becoming possible. So, the effects will never have to leave the digital domain, and productions can be edited without discernible generation loss.

Most facilities are creating hybrids of the different technologies to meet the needs of their clients and budgets. As hardware, software, operating procedures and terminology standards are established, the popularity of the three non-film editing methods will probably replace the "industry standard" film-style editing techniques of the past.

RE/P



Photo 4. An audio-for-video-post suite at Soundelux in Hollywood. Notice that the AMS AudioFile is built into the mixing console.

Photos by Cindy L. Manekofsky

Sockhops to Woodstock:

An Interview with Bill Hanley

By Barry McKinnon

Bill Hanley, a pioneer in the field of large-scale sound reinforcement, including systems for the Beatles, the Fillmore and Woodstock, reflects on the early years of rock and roll.

Every industry has its pioneers, those individuals who took the first stab at a field no one else considered or wanted to try, and the sound reinforcement industry is no different. Bill Hanley has been in the mobile sound reinforcement business for more than 30 years and has been involved with major artists and events such as the Beatles and Woodstock. He began all of this when off-the-shelf products for large-scale sound reinforcement were not available, and innovation and adaptation were necessities.

Hanley's interest in audio began in high school, oddly enough, through his interest in roller skating. The roller rink where he spent so much time had an incredible organ sound system and great acoustics. In his early teens, he studied radio, television and electronics, which led to his interest in the hardware being used in the rink. The 400W sound system, combined with the energetic playing style of the organist, contributed to the enjoyment that the patrons of the rink felt. "I fell in love with big high-fidelity sound, on a big scale for concerts; I found a lot of joy and happiness in it," Hanley said.

In 1952, he started doing school record hops, and then in 1954, started Hanley Sound with his brother Terry. Bill and Terry did some work for the Boston Arts Festival and ended up buying a system from them. His next project was to get that system into the Newport Jazz Festival when the film "High Society" came out, as the festival became a bigger event after

being exposed to the mass audience of the film. Soon afterward, he purchased another system from an electrical engineer in New York. He had been mixing the live sound at Newport since 1960 and then took over the live recording in 1962-1963 to help alleviate the forest of microphones that had been growing on the stage.

The early 1960s

In the early 1960s, Hanley began providing the house sound for Madison Square Garden as well. "They were get-

ting a lot of complaints about bad audio, and we came in and fixed that," said Hanley. He had a lot of problems dealing with the IBEW local at that time because he felt that they didn't understand the show-business nature of this scale of sound work and were slow setting up and tearing down the systems. The promoters were getting mad at Hanley because his more-elaborate systems took longer to set up. "It took a long time to make sure there was a squawker pointed at every ear in the house.



Hanley considers his work on the Beatles' sound system to be a turning point in sound reinforcement. (Photo: The Bettman Archives)

Barry McKinnon is sales manager at BSE Production Equipment and Services Ltd., Calgary, Alberta.

"The road shows would come in and throw up a system on stage, and that would be it." The promoter was paying thousands of dollars in labor for the electricians, and Hanley was getting \$500 to \$800 per night. "They didn't have a great foreman, and I wasn't a great foreman either, so that was part of the problem. They just didn't understand the business." He continued for three or four years until he was putting so much gear in there, he was losing money. His drive to put up the best sound system possible, no matter what, led to some philosophical shocks.

The Beatles years

Bill was traveling with George Wiens' Jazz Festival, bringing high-fidelity sound to stadiums and fields in Chicago, Cincinnati and Cleveland, when his search for larger shows led to his involvement with the Beatles. "I got to the Beatles when electricity became an extension of the musician, I had learned about and put together an enormous amount of gear. I had four RCA 600W amps that came off of a battleship. They weighed two or three hundred pounds each and didn't have great high-frequency response. You had to drive the inputs with 100W, but it was not enough. When those boys came out on stage, it was absolute pandemonium: you couldn't hear the sound system, you couldn't hear yourself think, 46,000 teenage girls screaming at the top of their lungs, 120+dB ambient noise."

The Beatles' system consisted of 12 Altec 210 low-frequency enclosures and Altec 203B two-cell multicells with 290 drivers. The 288 drivers had better high-frequency response but "a 288 didn't last the night. We were using the 290s, even though the 4Ω voice coils were an inconvenience," said Hanley. The mixing console was a custom-built unit, using 16 channels of Langevin modules.

The Beatles marked a turning point in sound reinforcement in Hanley's view. "Sound changed from being high-fidelity (faithfulness of reproduction); since the Beatles, it's become a battle of levels. With the Beatles, the on-stage levels were getting higher and higher, and the supporting sound system had the grow just to keep up."

The psychedelic 1960s

Hanley was doing permanent gigs for three years at the Fillmore in San Francisco, from its opening in 1967, again using the Altec 210s and 203 and 1003 multicells with 290 drivers, set up with left and right stacks downstairs and a counter-weighted center cluster, with four of the 210s in it, hung 4 or 5 feet out in front of the proscenium. He used a custom 20-input console, again made up of Langevin

AM16 pre-amps and *featuring* EQ on every input. This fed the system through LA2A limiters into passive low-level crossovers and then into MacIntosh MI-200 amplifiers. "In 1965 or 1966, we began bi-amping because we started to use so much power that we blew up the capacitors in the passive networks. They'd begin making some real strange sounds, then—Bang!—nothing.

"A great deal of the joy and happiness that happens for people at concerts is in the intensity and the fidelity. I was one of the first guys to really go bananas with it and try to bring it outside to the masses." Hanley's views were shaken up by two experiences, the first being exposure to Grand Funk Railroad. They had a sound man out of Detroit who had assembled a system using Electro-Voice full-range speaker systems assembled into boxes, augmented by bass speakers.

"Grand Funk shows up at Randall Island, with this kid and his boxes and a pile of DC300 Crowns and associated stuff and sets up beside my system. We didn't 'A-B' them, he just set up and when the band came out he turned it on. Christ, was it loud—distorted, but loud. Here I was,

"I stood in the audience and listened to it and realized it was a valid experience—distortion!"

proud of my fidelity, but I stood in the audience and listened to it and realized it was a valid experience—distortion!

"It was valid for what the band wanted for their audience. Up to that time I had been successful in promoting fidelity, but who was I to superimpose my values on their music? I really had to scratch my head and wonder how much money to spend on fidelity. Here was this kid with his system full of \$28.50 EV horns, and piles of them, two or three hundred of them. He got heavy bass out of his bass cabinets and had incredibly efficient output, and it was a valid experience.

"The bands weren't hiring me, they were booting me out of jobs because it wasn't what they wanted. They were getting into this distortion thing; everybody was loving it. Grand Funk was the top group in the country at the time. If they didn't want to hire me, I had to alter my thinking.

"One of the other things that drove me bananas happened in Newport. I remember having a constant tone on the system. It was four or five 210s on each side with some two-cells and 10-cells for inside close, and I walked across the stage and

there was no sound. I walked a little further and there was sound again. It was going on and off with phase cancellation. You beat your head on getting everything flat and then you find out it depends where you stand whether you hear it or not. I will never forget that day when I walked across the stage. I knew about phase cancellation indoors, but I never thought about it in an outdoor situation."

Supporting the separation of church and state

Hanley was not doing concerts exclusively, however. In addition to college and university commencements, he provided sound for the presidential inauguration in 1968. In doing so, he introduced the Whitehouse to high-fidelity sound and Shure microphones.

"They had a whole bunch of those University spun-aluminum speakers with the 12-inch woofer and the tweeter in the center, up on scaffolding towers. They were awful. I put up six 210s, three on each side, and blew them away. They had taken pictures at Kennedy's inauguration, and the head sound man for the Catholic churches wrote the specs for the sound system from looking at the pictures. They had lights marked down as speakers and all kinds of things. They threw the Mormon Tabernacle Choir at me at the last minute. This was a great opportunity to bring in my 210s. We set up two systems and 'A-B'ed them, and never listened to his system again. That was the last time I did an inauguration; he never let me come back."

Hanley did not jump on new technology just because it was new. Other than equalization in the channel strips, Hanley had not yet found any pressing need for more elaborate equalization, such as the Altec Acousta-Voice process introduced by Don Davis. "I had talked to him a couple of times out in Anaheim, but it looked like something more for voice systems, not music. I wasn't having any feedback problems. I maximized the front-to-back ratios and tight-miced everything. I never got into EQs; I never learned to use them until later. Then I used them in ways not originally intended, more for artistic purposes." It's not surprising then that Hanley's favorite feedback story doesn't involve feedback at all.

