August 1987

Recording
ENGINEER/PRODUCER

The Technical Journal for Audio Professionals

AN INTERTEC PUBLICATION

Using Music Libraries
Page 74

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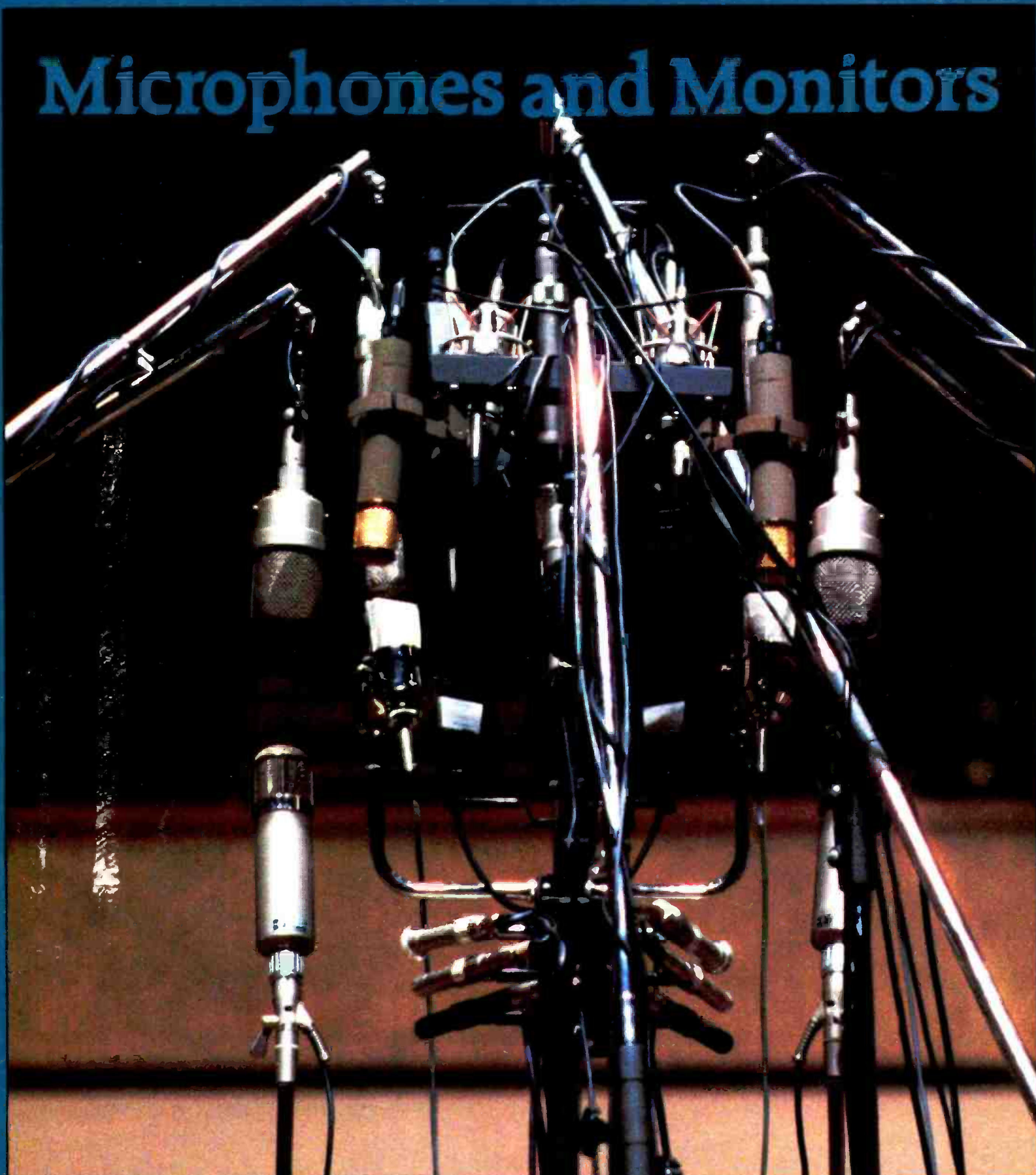