"I was doing Satchmo's birthday party at the Sugarbowl. It was a rush job, and I had just built a new console, checked it out on the bench, and it looked OK. In the middle of the introductions, all of a sudden, it squealed like crazy. Here we are, in front of 20,000 people, and George Wiens is on stage, screaming at me, 'Don't you know what you're doing after all these years?!' He's standing on stage, all but call-

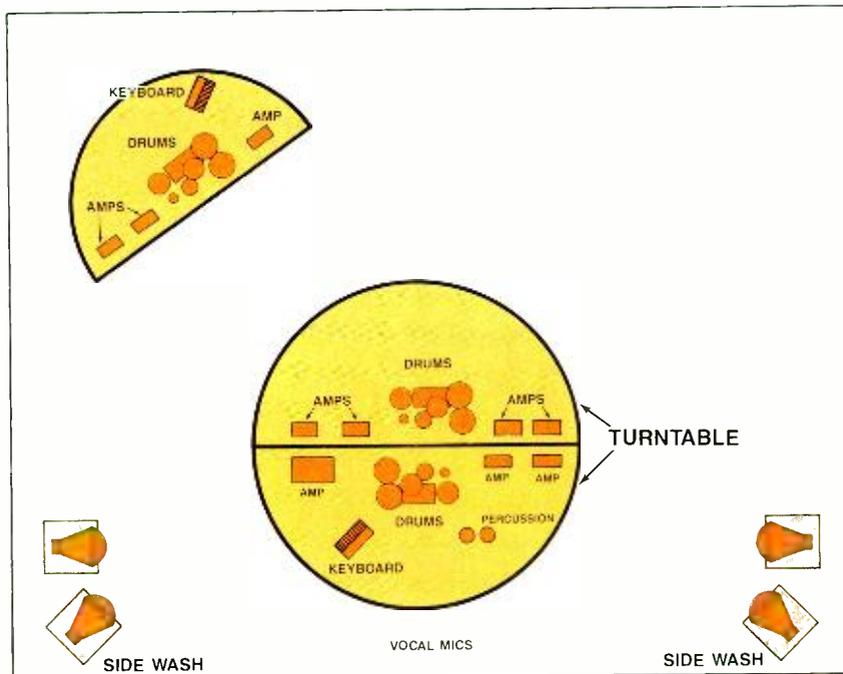


Figure 1. Woodstock stage (before the turntables broke).

ing me a f----- a----- in front of these people. I didn't know if there was a problem with the console I had never heard before. It sounded like it was going into oscillation: It wouldn't snap on, it kind of rose up in level. Here I am, banging on the console. Someone spotted this guy standing up by the mic box. Here was this jerk from a recording studio, plugging his signal generator into the mic inputs. Luckily, someone caught this guy. Was I embarrassed!"

Woodstock

Hanley's experience with large outdoor festivals such as the Miami, Atlanta and Dallas pop festivals and the Randall Island festival had established the reputation that led the promoters of Woodstock to him. Michael Lang, the executive producer, wanted a sound system with no screwups, regardless of cost. At that time, there were no precedents for a venue as large as the one contemplated on Yasgur's 700-acre field. After some inquiries, several sources recommended Hanley, because of his involvement in the other festivals.

Because of the scale of Woodstock, Hanley built custom speakers for the event. The low-frequency systems were similar to Altec 210s, but contained four Above those were boxes that each contained eight direct-radiator JBL 15-inch drivers. There were two of each system per side on the lower tier of scaffolding and four each per side on the highest tier. Multidriver 10-cell Altec multicells with 290 drivers provided highs.

"We had done a lot of tests and had a lot of failures with JBLs on the high frequencies, and they wanted a fortune to re-diaphragm them for us. We got into the 290s because they would stay together longer. Then we used 288s turned around with the back covers off as direct radiator HF because that's all there was. I wasn't into speaker design." The system was passively bi-amped and driven with a combination of MacIntosh M1200 and M1350 amplifiers and the then-new Crown DC300 for a total power of more than 10,000W.

Stage monitoring was done with a pair of side-wash monitors, each consisting of a JBL 4530 LF enclosure with a JBL driver, and an Altec 311-60 horn with 290E driver for highs. These were driven by DC300s.

The console that he had planned to use for live sound had to be diverted to recording at the last minute. Hanley had chartered a plane to bring a console up at the last minute. Typical of his consoles, they used Langevin pre-amps and EQ. The 20-input house sound console was augmented by Shure M67 mixers as required and fed the system through LA2A limiters. These limiters were also used on the vocal tracks that were laid down on the 8-track Scully recorder.

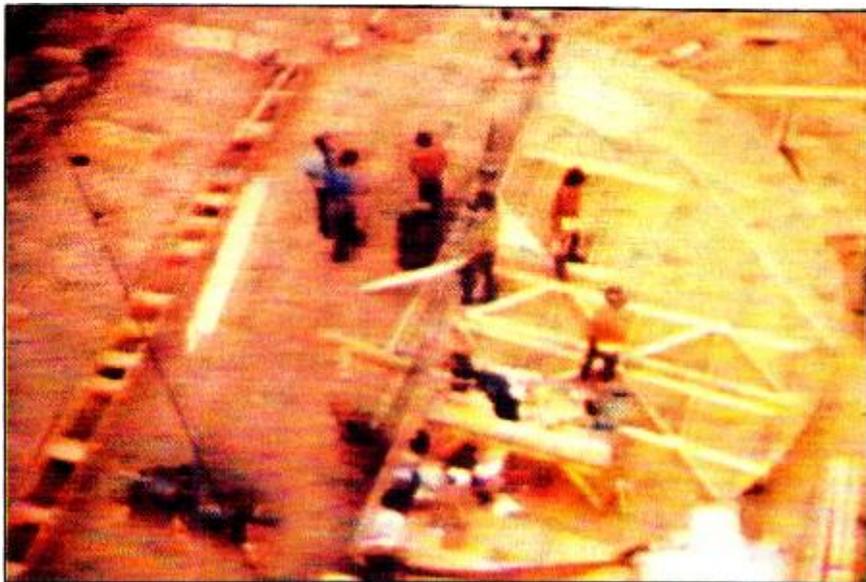
The stage featured a turntable to allow quick changeovers of the acts. Two 19-pair snakes, custom-built by Bill and Terry, were fed into a custom switchover box that allowed fast changeover at the console. "We were mixing blind. We were way out from the stage, and by the time some-

one got out to you through the audience with the microphone list, they were already playing." Hanley credits mix engineer Lee Osborne for much of the success in these difficult circumstances, citing his quick grasp of problems under pressure, and his knowledge and experience in live sound work.

Despite the well-publicized problems at Woodstock, the torrential rain and the financial problems of the promoters, it went well for Hanley. "The only things that didn't fail were the sound system, the water supply and the stage security. We'd have been OK if the turntables had stayed together. There were three half-moons that could be set up and hooked to the turntable. They used these cheap 8-inch casters, and the plates eventually tore out of the platform." Hanley had no equipment failures, having been prepared for the worst that Mother Nature and rock music could muster. "We'd done lots of outdoor shows; we were prepared. It was a matter of good planning.

"Woodstock didn't feel like a historic moment. I never thought about it in those terms from an intellectual, historical point of view. You're busy making everything work, making it sound good. It wasn't as dramatic for me as it was for everyone else, it was old hat. Being a technocrat as I am, I was shut out of the political and historical significance."

That's not to suggest that he was unmoved by the experience. "All those people, the joy of the people that were there, the idealism of all those people—it was a



Constructing the stage at Woodstock.

culmination of that kind of thing. A lot of high idealism was why I got into it in the first place."

The console Hanley had built was curved, with inputs in groups of 10. Multiple planes allowed the mix engineer to see and feel every knob. The work with this console led him to start work on an even-more-elaborate console. "Right after Woodstock, I had started to build a console in the round, a semicircular shape that had three planes, two two-band EQs per channel, and presets. No one's figured out yet that what's important in a sound reinforcement console is speed. Get everything into the audio engineer's field of vision; you've got to be able to touch and feel everything. This thing where it takes two or three guys to mix or you need roller skates is crazy." The console remains three-quarters built, a victim of the cash crunch Hanley encountered after Woodstock.

"After Woodstock, the festival markets dried up. I lost a quarter of a million bucks worth of business in three weeks, and I was already three or four hundred thousand dollars in debt." Competition was starting to heat up; other people like the Claire brothers were getting into the market. "I was believing my own bull---, I wasn't countering the competition."

The 1970s

Hanley was still doing touring work in 1970. He was involved in Festival Express, probably the only major rock tour to travel by train. "That was a lot of fun. It was a really great time." He recalls an incident with Peter, Paul and Mary: "They had a really sharp sound man, they had to mix their own thing and just send me a feed.

This was the same guy who did 'Jesus Christ Superstar.' They had to close the show for three weeks because the sound was so bad. One night it sounded terrible, and their manager, Albert Grossman, was screaming at me, 'None of my acts will ever use your system! I'll send letters! You'll never work again!' It was just luck that I had a Nagra on the limiter input from their feed that night, and it came to me distorted. Garbage in, garbage out. So, I'm saying, 'Wait a minute, wait a minute.'

"Back in those days at the festivals, you never really got a chance to do a sound check. The pressure was really intense. I guess that's what I liked about doing it live instead of in the studio."

Hanley has many memories of the developing concert sound business. His position as one of the earliest participants gives him a unique perspective. "We were the first people who were temporary sound reinforcement engineers. I had to learn the hard way that in dealing with live performance, I was dealing with sound reinforcement instead of sound reproduction. It's a different school of thought.

"There wasn't anyone else when I started. I started with six, eight, 10, and finally something like 30 Altec 210s in running inventory. That took some serious trucking." It took quite some time before Hanley was forced to think about the need to make money at it, and not do it just for satisfaction. "I got so much positive feedback from what I was doing that I never came to terms with the need to make money at it."

For Hanley, strong idealism often led him into conflicts with his employers. "When you take the laws of physics as

your ideals, you get overconfident. It's still the musicians that call the shots on who gets the jobs.

"Arthur Fiedler used to get mad at me because I wanted to put up lots of microphones, get as much direct sound as I could get, and then add reverb artificially. That way, everyone would hear what's going on in context. When you have a solo flute playing, you drop 30dB in level, and the people 300 feet away are now in the ambient noise. I used to think that was poor. The conductors at commencements would ask for one mic for a 35- or 40-piece orchestra, but now they often go along with multiple-microphone techniques."

In the early days of touring sound, the manufacturers did not look at it as a serious market. "I had a good relationship with MacIntosh, but other manufacturers weren't as cooperative. I had talked to JBL about getting into it, but they thought it was stupid." Now, of course, portable sound reinforcement markets likely make up a significant portion of professional audio sales. At that time, it wasn't seen as a growth market or a good test bed for new products. The needs and demands of touring sound were not yet a concern.

Current projects

Hanley's current project is an automated stage, a folding structure that would contain a stage prewired for mics, monitors and ac. It would include ac distribution, monitor power and light dimming equipment in it. In conjunction with that, lift systems to hoist sound systems into place, prewired and configured. "I've wanted to mechanize this for 13 or 14 years. It's insane: you hump all the stuff into and out of the truck; you do the same thing all the time. All these men doing all this work, two-day marathons to set up a stadium system. I just wanted to push buttons and make it happen." It's an ambitious project that has been consuming a lot of time and capital, but it gets closer all the time. "It would be nice to go to the store and buy one," Hanley said.

Bill Hanley, now 50, still does a few commencements, such as MIT's, and other interesting sound jobs, operating from Hanley Sound, in Peabody, MA. His brother Terry operates Terry Hanley Audio in Cambridge, MA.

The history of the concert sound reinforcement business has never been as well-documented as the movie sound business, but it is every bit as interesting, and, in some ways, more relevant to many of us, amplifying, as it did, the soundtrack of our lives.

REP

STUDIO UPDATE

Northeast

Hit Factory (New York) has purchased four ANT Telecommunications E-413/24 telcom noise reduction systems, providing the studio with 96 channels of noise reduction. 237 W. 54th St., New York, NY 10019; 212-664-1000; fax 212-246-2252.

Blank Productions (Stamford, CT) has purchased three Sony R-DAT machines. Other recent additions include a Roland D-550 with Opcode Librarian software, an MKS8 patcher and a 16-bit upgrade of the facility's Akai S-900 sampler. 1597 Hope St., Stamford, CT 06907; 203-968-2420; fax 203-329-7193.

Editel/NY Sound Suite (New York) has installed five Magnatech film recorder/reproducers for synchronization with Editel's 24-track system.

LRP Video (New York City) has appointed Marjorie Myers as account supervisor. 3

Dag Hammarskjold Plaza, New York, NY 10017; 212-759-0822.

Island Media Studios (W. Babylon, NY) has purchased two Valley 430 Dynamites and one Gatex from Manny's Music. 841 Sunrise Highway, W. Babylon, NY 11704; 516-669-1872.

Eastern Standard Productions (Buffalo, NY) has expanded its real-time duplication and tape loading facility. Duplication production has been expanded by a factor of more than two, and loading capability has been tripled. Direct on-cassette printing has been added, allowing cassettes to be produced with a variety of colors and foils. New real-time equipment includes Denon DRM-44HX units, modified by ESP with a proprietary record system to improve recording accuracy. 26 Baxter St., Buffalo, NY 14207; 716-876-1454.

Forge Recording Studios (Malvern, PA) has added Sony's line of pre-mastering and digital editing equipment, including the

DAE-3000 editor, PCM 1630 processor, two DMR 4000 recorders, DTA-2000 digital tape analyzer, DAL-1000 digital limiter and a PCM-2500 R-DAT recorder. According to the studio, the additions make it the Philadelphia area's only recording and CD mastering studio. 119 Great Valley Parkway, Malvern, PA, 19355; 215-935-1422.

Times Square Studios (New York) has named Jorge Silva chief engineer. He will also continue his current responsibilities as director of audio. 1481 Broadway, New York, NY 10036.

New York Audio Productions (New York) has completed a three-month renovation with the installation of a 16-input TAC Scorpion console. The addition will enlarge the studio's music production and voice-over capabilities. Other new equipment includes Drawmer DS 201 gates and Yamaha Q 2031 equalizers. 140 W. 22nd St., New York, NY 10011; 212-243-6826.



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STUDIO UPDATE

Sound Logic Studios (Silver Spring, MD) is the Washington, DC, area's newest recording studio. Designed by John Wesley Gardner, the facility has 15'x18' and 10'x7' studios with variable acoustics and a 15'x25' control room. The console is a 32x8x24 Soundcraft Series 600 console. Tape machines are a Sony/MCI JH 24 24-track, and Studer A810 and Teac 35-2 2-tracks. The heart of the studio is a Kurzweil K250 interfaced with a SMPTE-locked Apple Macintosh Plus running Mark of the Unicorn Performer software. A variety of outboard gear is available. Hank Bartlett is the owner; studio manager is Mark Greenhouse. 3209 Birchtree Lane, Silver Spring, MD 20906; 301-871-0200.

Roar Productions and Musical Services (Columbia, MD) has added Larry Adler to the staff. During the fall, the facility offered special seminars in live sound, video production techniques, MIDI and careers as a voice-over talent. In conjunction with the Gerard Co., Roar was a finalist for its first entry in the Telly Awards competition. The entry was a TV PSA for the Federal Aviation Administration titled "Discover Today's FAA." 6655-H Dobbin Road, Columbia, MD 21045; 301-596-2600.

Southeast

Woodland Sound Studios (Nashville) has made several additions to the staff. Katie Dungan is studio manager. Barbara Smith is assistant manager. Suzi Ragsdale has been appointed secretary/receptionist. Tim Farmer continues as chief engineer, and is also acting in a consulting management capacity. 1011 Woodland St., Nashville, TN 37206; 615-227-5027.

Reelsound Recording Co. East (Nashville) is a 24-track remote unit housed in a 1948 Flixible road bus. The unit is equipped with an MCI 428LM console and a JH 24 24-track recorder. Outboard gear includes units from Valley People, dbx, AMS, Yamaha, Teletronix, UREI and RTS. Producer Dave Perkins manages the bus for studio and live projects. 1701 Green Hills Drive, Nashville, TN 37215; 615-385-0220.

Midwest

Covideo, a subsidiary of Anthony M. Franco, Inc. (Detroit) has been selected as a finalist in the "Guidance and Counseling"

category at the 1988 International Film & TV Festival of New York, for its documentary production of "Wasted Dreams." The documentary deals with six Detroit-area young people who have been injured by gunshots. 400 Renaissance Center, Suite 660, Detroit, MI 48243; 313-567-2985.

Minnesota Public Radio (St. Paul, MN) recently recorded the 200th show of "St. Paul Sunday Morning" in the facility's Studio M. The show was broadcast on Christmas Day. 45 E. Seventh St., St. Paul, MN 55101; 612-290-1500; fax 612-290-1243.

TRC Studios (Indianapolis) has upgraded Studio B to a fully functional film-video mix room. New equipment includes an MCI JH110 1-inch video layback machine with Dolby SR; a center-track time code update for the JH110B; a Wide Range Electronics 16/35mm mag transfer machine; an Adams-Smith 2600 synchronizer with remote control; an automation upgrade to its Digital Creations Diskmix automation; a Valley People rack with two Gainrain IIs, two MaxiQs, four Kapex IIs, and two de-essers; a Klark-Teknik DN780; a TC Electronic 2290 sampler and 2240 stereo parametric EQ; and the Sound Ideas compact disc sound effects library. 5761 Park Plaza Court, Indianapolis, IN 46220; 317-845-1980.

DRC Studio (Springfield, IL) has expanded its inventory. Additions include the Tascam MS-16 recorder with dbx noise reduction, MegaMix automation, Atari Mega ST2 computer, Yamaha REV-5, dbx 160X compressor/limiters, BBE 822 Sonic Maximizer, AKG 414 B-ULS microphones, Sound Ideas sound effects on CD series 1000 and 2000, Simmons SDS1000 MIDI drum kit and an E-mu Systems EIII with more than 200Mbytes of sounds on hard disk. 2416 S. Walnut St., Springfield, IL 62704; 217-753-0409.

Paisley Park Studios (Minneapolis) hosted the first Sound Stage 88 in early October, featuring two days of music, pro-audio and video equipment exhibits, seminars and instruction. The studio's 12,000-square-foot sound stage housed representatives from more than 70 exhibitors, including Lexicon, Grass Valley, 3M and Ampex. Paisley's three studios were used for instruction and public viewing. About 2,000 people attended, and plans are underway to host the event in 1989.

Sean McMahon, engineer at **Smith/Lee Productions** (St. Louis) has recently purchased new outboard equipment, including an Eventide H-3000 Ultra-Harmonizer, Lexicon LXP-1 digital effects processor, Lexicon MRC MIDI reverb controller, a software update for a Yamaha SPX-90, and a Valley International PR-2 mainframe with two Commander modules.

Ron Rose Productions (Royal Oak, MI) has named Anita Lanning as studio manager. 25885 York Road, Royal Oak, MI 48070; 313-545-1696.

Mountain

Coupe Studios (Boulder, CO) has upgraded to 24 tracks with the installation of an Otari MX-80 recorder. The facility specializes in radio and television broadcast commercials and album production. 2888 Bluff St., Suite 115, Boulder, CO 80301; 303-447-0551.

Southwest

Production Masters (Phoenix, AZ) has named Bruce E. Reid as general manager. 834 N. Seventh Ave., Phoenix, AZ 85007; 602-254-1600; fax 602-495-9949.

Southern California

Todd-AO Glen Glenn Studios (Hollywood) has entered an agreement with Comlogic to supply six custom designed Neotek Essence consoles.