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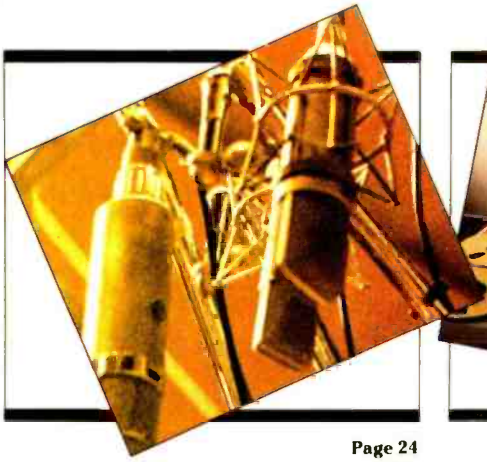

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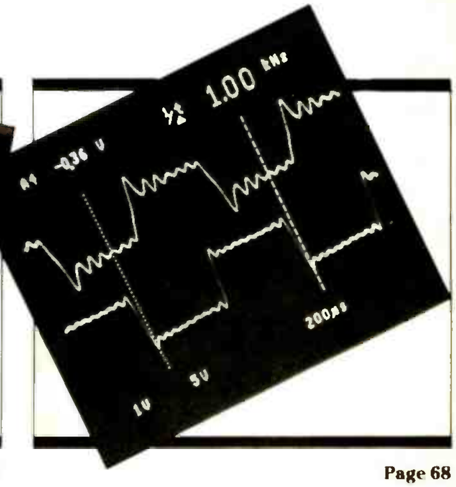

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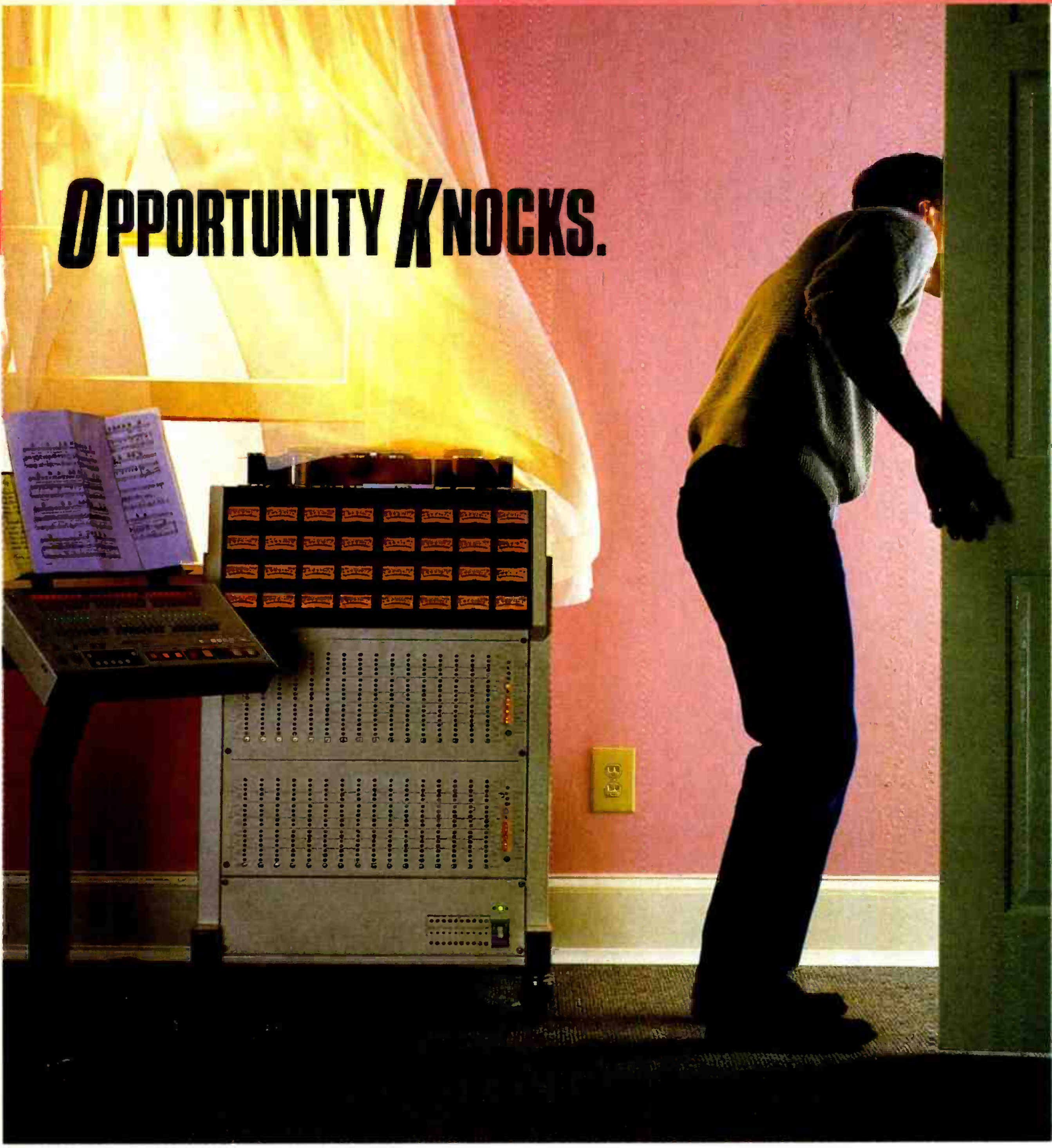
On the Cover:

A cluster of stereo microphone arrays at the University of Iowa during one of Professor Lowell Cross' evaluation sessions, the results of which form the subject of a feature article beginning on page 24. Photography by the University of Iowa.

RECORDING ENGINEER/PRODUCER—Volume 18, No. 8—(ISSN 0034-1673) is published monthly by Intertec Publishing Corporation, 9221 Quivira Road, P.O. Box 12901, Overland Park, KS 66212-9981. Second-class postage paid at Shawnee Mission, KS, and additional mailing offices. POSTMASTER: Send address changes to Intertec Publishing Corporation, P.O. Box 12901, Overland Park, KS 66212-9981.



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Circle (5) on Rapid Facts Card

Editorial

Audio Potentials

By Mel Lambert, Editor

It cannot have escaped anybody's attention in the pro-audio industry that the recent CopyCode scanner proposals have been receiving a great deal of media attention during the past several months. I have chosen to remain silent on the entire puzzle, simply because I feel that the subject of piracy and anticopy legislation represents a smoke screen erected by a record business that will clutch at just about any excuse—however feeble—to explain away falling album sales.

Aside from the fact that I consider the volume of album sales the record companies claim to be losing through home taping is extremely overinflated, I would suggest that the record-buying public is ill served by A&R staff at major labels who show a bewildering lack of foresight in attracting new talent. I am convinced that stunted album sales can be attributed more to lackluster product than to consumers trading albums so that they can make cassette copies rather than buy them.

Sales motivation aside, however, I feel that the way in which the anticopying legislation has been presented bears further examination. I am stunned that a functioning notch filter capable of transparently removing the required 200Hz window between 3.7kHz and 3.9kHz has yet to be demonstrated to the record industry. Only within the last several days have I received details of a digital processor which, according to every report I hear from trusted ears throughout the industry, produces this deep frequency notch with *no audible side effects whatsoever.*

Of course, the entire debate, at least as far as artists, producers and engineers are concerned, centers on the fact that whatever the record companies plan to do to reduce the perceived level of home taping, the chosen technique should not damage the product that they have worked so long and hard in the studio to achieve.

All the information I have seen to date details the subjective and measured effect of *analog* filters that have been designed to produce the desired notch. I understand that even CBS Laboratories has failed to produce a design whose action on program material wasn't immediately obvious.

And yet the record industry expects our legislature to pass tough measures that will force manufacturers of R-DAT (and other) machines to insert a scanner that would prevent recording when it detects the frequency notch. It remains to be seen quite what the system's inventors plan to do about my simply adding a minute amount of time-dependent pitch shift to the encoded material prior to making an R-DAT recording, and hence fooling the detecting scanner. I'm sure that more than one manufacturer must be looking at providing such hardware to the consumer market, so that we can make personal-use archive copies of our favorite CopyCoded vinyl albums and CDs.

All of which is off the point, however. What dismays me more than a little about this entire affair is that an alarming number of audio professionals have become too bewitched by the fact that, because we cannot construct an inaudible notch filter using current technology, we should do our damnest to prevent the introduction of CopyCode legislation. I applaud, therefore the fact that at least one independent company has developed an *inaudible* digital notch filter that could be used to encode CD and vinyl releases. Not just because this practical solution represents feasible technology, but because its very existence highlights the lack of integrity on the part of the record companies that wanted to introduce such a system in the first place.

It seems to me that, given the record industry's desire to curb home taping, the *first* thing it should have done was to develop the necessary technology to achieve the intended result. The system should then have been made available for independent evaluation by as many industry professionals as possible. Only when we had all agreed that—strictly as a piece of viable technology—the system did not produce any audible degredation of the program material, should it have been offered as a potential anticopy measure.

Having established independently the credibility of such technology, I suggest that its proponents could then have made a far better case for it being implemented than has been seen to occur during the past several months. How anyone could have advocated that legislation be passed to force the use of technology that, at the time, didn't even exist in physical space—and could not therefore be shown to produce the desired subjective results—is quite beyond belief.

How this issue has developed has left me more than a little puzzled. To my knowledge, there has never been a technology that has tried to be outlawed before it was introduced. And now we're being asked to consider an encoding system whose effectiveness is uncertain at this point.

What's worse, most of the arguments on either side of the issue have focused on the consumer, while ignoring the working professional. At **RE/P**, we are attempting to so something about this.

Very soon, you may get a questionnaire about the R-DAT issue. If you do receive one, fill it out and return it to us immediately.

We need your input. Don't delay. DAT is a vital issue that, whatever the outcome, will profoundly affect how you do your job in the coming years. R·E/P

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News

John Hammond dies

Producer John Hammond, whose musical discoveries and recordings have had a lasting impact on American popular music, died July 10 in New York. He was 76.

For more than 50 years, Hammond was responsible for discovering some of the most influential musicians of their generations, including Billie Holiday, Count Basie, Teddy Wilson, Lester Young, Aretha Franklin, Bob Dylan and Bruce Springsteen.

His recording sessions were among the first to be racially integrated; he also organized the "From Spirituals to Swing" concerts at Carnegie Hall in 1938 and 1939, believed to be the first concerts to present black artists in a concert setting.

Hammond's first sessions were in 1931. Born into a wealthy family, he subsidized these recordings with a trust fund set up to provide him with a yearly income.

Among the more notable studio dates through Hammond's career were Basie's first recorded sessions, blues singer Bessie Smith's last recordings and Dylan's first two albums.

Producing "involved getting the musicians to play the way you wanted them to play and insuring that the engineers preserved the sounds faithfully," he wrote in his 1977 autobiography, *John Hammond on Record*. "If everyone did his part diligently you were likely to emerge with a nice record."

Digital Audio Disc Corporation presses 50-millionth CD

The first Compact Disc plant built in the United States is now the first American facility to reach the 50-million mark. In honor of the manufacture of the 50 millionth disc (a CD version of Barbra Streisand's *One Voice* album), Michael P. Schulhof, president of DADC, presented a plaque to Akio Morita, chairman and co-founder of Sony Corporation, Tokyo.

According to James Frische, DADC executive vice president, "When we started production in 1984, we were employing fewer than 100 people. Today we are able to provide jobs to nearly 600. This trend will continue as we build toward our goal of 70 million CDs a year by 1988."