Northern California

The Saul Zaentz Film Center (Berkeley) has acquired an 8Mbyte version of E-mu Systems' Emulator Three digital sound production system. The system was purchased from Audio Images Corp., San Francisco.

Northwest

Spectrum Sound Studios (Portland, OR) has added several staff members and expanded its equipment. New to the staff is Dick Starr, system design sales representative, Matthew Tonjes, maintenance technician, and Kellie Hager, part-time assistant. Spectrum's SSL console has been expanded to 40 channels, and a Total Recall computer has been added. Apogee

STUDIO UPDATE

filters have been added to its Mitsubishi X-86 recorder. Also purchased were TimeLine Lynx synchronizers and a Yamaha G7 grand piano. 1634 SW Alder St., Portland, OR 97205; 503-248-0248.

Hawaii

Audio Resource Honolulu (Honolulu) has opened just outside Waikiki. The 24-track facility contains an automated Harrison MR-4 console, Sony/MCI tape machines, a Macintosh-based MIDI/SMPTE sequencing system, Westlake and Yamaha monitors, and out-board gear by Eventide, UREI, Orban, Pultec, Drawmer and Yamaha. The studio offers music recording and audio-for-video post-production. Tony Hugar is the studio manager, and Milan Bertosa is the chief engineer. Audio Resource is affiliated with The Audio Lock Up, a Chicago facility. Century Center, 1750 Kalakaua, Honolulu, HI 96826; 808-944-9400.

Canada

Comfort Sound (Toronto) has renovated its mobile unit to improve monitoring. Changes have also been occurring in the studio, including the installation of a wall in the recording room, which created a more intimate voice-over area and facilitated better isolation for music bed-track sessions. An Eventide Ultra-Harmonizer has also been purchased. 26 Soho St., Suite 390, Toronto, Ontario, Canada M5T 1Z7; 416-593-7992.

Metalworks Studios (Mississauga, Ontario) has opened a second studio, SongLab. The facility is a MIDI room with up to 48-track capability. 3611 Mavis Road, Unit 5, Mississauga, Ontario, Canada L5C 1T7; 416-279-4008.

Pathe Sound and Post-Production Centre (Toronto) has named Ted Rouse as vice president and general manager. The facility has also added a Magnatch High Speed Dubber System and a THX-Lucas Monitoring System.

Seacoast Sound (Victoria, British Columbia) has appointed Geoffrey Bate as general manager. 825 Broughton St., Victoria, British Columbia, Canada V8W 1E5; 604-386-1131; fax 604-386-5775.

Manufacturer announcements

Tascam has sold a DAT recorder/player to Showco Sound, Carrollton, TX; and a T2640 recording duplicator to Hummingbird Recordings, Melbourne, FL.

Amek has sold a G2520 master recording console to The Church, London, owned by Dave Stewart and Annie Lennox of the Eurythmics.

New England Digital has sold a 32Mbyte, 64-voice Synclavier to Flyte Tyme, Minneapolis, owned by Jimmy Jam and Terry Lewis.

Otari has announced the following tape machines sales: Dave Stewart, two MTR-90s for his personal-use studio in Southern California; George Tobin Studios, North Hollywood, DTR-900; Design FX Audio, Los Angeles, MTR-90; Sunset Sound Factory, Hollywood, DTR-900; LA Studios, Los Angeles, 12 MTR-12 2- and 4-tracks; Front Page Recorders, Costa Mesa, CA, DTR-900; Turner Broadcasting, MTR-12CT; Curtis Mayfield, an MX-80 for his personal use studio; Edit Works, MX-55TM, the first unit delivered in the U.S.; NBC Sports, New York, 16 MKIII 8-track ½-inch machines, used at the Summer Olympics; Howard Schwartz Recording, New York, two MTR-90IIs; Island Media Services, West Babylon, NY, MX-80; If Walls Could Talk Studio, North Caldwell, NJ, MX-80; composer/musician Bob Telson, MX-80; guitarist Jimmy Ryan, MX-80; Manhattan Center Studios, New York, DTR-900; Superdupe Creations, MX-80; Wild Twin Recording, New York, MTR-90II; Backer and Spielvogel Advertising, two MTR-10s; Nutmeg Music, New York; MX-80; and the Hit Factory, New York, two MTR-90s.

Alpha Audio has sold 2,000 square feet of Sonex acoustical foam to Will Vinton Studios, Portland, OR.

Sony has sold a PCM-3324 recorder to Professional Media Services, Gainesville, FL, owned by Grammy-winning engineer Mark Pinske. It is the first 3324 in Florida.

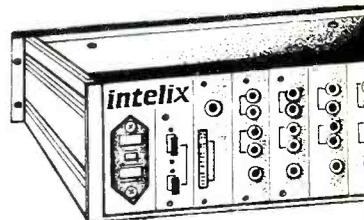
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THE CUTTING EDGE

By Laurel Cash

Advances in Tape Machine Technology

Otari announces first of B series digital multitracks shipped

Otari shipped the multiple units of its new DTR-900B series digital multitrack recorder last month. According to John Carey, marketing manager for Otari Corporation, "The B series includes all of the changes developed since our earliest machines were delivered and offers the user higher performance, greater reliability and an enhanced ease of use." The new DTR-900B features entirely redesigned Auto-Locator/Remote software and hardware, new proprietary VLSI technology, in-house manufactured heads, and upgraded power supplies to accommodate the use of optional Apogee Electronics low-pass filters in the A/D and D/A sections. Otari has also announced the availability of two new accessories for its DTR-900 machines: the EC-104 plug-in chase synchronizer module and the CB-503 PD (PRODIGI)-to-DASH format converter. The format converter allows bidirectional digital transfers between the DTR-900 and any DASH multitrack machine. No price increase is expected for the new model, and most of the new features of the B series can be retrofitted to the earlier models of the DTR-900 machines, according to Carey.

Circle (170) on Rapid Facts Card

Mitsubishi introduces X-880 digital multitrack recorder

Mitsubishi Professional Audio has announced the new X-880 digital multitrack recorder, an upgraded version of the X-850. Circuitry enhancements include a new, convenient pull-down front panel that provides easy access to audio monitor, ping-pong jacks, emphasis/de-emphasis switches and status switches. The design also adds a comprehensive system status display on the front panel. This display provides visual confirmation of external sync, sampling frequency, playback servo and other system status requirements.

Other circuitry improvements include newly designed linear-phase active analog filters, which are reported to improve the sound quality substantially. These are

Laurel Cash is RE/P's executive consultant and a free-lance writer based in Los Angeles.

Mitsubishi-designed, Murata-manufactured and are said to be similar to those of Apogee Electronics. Mitsubishi has elected to use Burr-Brown monolithic A/D and D/A converters. This is also said to increase reliability and further improve the sound quality.

The company has also redesigned the auxiliary analog track circuitry for a higher S/N ratio and lower distortion. A master safe switch has been added to prevent accidental erasure during mixing sessions. The transport enhancements include faster winding modes and a motor driver that has been redesigned both electronically and mechanically for higher performance and reliability.

The X-880 comes prewired for the optional plug-in CS-1 chase synchronizer. The CS-1 resolves to a digital sample frequency that is equivalent to 20s. According to Mitsubishi, absolute phase-accurate 64-track lock-up or intermachine editing is assured when using the CS-1. It is claimed that when the transport is under synchronizer control, a varispeed resolution to 0.01% eliminates audible pitch changes. Also, redesigned transport control circuitry includes switches and trim pots, allowing you to optimize the transport for 7-inch, 10.5-inch and 14-inch reels.

The X-880 is available now, and has a suggested retail price of \$180,000.

Circle (171) on Rapid Facts Card

Sony introduces analog 24-track

Sony Professional Audio has introduced the APR-24 multitrack recorder, which is said to be available for shipment immediately. It is reported that many of the APR-24's features, which are included as standard equipment, are optional accessories on machines available from other manufacturers. Some of the features that come standard with this machine include an on-board time code generator, reader and synchronizer. As a result, the remote unit provides audio, transport, locator and synchronizer control in a single package requiring only one connecting cable.

The synchronization facilities allow you to synchronize to longitudinal time code and VITC in various formats. You can resolve to time code, tone (59.94Hz or

60Hz) and house sync, "burst" time code output during fast-wind modes and "bit bump" with subframe offset accuracy.

On the remote control are edit keys that allow the actuation of rehearsed and externally programmed "punch in" and "punch out" operations. Sony is said to be targeting the APR-24 to video post-production houses, film audio production and music recording studios. As of press time, the suggested retail price is reported to be \$45,500.

Circle (172) on Rapid Facts Card

48-channel DASH recorder launched by Sony

The AES convention in November was the first glimpse most of the industry had of Sony's new 48-channel, DASH-format digital multitrack recorder, the PCM-3348. This unit is reported to be upwardly and downwardly compatible with Sony's PCM-3324 and 3324A. It is said that you can record on channels 1 through 24 on the PCM-3348 and play that master back on a 3324/3324A. 96kHz oversampled electronics are featured on the inputs and outputs. Other features include newly designed digital filtering, real-time ping-pong (with no time delay) and variable crossfade times.

The advanced digital output allows for up to 250 words of digital audio (about



Mitsubishi X-880 digital 32-track.

5ms) to be output from the recorder (before the analog output) with one-word resolution, allowing for compensation through other digital devices, thereby assuring absolute phase on the tape. The PCM-3348 also has 20 seconds of solid-state RAM on board with 16-bit resolution. This allows you to store 20 seconds of audio from the tape into RAM and reinsert the RAM data elsewhere on tape. It can be triggered externally either by MIDI, by a gate or manually from the Auto-locator. You can select any two channels to be AES/EBU I/Os. There are a remote channel arming interface and a remote 48-channel multifunction meter system. As of press time, the suggested retail price is reported to be \$240,000, and the PCM-3348 should be available for shipment this month.

Circle (173) on Rapid Facts Card

Studer previews new multitrack at AES

Studer Revox previewed its new multitrack recorder, the Studer A827, at

AES in November. This is the first multitrack addition to its A820 line since the A820-24 and is based on the A820 transport.

Like its top-of-the-line brother, the A820-24, the A827 offers many of the features possible with microprocessor technology. Some of its more notable features include a 14-inch reel capability, three tape speeds with integrated varispeed controller, two tape types storable for each speed, phase-compensated MDAC-controlled amplifiers with switchable Dolby HX Pro and an optional internal synchronizer. Also included are parallel and serial RS-232/422 control ports for easy system integration.

Priced beneath the A820-24, the A827 is said to incorporate the latest technology and efficient production techniques to reduce construction costs. Formal introduction of this product is planned for the AES in Hamburg, with the first deliveries expected in May 1989.

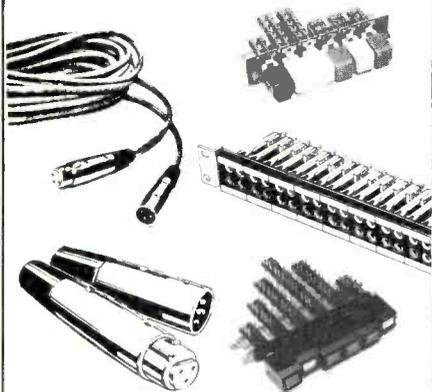
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Sony APR-24 analog 24-track.

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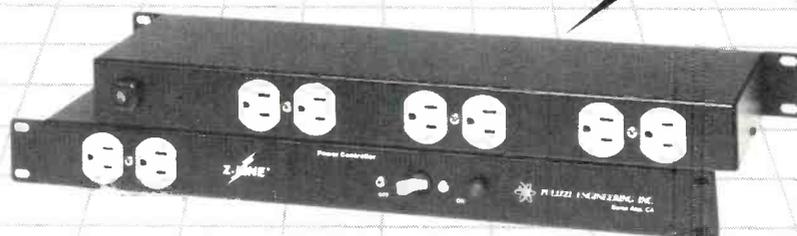
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Circle (30) on Rapid Facts Card

Continued from page 45

The 2-inch remix master track-format was:

Track 3—left audience to be inserted later

Track 4—right audience to be inserted later

Track 5—next pair left audience to be inserted later

Track 6—next pair right audience to be inserted later

Track 7—left stereo music remix

Track 8—right stereo music remix

Track 9—mono music remix

Track 17—P-1 production microphone

Track 18—P-2 production microphone

Track 19—P-3 production microphone

Track 23—timecode (copied from live original 24-track)

Track 24—timecode (reshaped and copied from live original 24-track)

The three production microphones were included to support the sweetening process. If a dialogue part needed to be pulled up over the applause, it could be done at the very last minute. We never recorded the applause and effects microphones on the original live-music multitrack, so it was difficult to judge the proper ratio of

dialogue to the applause sound. Basically, David Palmer had to concentrate on making the music remix as pretty as possible. A normal axiom for mixing TV sound is if you can see it, you had better be able to hear it—loud! David did not subscribe to that philosophy. He normally mixed albums, not TV shows.

The show was mixed in the Fanta truck, which has a permanent setup for mixing to picture. Since we recorded there, we thought there would be less of a learning curve when mixing. Also, David could concentrate on the job, but still be near enough to the Fanta studio crew to get help, if needed. Instead of making the mix jump up and down to match the picture, we discovered that the smooth playing allowed subtle changes in the mix. If you saw a guitar player soloing, you could hear the solo as well as the other music that fit around it.

Sweetening

Sweetening this show was in the hands of Mark Repp and editor Tom Edwards at TNN. To put the final version of the show together, they synchronized the 24-track remixed master with the final video edit of the show. They also synced up the

original audience-only 24-track tape. As they watched the video, they mixed the appropriate audience sound on to the 24-track remix master. When they were finished, they had a 24-track remix master that had the proper music remix, the correct applause and audience sounds, and any other special elements that were necessary to complete the audio (possibly an announcer or special effect).

The 24-track remix master, which now had every song and every piece of dialogue, was then conformed or transferred to the edited master videotape. The audio was recorded, Dolby encoded, on tracks 1 and 2 of the edited 1-inch video master tape.

Some aspects of this show were special. The editor was consulted from the first pre-production meeting to the final cuts. The music remix engineer was involved from the first rehearsal to the final sweetening. And all the team members were truly dedicated and honored to be working on "A session with Chet Atkins, C.G.P."

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Circle (35) on Rapid Facts Card

As reported in the October issue, the recently completed AES Convention in Los Angeles was a bonanza of new products. In compiling the show preview, many companies gave us advance information on their product introductions. Product Preview showcased these products. Other companies, however, chose to wait until the show opened to announce their new products. This month's "New Products" is expanded and contains products introduced at the show but not included in October's "Product Preview." More products will be featured in the February issue.

AMS/Calrec Edit 1 mixer

The Edit 1 mixer uses digital signal processing and consists primarily of eight input faders plus two group or output faders. There are four outputs, two stereo or two double mono; it is possible to mix the two output faders together. Any reasonable number of inputs can be accommodated because the input matrix system eliminates the need for a patchfield. The mixer also features four assignable Logicator (rotary) controls, each with an arc of LEDs projected through light-guides, and an associated four-character alphanumeric display showing the function of the control.

Circle (100) on Rapid Facts Card

Tascam DA 800/24

Tascam's first DASH machine incorporates proprietary ZD circuits in the opto-isolated A/D, D/A converters, along with 2x oversampling in the record and playback. A special pinch-roller drive system provides precise tape-handling and stability. With two digital format I/Os for S DIF-2 PCM equipment and AES/EBU systems, the machine allows digital transfers to various digital devices. The meter bridge also accommodates remote installation, and a second meter bridge may be added for remote use. Suggested retail price is \$99,000.

Circle (101) on Rapid Facts Card



Mitsubishi CS-1 chase synchronizer

Designed to facilitate Mitsubishi 32-tracks in sound-for-image projects, the CS-1 allows two machines to be locked up for 64-track capability. Resolution is accurate within the digital audio "word" (± 20 microseconds) with two X-880s. It can be interfaced to VCRs, VTRs and ATRs, and can also be programmed with other master machine parameters. The unit consists of a plug-in circuit card for the new X-880 transport and a remote control; an optional card rack allows the CS-1 to be used with the X-800 and X-850.

Circle (103) on Rapid Facts Card

Lexicon MRC MIDI remote controller

The MRC is an intelligent MIDI remote designed to be used with Lexicon's LXP-1 and PCM 70 effects processors. With the LXP-1, the MRC takes uses beyond the front panel to six "hidden" parameters in each of the LXP's 16 programs. The parameters can be altered in real time, and personal setups can be stored in memory. With the PCM 70, the MRC's faders can be used for real-time MIDI System Exclusive control of 12 key parameters for each algorithm, allowing users to tailor factory presets or user registers. The unit can also be used with Yamaha's DX-7s, TX-816 or TX-802 as a patch modifier.

Circle (104) on Rapid Facts Card



NED Synclavier 9600

New England Digital's largest system, the 9600 features 96Mbytes of RAM, 96 polyphonic stereo voices, nearly 3Gbytes of on-line sound storage, SMPTE and VITC synchronization, and a 76-note velocity/pressure keyboard and button control panel. The system supports multiple user interfaces, such as the customized Macintosh II graphics workstation, the keyboard

and button control panel, the Digital Guitar interface, a remote controller/editor/locator, or any other MIDI device. Open architecture allows for future expansion and upgrades.

Circle (102) on Rapid Facts Card

Sanken CU-44X mic

Available from Audio Intervisual Design, the Sanken CU-44X is the first transformerless model that features dual-capsule condenser design in a mono microphone. The cardioid pattern uses 1-micron titanium diaphragms, which are corrosion-free and impervious to temperature and humidity changes. The dual-capsule design provides handling of both low and high frequencies, 20Hz to 20kHz within +1dB. Self-noise is inaudible (18dB or less) and maximum SPL is 140dB.

Circle (105) on Rapid Facts Card

Soundcraft MIDI computer

For use with the Soundcraft 6000 console, the computer is a mute control system with a non-volatile RAM memory that stores up to eight songs, each containing 100 patches of complete mute settings. By using the keypad and 10 numeric buttons, each song and patch can be individually named. The supertwist backlit LCD allows information to be viewed from a variety of angles and in varied lighting conditions. When used in a MIDI studio, the computer records mute settings as they are made, at the correct moment within the sequence.

Circle (106) on Rapid Facts Card

Turbosound TXD-520

For small venues or areas that cannot be directly reached by the main loudspeakers, the TXD-520 is designed to handle high power and provide coherent sound over a side area. Universal mounting and fixing hardware allow for easy installation and enclosure orientation. Components used are one 10-inch drive and one soft dome tweeter. Frequency response is 100Hz to 18kHz, and average dispersion is $90^\circ \times 60^\circ$.