DADC is a wholly owned subsidiary of Sony Corporation of America.

NED optical disk storage and retrieval system

The new optical disk-based storage and retrieval system for the Synclavier Digital Audio Workstation is designed to augment the existing hard disk or tape storage system, allowing users to store more than 5½ hours of digital material on each optical disk.

According to Mark Terry, NED's director of marketing, "Sounds can be broken down any way the user wants into music, sound effects, dialogue, etc. As many as 1,300 15-second effects, for example, can be stored, archived and accessed with a single keystroke from the Synclavier keyboard."

A software menu allows users to name

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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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News

and classify individual sounds into a cross-referenced library. The new system also performs global searches for individual sound files.

CompuSonics signs OEM deal for European markets

The company recently completed an OEM deal with AVM Ferrograph, headquartered in Jarrow, Tyne & Wear, England, whereby CompuSonics products will be marketed in European countries under the Ferrograph brand name.

According to company president David Schwartz, AVM Ferrograph has placed an initial order of $200,000 worth of CompuSonics equipment, which will be distributed in England and Europe carrying both the Ferrograph brand name and the CompuSonics CSX processing technology trademark. In the second phase of the agreement, AVM Ferrograph will manufacture CompuSonics products at its factory in England.

Schwartz estimates that, by the end of 1988, sales in England and Europe will equal U.S. sales, which presently amount to more than $1 million.

Brooke Siren Systems changes name

The manufacturer of crossover systems and signal processors recently changed its name to BSS Audio Ltd., to better reflect the company's intended growth and expansion within the industry.

Part of the EdgeTech Technology Group, BSS was formed by Chas Brooke and Stan Gould, chief operating officer of Siren Controls. Joining forces in 1979, the duo created Brooke Siren Systems.

AEA sets up new MIDI company

Audio Engineering Associates, Pasadena, CA, has opened MIDI Works, a new division that specializes in the sale of MIDI-based audio system products. New lines include Sequential keyboards, including the new Studio 440 SMPTE-based sampler/drum machine; the entire Akai range of pro-audio keyboards, including the MPX-80 8-channel automated MIDI mixer; the Fostex line of synchronizer systems and MIDI autolocator; and various software products to support the above.

For additional information, contact Karl Wirz at: 818-798-9127.

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Letters

MIDI in the studio

From: Rick Shaw, Music and FX, Marietta, CA.

As usual, it takes me some time to find time to read my copy of **RE/P**, and so I have just got around to reading Paul Lehrman's article on "The Celtic Macintosh," published in the March issue. I'm no authority on Irish music, but it was an enjoyable article. (Even if it took some time to figure out the correct order of pages in the mag.) I have read previous articles that Paul has written on the topic of MIDI and always learn a few good tips.

In the March article, I was particularly interested in Paul's use of Southworth Total Music software on his Apple Mac; I have also been using this system for some time and thoroughly enjoy it. Total Music has allowed me the advantage of late-night improvisation with the ability to clean-up a musical thought in a "post-production" sense. The amount of reasonable music that I have been able to turn out since I started using my Mac+ and Total Music software has just about doubled. It has also been a tremendous time saver in the composition of background music for television production—for me that amounts to mostly spots and themes.

Just for your information, my system drives the following devices: a Yamaha DX-7, Roland JX-8P, Yamaha FB-01, Ensoniq Mirage, Sequential Circuits Drumtracs (I purchased it a couple of years ago, mostly because I could use additional chips from Digidrums), Roland Planet P, and a Yamaha REV-7 digital reverb whose patches can be selected from the computer. Like Paul, I haven't used the built-in programmer on the Drumtracs since I started using Total Music.

The ability to use Total Music in the composition of music for video has been a tremendous plus for my business. Recently, I was asked to write three pieces for an ad agency here in Atlanta that was putting together a production for Charter Lake Hospital. The spot they were working on was about children that had learning disabilities, and the video was shot "point-of-view" as if you were a third-grader at school. The music needed to be somewhat of a dramatic nature, but the agency wanted three backgrounds to choose from.