Circle (107) on Rapid Facts Card

dbx RT-60 option chip set

An option for the RTA-1 analyzer, version 2.0 of the RT-60 option chip set allows measuring the reverberation time of a space in each of the RTA-1's 31 bands. In addition, it provides enhanced room-response curve capabilities, and adds a

NEW PRODUCTS

microphone calibration with database for 10 microphone sensitivity memories, absolute SPL measurements capabilities and a custom printout feature. In the room response curve mode, 15- and 30-second functions have been added. Retail price for version 2.0 is \$499.

Circle (108) on Rapid Facts Card

Tascam ATR-80/32 multitrack

The 32-track, 2-inch machine accommodates 14-inch reels and has reel and PPL capstan motors that use Samarium Cobalt, which reduces mass and produces higher torque than in conventional motors. The result is a fast-wind speed of up to 280ips, more responsiveness and more reliable lockup. A rehearse feature enables users to preview edits without affecting the master. Also included is a backup memory for tape time, pitch control and amplifier mode settings. Retail price is \$45,000.

Circle (109) on Rapid Facts Card

Aphex 612 expander/gate

In addition to gating functions, the 612 also features downward expansion with a variable ratio, allowing a wide variety of dynamics control. The unit is also a "ducker," allowing a key input to lower the level of the audio input. The 612 is the first product to incorporate the Aphex VCA 1001. Suggested retail price is \$795.

Circle (110) on Rapid Facts Card

DAR SoundStation II configurations

The two- and eight-channel configurations of the SoundStation II feature stereo time-warp, fully animated playback display, punch-in record, long crossfades, and full chase synchronization. Combined with SoundStation II's sound segment cutting and trimming, edit sliding, track slipping and reel-rocking editing, these features make the systems the most powerful multichannel digital audio sound editing and processing systems available, the company says.

Circle (111) on Rapid Facts Card

CTI/dbx A/D chip set

The F410/D20C10/A1520 chip set, an 18/19/20-bit analog-to-digital converter, uses noise-shaped oversampling at 6MHz and flash four-bit conversion. A two-stage digital filter/decimator reduces the sampling rate to 48kHz, increases resolution to 20 bits and eliminates high-frequency

noise. The basic configuration of the A/D converter delivers greater than 105dB signal-to-noise ratio; less than 0.005% THD at maximum input, 0.01% maximum THD (untrimmed) at -20dB and -40dB input; and 0.00000076% differential linearity. Price is \$130 in quantities of 100.

Circle (112) on Rapid Facts Card

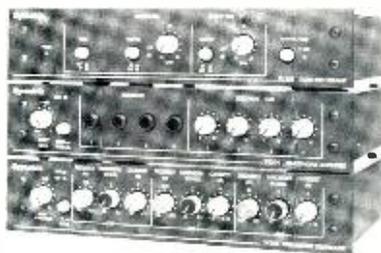
Symetrix "Half-Rack" SX200 series

All of the products in the SX200 Series feature high headroom, balanced inputs, low noise, wide dynamic range, low distortion, studio-quality circuitry, and low-impedance, high-current balanced and unbalanced output line drivers. The SX201 parametric EQ/pre-amp includes +15dB boost and -30dB notch filter capability, with unbalanced pre-amp input, balanced/ unbalanced line-level input and balanced line driver output. Retail price is \$239.

The SX202 dual microphone pre-amplifier features two microphone pre-amplifiers with variable gain, 15dB pad, +48V phantom power, and left, right, and left + right outputs. Retail price is \$219.

The SX204 headphone amplifier includes a one-in, four-out amplifier with proprietary high-voltage converter technology to drive high-impedance headphones like a big power amp, while providing ample power for low-impedance phones. Retail price is \$269.

Circle (119) on Rapid Facts Card



Court Signature Series loudspeakers

The soft dome studio monitor loudspeakers include three models. All feature a 1"×8" homopolymer mid and 1"×1" ferro fluid soft dome HF. In addition, the 150W SN20 features 40Hz-20kHz ±3dB frequency response, two-way passive crossover. The 250W SN30 includes a 1"×15" roll surround LF, 30Hz-20kHz ±3dB frequency response and a three-way passive crossover. The 400W SN60 features a 2"×15" roll sur-

round LF, 25Hz-20kHz ±3dB frequency response and a three-way passive/active crossover.

Circle (113) on Rapid Facts Card

Soundtracs In Line series

This console series, initially available in 36- and 48-input mainframes, is now available in a 32-bus format. A number of options, including stereo inputs and effects return modules, are available. Analog specification matches or exceeds the ERIC console without the digital control.

Circle (114) on Rapid Facts Card

Steinberg tape controller TC 1

The TC 1 can remotely control all functions of the Fostex A and B series recorders, except the built-in auto-locator. It positions the tape with an accuracy of ½-frame for up to 255 programmable punch-in/out points. The control time code can be either SMPTE or MIDI; switching between the two is automatic.

Circle (115) on Rapid Facts Card

FM Acoustics Forcelines

Forcelines are high-energy transfer cables designed for lowest loss of wide-band power transfer between amplifiers and loudspeakers. They feature AWG 5 size, are rated for a current of 200A rms continuous and 1,200A peak, and have a minimal resistance of 1.9Ω/km throughout the audio band. They are recommended for smaller monitors and close-field monitor systems, and for wiring mid- and high-frequency drivers in multiway systems.

Circle (116) on Rapid Facts Card

ProCo Patchmaster patchbay

The Patchmaster Series model PM-148 is a single-space, 48-point, unbalanced patchbay designed for use in recording studios, A/V production facilities, commercial sound installations and portable audio systems. The Selectapatch Switch provides easy user modification of the signal flow. Setting the switch determines whether a set of jacks is full-normalled; half-normalled; parallel or open; and no soldering or change in wiring is necessary. Retail prices is \$375.

Circle (117) on Rapid Facts Card

Soundtracs FM range console

This broadcast version of the FM range can be configured to double as a small pro-

duction console for program editing. Available with mono, stereo and telco input modules, the FMB features clean feeds, signal ducking, a program timer and comprehensive monitoring within a choice of two frame sizes. An AFW feature is also available for post applications.

Circle (118) on Rapid Facts Card

Klark-Teknik DN726 delay

The stereo delay line accepts two inputs and provides stereo, in-phase outputs. The frequency response is 20Hz-20kHz. Specifications on the range of delay for the unit is 0 to 1.3 seconds, adjustable in 20s increments, accomplished with 16-bit linear processing for superior performance in demanding situations. A backup battery for memory retention is featured. Retail price is \$3,900.

Circle (120) on Rapid Facts Card

Summit Audio Warm Interface

Used as an interface between CD players, R-DAT recorders and analog

equipment, the unit provides a tube sound and level matching. The interface features two channels, electronically balanced input/output, +25dBm maximum output, input design to work with -10dB or +4dB systems, screwdriver trim gain adjustment from the front panel, maximum gain of 20dB, maximum input of +24dBm, dynamic range of 110dB, harmonic distortion of less than 0.1%, and frequency response of 3Hz to 90kHz. Suggested list price is \$950.

Circle (123) on Rapid Facts Card



Apogee 924 analog filter/signal processing module

The 924 analog filter/signal processing module features programming pins that enables the formation of complete analog processing in the front or rear end for

digital audio. Also included are an input buffer, an RF filter, a linear phase filter, pre- and de-emphasis, dc servo and an output buffer.

Circle (121) on Rapid Facts Card

ProCo Multiface series switchers

Two new additions to the Multiface series are the RMS-1 and RMS-2 Remote Monitor Switchers, designed to switch power amplifier outputs to alternate speaker systems. The RMS-1 uses one stereo pair of inputs with switches for main/alt and alt A/alt B. The RMS-2 operates in an identical manner, but accommodates two stereo pairs. Suggested price is \$275 for the RMS-1, and \$389 for the RMS-2.

Circle (122) on Rapid Facts Card

NED Synclavier 3200

The 3200 is a modular compact workstation aimed at customers that could not afford a regular Synclavier system.

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NEW PRODUCTS

The 3200 provides up to 32Mbytes of RAM, 32 mono voices and 720Mbytes of hard disk sound storage capacity. The customized Macintosh II graphic workstation, which is standard, controls the system. Other figures include 16-bit, 100kHz multirate sampling, 200-track sequencing and SMPTE/VITC synchronization.

Circle (126) on Rapid Facts Card

Otari MX-50 two-track recorder

Available in both 15/7.5ips and 7.5/3.75ips versions, the MX-50 features built-in tape timer displays with search-to-cue and search-to-zero, front panel $\pm 7%$ varispeed, 10.5-inch reel capacity, headphone amplifiers, electronic lifter control and a dump edit function. An optional Voice Editing Module (VEM) is available, which provides 2X playback without pitch shift. Price is \$2,495.

Circle (124) on Rapid Facts Card



Midas XL2 mixing console

For touring sound, theater and broadcast applications, the XL2 comes in frame sizes of 24, 32 and 40 inputs. A 16-input channel expander section couples to the existing frame via a single, multipin connector. In addition to eight subgroups, there are eight aux sends with full pre/post switching, which enables the console to provide both main or monitor mixes without requiring a separate, dedicated monitor console. Two matrix outputs can receive a submix from any or all of the eight subgroups.

Circle (127) on Rapid Facts Card

Tannoy TPI reference monitor

The TPI features a high-frequency unit developed as a result of the company's Dif-

ferential Material Technology. Using a deep drawn duralumin diaphragm and skirt with a separate silicone-based suspension gives the piston rigidity associated with titanium but without HF breakup modes in the passband. The coil is ferro fluid-cooled and the driver uses a sculptured asymmetrical phase plate. Both are knitted together with a hardwired crossover.

Circle (128) on Rapid Facts Card

API 525B compressor/limiter

Based upon the 525C design, the 525B has an added motorized gain-reduction device. Three modes can be selected: the standard 525C configuration, a passive element with an adjustable output stage for additional gain, and a completely passive element with no electronics in the circuit. According to the company, the passive mode allows for compressing and limiting with no distortion or coloration.

Circle (129) on Rapid Facts Card

Tannoy SGM-15B monitors

The SGM-15B is designed for users needing a small, yet powerful monitoring system. The monitor contains a 15-inch K-3809 dual-concentric in a cabinet measuring 26½"×19¾"×18½". The crossover uses the same hard-wire construction and a high-current EQ used in the Super Gold Monitor series.

Circle (152) on Rapid Facts Card

Alpha Audio Boss/2 automated audio editor

The successor to the Boss 8400, the Boss/2 features digital waveform editing and concurrent multiprotocol communication using RS-422, RS-232, SMPTE and MIDI. The unit can talk directly to any machine that speaks Sony, Ampex, ES-BUS or other serial protocols. Users can also select whatever combinations of synchronizers are best for their system.

Circle (125) on Rapid Facts Card



Milab D-37 mic

The D-37 is a dynamic cardioid mic designed for rugged use while maintaining high performance standards. The mic is constructed of solid brass and includes such features as a heavily shock-mounted moving coil element for minimum handling noise and built-in pop protection. Frequency response is 50Hz to 20kHz, with a favorable boost in the vocal/presence range.

Circle (132) on Rapid Facts Card



Technics SL-P1300 CD player

The pro unit incorporates 8x oversampling, four A/D converters—two for each channel—and 18-bit technology. Using four A/D converters allows using one for each half of the analog waveform, improving the digital processing of low-level signals that are difficult to capture. Features include two-speed search dial cueing, a rocker control that moves the laser by one pit track, cue point memory and a numeric 10-keypad. Suggested retail price is \$1,800.

Circle (130) on Rapid Facts Card

ASC Quick Sound Field

Acoustic Science's Quick Sound Field consists of ½-round Tube Traps on the walls or ceiling of a studio or voice-over booth. The effect is a fast decay rate that is consistent with all frequencies and very diffuse at high frequencies. Additionally, the company says, there is no loss in the upper frequencies and no bass boom associated with smaller rooms. The QSF is available as a retrofit for existing spaces. Retail price is \$22 per square foot. Custom free-standing booths are also available, and a 3'×4' QSF baffle for portable use is also available at \$375.

Circle (131) on Rapid Facts Card

Hardware and software updates

NED PostPro enhancements

The 8-track Direct-to-Disk multitrack recording and editing system now features a dedicated remote controller/editor/locator, which features programmable buttons that can define command functions. Software enhancements include time compression, direct digital transfer, VITC/SMPTE synchronization and CMX edit list conversion.

Circle (161) on Rapid Facts Card

Sony upgrade to APR-5003 recorder

The upgraded version of the two-channel recorder, called the APR-5003V, features a nine-pin serial cable interface, allowing the recorder to be connected to Sony BVE-900/9000 video editors. The synchronizer features the ability to externally reference a video signal of 50Hz and 60Hz tone, and both LTC and VITC and also accepted. The "bit bump" feature allows for bit-accurate manual adjustment in synchronous mode. List price is \$11,950.

Circle (162) on Rapid Facts Card

Impedance alternative for JBL 2204, 2123

Previously available only in 8 Ω versions, the 2204 12-inch LF loudspeaker and the 2123 10-inch midrange transducer are now available in 16 Ω versions, allowing more flexibility in the design and installation of sound systems. The 8 Ω version is designated by H (as in 2204H); J refers to the 16 Ω impedance.

Circle (163) on Rapid Facts Card

New software for Lexicon 2400

Version 2.20 expands the 2400 audio time compressor/expander's interfacing capability to include several new videotape machines, including the Ampex VPR-6/VPR-80, Sony BVH-3000 and Panasonic AU-660. A Bypass Play command issues a servo-locked play command from the front panel. Software-assignable relays and inputs provide remote control of this feature, as well as remote control of the 2400's machine control capability. Version 3.0 software provides all of the machine interfaces in V 2.20, as well as dc servo capability for controlling the Panasonic AU-650 MII VCR. Price for V 2.20 is \$125. V 3.0 is priced at \$250.

Circle (164) on Rapid Facts Card

EQ option for Lexicon Opus

The EQ/filter option allows the Opus to provide 12 channels of digital equalization. The option is comprised of three basic elements. The DSP modules are installed in the host card cage. The EQ control strip is installed in the workstation. EQ application software drives the system. Each of the 12 channels contains four independent filter section, each covering 20Hz to 20kHz. Each of the channels can be assigned to one of six different filter characteristics, allowing flexibility and control over the program material.

Circle (165) on Rapid Facts Card

Updated version of Otari DTR-900

The 900B features a redesigned autolocator/remote software and hardware, proprietary VLSI technology, in-house manufactured heads and upgraded power supplies to accommodate the use of Apogee Electronics low-pass filters in

the A/D and D/A sections. No price increase is expected, and the new features can be retrofitted to earlier models. Two accessories have also been introduced: the EC-104 plug-in chase synchronizer module, and the CB-504, a PD-to-DASH format converter.

Circle (166) on Rapid Facts Card

Improvements to Agfa tape and reels

Agfa's PEM 469 mastering tape has a new backcoat, base film and oxide improvements, enabling the tape to withstand more tape passes. The PEM 291D digital audio master tape has a new formulation designed for the current generation of digital recorders and ensures low error rates. Also, the company has redesigned the flange on the reels of the 468 and 469 tape lines, making it easier for users to thread and load the tape.

Circle (167) on Rapid Facts Card

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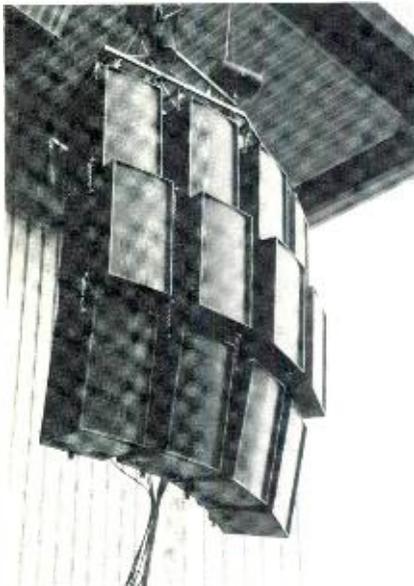
Circle (33) on Rapid Facts Card

NEW PRODUCTS

Martin Audio F2

The F2 is a two-box touring sound system that features full horn loading. The top is configured as a rack-mount shell that can accept different horn and driver combinations, depending on the application. A rack can be made up of only mid or high horns, for long-throw use, or can be configured as a mid/high pack for other applications. A rigging system enables arrays to be built with extended columns of bass, mid and high horns, which can provide clean, high-level music to all audience areas, while using a minimum number of cabinets and amplifiers.

Circle (133) on Rapid Facts Card



Audio Precision DSP-1

For use with the System One, the DSP-1 provides signal generation and analysis in the digital domain, as well as providing feature enhancements for the System One's analog measurement capabilities. For digital applications, the DSP-1 provides digital waveform generation and digital analysis capable of different functions via downloadable software. Direct digital I/O ports are optional. Additional analog measurements for the System One include FFT-spectrum analysis, individual harmonic distortion and total harmonic distortion without noise.

Circle (135) on Rapid Facts Card

Turbosound TXD-530

The 530 is designed to provide extra-wide dispersion in applications that re-

quire coverage over a wide area from one source, such as an under-balcony fill. Components used are two 10-inch drivers and one slot tweeter. Dispersion is $120^{\circ} \times 50^{\circ}$, with a frequency response of 100Hz to 18kHz.

Circle (136) on Rapid Facts Card

Intelix Studio Psychologist

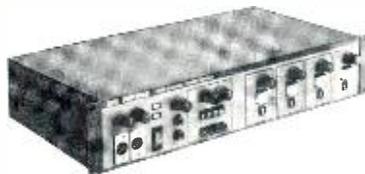
The remote-controllable matrix mixer/router system allows individuals to mix their own headphone mix. Actual mixing takes place in a modular rack-mounted unit. Each module has 16 mixer nodes using digital attenuators with 85dB control range and a 1.5dB resolution. Inputs and outputs are TRS-balanced. The mixer may also be controlled by a computer with an RS-232 interface. Applications include recording studios, broadcast routing, sound contracting and theatrical sound.

Circle (137) on Rapid Facts Card

HME additions to 700 series

The RP743 and -753 four-channel power stations are the newest additions to the 700 series cabled intercom line. With the 742, two headsets can have communication access to any of the four independent channels. The 753 is a four-channel mix matrix power station with a panel of matrix switches that assign 12 stations or groups to one of four independent intercom channels or two private lines. Both are compatible with 700 series line, as well as most three-wire intercom systems.

Circle (138) on Rapid Facts Card



Dolby 363 two-channel noise reduction

The 363 allows Dolby Spectral Recording and A-type noise reduction to be switchable. The unit contains two channels in a 1U frame. Both channels are equipped with a built-in record/playback changeover capability, allowing a single unit to be used in stereo applications. The 363 is normally supplied with two Cat. 300 modules, which contain both SR and A-

type. Optional SR-only or A-type-only versions can be ordered.