I saw the spot for the first time the night before the production, and they needed all three versions done by 4 p.m. the next afternoon. It would have never been possible for me to get three tunes recorded and somewhat presentable in this length of time before, especially working with visual cues on my videotape reference copy that needed a little musical sweetening.

All in all, I am very happy with the Mac, which seems to have become one of the most valuable pieces of equipment in my studio. R·E/P

Wave goodbye to one-color sound

The full-color sound of the ESQ-1 Digital Wave Synthesizer makes other synths sound . . . well . . . black and white by comparison.

After all, a broad pallette of sound is your first criterion for a synthesizer. And the major international music magazines who've reviewed the ESQ-1 seem to agree.

The tone colors possible with 3 digital wave oscillators, 4 envelopes, 4 DCA's, 3 LFO's and 15 routable modulation sources for each ESQ-1 voice impressed KEYBOARD magazine's Jim Aikin. "The ESQ's voice offers far more than what you'll find on a typical synthesizer — even on some instruments costing twice as much".

In somewhat colorful comparative terms, Peter Mengaziol of GUITAR WORLD wrote, "The ESQ-1's sound combines the flexibility and analog warmth of the Oberheim Matrix-6, the crisp ringing tones of a DX-7, the realism of a sampler, the lushness of a Korg DW-8000 and polytimbral capacity of the Casio CZ-1".

There are hundreds of ESQ tone colors available on cartridge, cassette and disk. And now, each ESQ-1 includes an Ensoniq Voice-80 Program Cartridge. Added to the 40 internal programs, that's 120 great keyboard, acoustic, electronic, percussive and special effect sounds right out of the box.

MUSIC TECHNOLOGY's Paul Wiffen had a great time mixing colors with the ESQ-1's 32 on-board waveforms and 3 oscillators per voice. "After a few minutes of twiddling, you can discover that, for example, an analog waveform can make the piano waveform sound more authentic, or that a sampled bass waveform can be the basis for a great synth sound. Fascinating stuff!"

Even though its flexibility is unmatched in its class, creating sounds on the ESQ-1 is simple and intuitive. Mix a little blue bass with some bright red vocal and pink noise and get a nice deep purple tone color.

But there's one color you won't need a lot of to get your hands on an ESQ-1 — long green. The ESQ-1 retails for just $1395US.

There are sound librarian programs for the ESQ and most personal computers, so you can save and sort your creations quickly and easily. If you'd rather just plug it in and play, there are hundreds of ESQ sounds on ROM cartridges, cassettes and disks available from Ensoniq and a host of others.

The easiest way to see the possibilities for yourself is to follow the wave to your authorized Ensoniq dealer for a complete full-color demonstration.

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Circle (12) on Rapid Facts Card

Managing MIDI

By Paul D. Lehrman

A couple of months ago, I wrote a list for Santa telling him about some of the new software toys I'd like to see this year. I wrote it early, because in the music industry Christmas comes in June at the Chicago NAMM show. What I didn't put on the list were my ideas for new hardware I'd like Santa to bring me. So here is that list, also nice and early, to give Santa's R&D elves plenty of time to get their act in gear.

• A completely assignable MIDI control panel.

There are a number of products that let you change MIDI controller numbers and other information in real time, so that, for example, you can pretend your modulation wheel is a breath controller. There are also devices that give you a few extra sliders, or jacks for pedals, so that you can play with controllers besides the ones your keyboard already has. These are nice, but I want more.

When Lucasfilms was getting started with their SoundDroid, they had in the lab a prototype of a completely "soft" console—every button, knob, switch, and slider on the control surface was assignable to any function that the system was capable of performing. The SoundDroid is dead (killed by a duck according to reliable reports), but the concept of a soft console deserves to live.

Imagine a control surface that looks vaguely like a mixing console, but in addition to the usual sliders and buttons, there are a couple of ribbon controllers, a few synth-like wheels, and a cluster of touch pads for good measure. With the proper hardware hooked up, you can use it as a mixing console, a bank of equalizers, a patch bay, a bunch of pitchbend controllers, a multi-parameter synth programmer, or any combination or variation thereof.

Each control sends out MIDI note, velocity, or controller data, and every one is completely mappable. Next to each control is an LCD saying what it is, and at what value it's set. There's an internal memory for several hundred configurations, and they can also be transferred to an external storage device over MIDI system exclusive. Configurations can be instantly and silently changed with MIDI program changes.

The unit should be able to be manufactured pretty cheaply. After all, audio performance isn't an issue—there isn't any audio! If someone wants to be fancy, they can put little motors in the controls that physically move them to the correct positions whenever data is received from a sequencer.

• A drum pad with velocity, position, pressure and duration.

The problem with most MIDI drum pads is that they know when and how hard they've been hit, but that's about it. We're beginning to see pads that know *where* they've been hit, and can use that information to control volume, pitch, pitchbend, filter cutoff, or whatever. But there are still other parameters to look at.

Drum pads normally send out a note-on followed by a note-off after a fixed, usually very brief, interval. Although most of the time this is appropriate, there are occasions when a drummer does something with a drum head *after* he's hit it—like put pressure on it, or move the stick around, or mute it with his hand, or just hold the stick down to change the way the drum rings. How about MIDI-ing these gestures, using both duration and aftertouch? Aftertouch could be used to add vibrato, or noise, or even pitchbend after the stroke, while the sound was still ringing. And for folks (like me) who have no drum technique, how about mapping the duration to a repeat function, so that when you hit and hold a note, it executes a perfect two-stroke roll?

> ***The idea of associating a compression curve with a particular synthesizer patch sounds very appealing.***

• Real-time MIDI equalization and compression.

We're already pretty close to this on two fronts. There is on the market a graphic equalizer with memory that can be addressed over MIDI, and a new expensive mixer has both parametric equalization and rudimentary compression that are addressable by MIDI. But I'm looking for more than that, and I want it cheap.

Not too many digital synthesizers let you change voice parameters smoothly in real time, primarily because they're dealing with discrete digital values, not analog voltage levels. A properly designed equalizer (maybe with a little bit of hysteresis built in to keep it from sounding digitized) could conceivably do a much better job. There are significant uses for real-time, automatable control over the parmeters of a complex equalizer with acoustic sources as well.

The same goes for compression, although the real-time aspects of it are probably less important. But the idea of associating a compression curve with a particular synthesizer patch sounds very appealing.

• A cheap MIDI mixer.

There are MIDI-controlled fader boxes, and there are one-rack-space 8x2 mixers perfect for use with a rack full of synth modules. I would like to see these products combined.

Give the box a bunch of internal memory for storing level settings as program changes, and also let each fader be accessible in real time using controllers, note numbers, and/or velocities. Make it sound good, be easy to use and inexpensive. Give me three of them, and I can rule the world.

• A drum synth that isn't a drum machine.

I hate drum machines. I used to program them a lot; then I got a MIDI sequencer, and now I think I've completely forgotten how. Good riddance. How can anyone who's used a powerful sequencer to program tracks, using a velocity-sensitive keyboard or drum pads for input, ever go back to pushing those stupid little buttons? With megabyte computers in every studio, the old excuse that sequencers don't have enough memory to handle drum tracks isn't valid any more. So why are there no drum synths on the market that just have *sounds*, without all that fancy (and expensive) programming junk? I never use it, so why do I have to pay for it?

Some folks use samplers for their drum machines, but that strikes me as a waste, both of money and of good hardware, and it adds a completely unnecessary level of complexity to the process. Here's what I want: a synthesizer module that looks like any other, except that it uses a slew of onboard 16-bit PCM samples as sound sources, and lets you adjust them for pitch, equalization, and envelope, and then stores those settings as MIDI notes and/or program changes.

I'll bet that with the savings on R&D, memory and all the extra buttons and displays that have to go into or on a conventional drum machine, some company could put out the world's most amazing percussion synthesizer for less than $500.