Circle (139) on Rapid Facts Card

Hard disk version of Publison Infernal Machine

The Infernal Machine's hard disk version allows one hour and 48 minutes of recording in mono mode, and 54 minutes in stereo mode. Sampling rate is 50kHz, with 16-bit linear A/D conversion. When used with the IM80 color editing software, the Infernal Machine is connected to an AT computer via RS-232 for the creation of editing lists, non-destructive editing and real-time chains. The optional DAB-3 rack allows up to six hours of digital sound to be recorded on a high-speed, high-capacity digital cassette.

Circle (140) on Rapid Facts Card

JBL 12SR loudspeaker system

Featuring a 12-inch LF transducer and a 1-inch exit compression driver on a Flat Front Bi-Radial horn, the Control 12SR is designed for high sound quality at high sound-pressure levels. The LF transducer features a 3-inch, diameter edgewound aluminum voice coil for clean response and high power handling. Using the Bi-Radial horn allows HF energy to be dispersed in a $90^{\circ} \times 40^{\circ}$ pattern.

Circle (134) on Rapid Facts Card



Chrystal Semiconductor A/D converter

The CSZ5126 is a low-cost A/D converter designed for a variety of applications such as DAT decks and digital audio

EQ systems. Manufactured in low-power CMOS, the chip features a patented self-calibration circuitry, which results in improved performance, according to the company. Dynamic range is more than 92dB in stereo mode and more than 95dB in 2x oversampling mode. The converter is initially available in 28-pin DIPs. Lots of 1,000 are \$27.20.

Circle (141) on Rapid Facts Card

Sonic Solutions Desktop Audio system

The workstation is designed to allow users to prepare master recordings with a computer and applications software in the same way as a desktop publishing system works for writers. The system includes a Macintosh II, one or more hard drives and a Sonic Signal Processor circuit card. Software used is CD PreMastering Desktop, which provides all-digital editing, mixing, EQ, dynamics and project management functions for mastering. A new version of NoNOISE software is available as an option. Basic system cost with the premastering software is \$44,100.

Circle (142) on Rapid Facts Card

Turnkey systems from Offbeat Systems

The turnkey systems for computer-based music scoring include the computer, circuit boards, and Offbeat's Streamline or Clickstation software. Three configurations are available: the Transportable Streaming System, Laptop Clickstation and Starter Station. Also available is the MIDI Synthesizer Drive, an optional PCB plus software that allows MIDI-based sequencers and devices to be directly controlled by the Streamline or clickstation systems.

Circle (143) on Rapid Facts Card

Agfa-Gevaert PE 647-947 tape

The chrome bulk tape is designed for users who need a high-performance, Bias II cassette formulation. Features include superior electroacoustic specifications, durability, consistency and slitting characteristics, according to the company. The tape is available in 8,200-foot lengths for C-60s and 11,500-foot pancakes for C-90s.

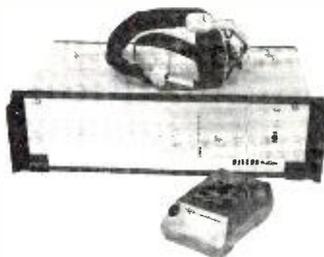
Circle (145) on Rapid Facts Card

Diless ProCom intercom

The full-duplex intercom features push-

button dialing for each station, allowing users to make the required connection by pushing one button. Any station may be used as a beltback or table-mounted unit, and stations are connected with standard three-core mic cables. The automatic central unit can handle 11 simultaneous connections, and no line is ever busy. A special remote station is also available with four public address outputs.

Circle (144) on Rapid Facts Card



Beyer DT 770/990 headsets

Both headsets feature a frequency response of 5Hz to 35kHz to reproduce

complex waveforms and upper-end harmonics accurately. The 770 features a large, but low-mass, diaphragm embedded in a bass-reflex system. The 990 is identical but features a circumaural, semi-open-air design.

Circle (146) on Rapid Facts Card

ADx-22 synchronizer

The synchronizer offers synchronization and emulation as a menu selection in the same unit. The unit will lock to a single time code bit in less than five seconds, making it possible for unmodified pre-roll times. Other features include two independent time code readers, one time code generator, an RS-422 data port, an optically coupled parallel port, four DPI relays, an external audio triggered mark function and multiple time code memory scratchpad.

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ProCo junction boxes

The model HJ-6 allows splitting of a mono or stereo amplifier output to feed



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Acoustic Products for the Audio Industry

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NEW PRODUCTS

six sets of 150Ω-600Ω headphones. It may be used in conjunction with the RMS-1 through a special output jack provided on the switcher. The HJ-4P splits the output of a stereo amplifier to four sets of 150Ω-600Ω headphones, but uses XLR-type connectors on the input and master left and right level controls. Retail prices are \$89 for the HJ-6 and \$149 for the HJ-4P.

Circle (157) on Rapid Facts Card

Teac LV210A videodisc recorder

The unit is the first 12-inch two-sided laser videodisc recorder for less than \$20,000, according to the company. The unit retains most of the interactive features of the LV200 recorder, eliminating the CLV mode, and has the same external computer protocol. Also available is the LV210P videodisc player, which is compatible with the LV200, minus the CLV mode. Price for the LV210A recorder is \$17,995; price for the VL210P player is \$4,995.

Circle (148) on Rapid Facts Card



Sony TCD-D10 PRO DAT recorder

A scaled-down version of Sony's PCM-2000 DAT recorder, the TCD-D10's design criteria is based on the TCD-DP PRO analog cassette recorder. Recording is at 48kHz, with playback at 44.1kHz. Also included are XLR I/O connectors, internal clock for tape location, AES/EBU digital I/P and 2x oversampled digital A/D and D/A converters. The recorder can operate up to two hours in the field with an internal rechargeable battery.

Circle (149) on Rapid Facts Card

Aphex 124 audio level interface

The 124 is designed to combine -10dBV consumer IHF audio equipment with +4dBm or +8dBm professional/industrial audio equipment, allowing both systems to operate at optimum performance levels, matching impedances and operating levels. The unit features transformerless,

servo-balanced inputs and outputs for extended frequency response and exacting transient response. Suggested retail price is \$219.

Circle (150) on Rapid Facts Card

Synergy One from Analog Digital Synergy

The Synergy One is an in-line, modular digital mixing console available in frame sizes from four to 64 channels. Standard features include digital four-band EQ, high- and low-pass filter, 1Hz offset filter, de-emphasis, remaining headroom indicator, digital bar graph metering for each channel and group faders. Each module is equipped with a true 16-bit fader, providing 65,536 level steps. Available options include time code-based full-function automation, total automation data reset and format conversions.

Circle (151) on Rapid Facts Card

Sony DAU series ¾-inch tape

The DAU series ¾-inch digital audio master cassettes feature the company's Vivax magnetic particles for clear digital sound production. Carbon mirror backcoating reduces the error rate to a minimum. The tape is available in 30-, 60- and 75-minute lengths. The tape comes in an anti-static shell to eliminate static electricity.

Circle (152) on Rapid Facts Card

Real World Research Audio Tablet

The Audio Tablet is a random-access audio editor designed for various applications, including dialogue and music editing, and CD and 12-inch mastering. The tablet is a two-channel system that encodes audio signals in a linear 16-bit format at 48kHz, 44.1kHz or 32kHz. Both analog and digital I/Os are available. The system consists of three components: the interface, the processor rack and a peripheral rack. The rack components occupy 8U of space, and the tablet interface is an ASCII keyboard and touchscreen.

Circle (154) on Rapid Facts Card

Lexicon LXP-1

The unit is a multi-effects processing module that provides a variety of digital reverberation and effects in a cost-effective format. Its 16 programs include halls, rooms, plates, gates, inverse reverbs,

delays and choruses. Stereo inputs and outputs are available. Using a two-level control system gives users access to more than 4,000 sounds. Also available are 16 factory presets and 128 user registers for storing personal setups.

Circle (155) on Rapid Facts Card

Beyer MCE 86 shotgun mic

Weighing 95 grams, the MCE 86 is one of the lightest shotgun mics available. Frequency response is 50Hz to 18kHz, with a maximum SPL of 148dB. The unit can be powered by any phantom power source generating 12V to 48V. Two suspensions are available, one for all purpose use and another for use with a video camera.

Circle (156) on Rapid Facts Card

Gentner Electronics EasyTerm

The rack-mount wiring termination system allows users to open the termination like a door to gain access to equipment inside the rack. Wires are run from inside the rack and are connected using an impact punch tool. An optional "kickbar" holds the unit open while running and punching cables. Two configurations are available. EasyTerm/66 uses a 66-style punch block, for use with solid wire only. EasyTerm/FB uses Gentner's Flexiblock, and is designed for stranded and solid wire.

Circle (158) on Rapid Facts Card



Soundtracs Tracmix

The centrally controlled, stand-alone fader and mute automation system operates via a remote keyboard and color monitor. The automation allows for up to 64 channels of fader and mute automation using dbx VCAs for low noise and distortion. Mix data is held in RAM to eliminate unnecessary disk storage, but it may be saved along with grouping information, tack listings and MIDI song data onto a 3.5-inch disk. A time code generator is also built-in.

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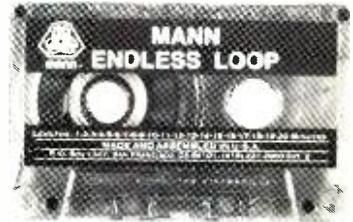
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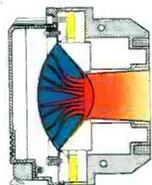
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