That's my list. For all I know, some elves in places besides the North Pole are already working on these very ideas. If so, they can make my holiday brighter by letting me know how they're doing.

R·E/P

Sound on the Road

By David Scheirman

For touring entertainment groups, the process of building an international audience, and then catering to it for a period of years, can be a long and difficult one. The right album at the right time, carefully calculated media appearances or just plain luck—all these factors work together to help create touring organizations that control every detail of a live show, from accounting to trucking.

As a touring organization matures and attempts to make sure that live performances will be consistent, one sure sign that the artist has *arrived* is a portable custom stage set.

Whether it is Bruce Springsteen or Barry Manilow, major touring artists go for performance comfort and a crowd-pleasing concert appearance by committing to the design and construction of portable, modular stages that are trucked to each venue and set up on a daily basis. While most auditoriums and arenas are equipped with in-house risers and staging that can be configured in several different ways, the more complex shows on the road prefer the streamlined and personalized custom stages.

Such stages can take months to build and are a significant expense. Only recently have staging designers realized that the entire touring package works much more smoothly when sound system requirements for a particular tour are taken into account *before* the stage set ever leaves the drawing board.

"It was very frustrating to try and cram all of the monitors on stage into tight little spots in the set at the last minute," related a touring sound technician who was recently faced with the need to integrate all of his mics, stands, cables and stage monitors into a hastily constructed custom touring stage within a week prior to hitting the road. Advance coordination between the stage designer, builders and the sound crew is beneficial to all concerned.

The most complex and innovative custom touring stages can be very costly. Although designing provisions in advance for the stage to contain hidden monitor speakers and cable harnesses does not add a significant amount to the overall package cost, it can mean the difference between a slick production and a big headache for the technical crew. Basic, modular arena stages that are designed for a particular artist's show regularly make it to the loading dock for several hundred thousand dollars; more extensive stages with sets designed around a particular theme and involving hydraulic lifts and other custom machinery cost several times that.

As stage design for outdoor and arena touring becomes a more widely developed art, greater attention is being paid to the needs of sound system and stage technicians. Integrating the various sound, lighting and band-gear production needs into the original design concept leads to a cleaner look, saving costly setup and teardown time.

As stage design becomes a more widely developed art, greater attention is being paid to the needs of sound system and stage technicians.

One major manufacturer of custom microphone cables and wiring harnesses has become known for working closely with stage designers to streamline the entire cable hookup system. This approach worked so well for Journey that the new stage for David Bowie's *Glass Spider* tour has been fitted with audiophile cables, fabricated and installed by the cable manufacturer.

Such costly and well-detailed custom touring stages are far beyond the reach of most touring organizations. It usually takes outside financial assistance to work up to one year ahead of time on the design and marketing concepts for a specific tour and its production needs. For example, a soft drink manufacturer is underwriting $7 million for the Bowie tour. Much of these funds went into the development of custom staging.

To be able to provide a significant advance technical input for a touring stage design, concert sound companies must be locked in to the project many months prior to the first show. Due to contract indecisions and competitive bidding, this is often not possible. When an artist does have a solid, long-term commitment to a touring sound company (such as Bruce Springsteen and Clair Bros., Barry Manilow and A-1 Audio, or Huey Lewis and the News with Sound On Stage), the entire production will flow much more smoothly. Stage monitor type and placement, deck PA stacking concepts and monitor mix positions can be planned well in advance.

Will the monitor-mix position be hidden, or in full public view? Will there be more than one? How much physical space is required for monitor system amplifier and processing racks? Can the power distribution system be of the quick-disconnect type that is easily integrated into the staging frames?

Are sidefills monitors required, and should they be stacked at stage level or flown overhead? Where on the stage deck are microphone input boxes required, and what type of floor slants will be used? In what positions?

These and many other questions often do not occur to anyone but the sound company involved with an upcoming tour. If such questions are asked in advance, and the answers diligently pursued *prior* to production rehearsals, the resulting concert effort will be significantly improved.

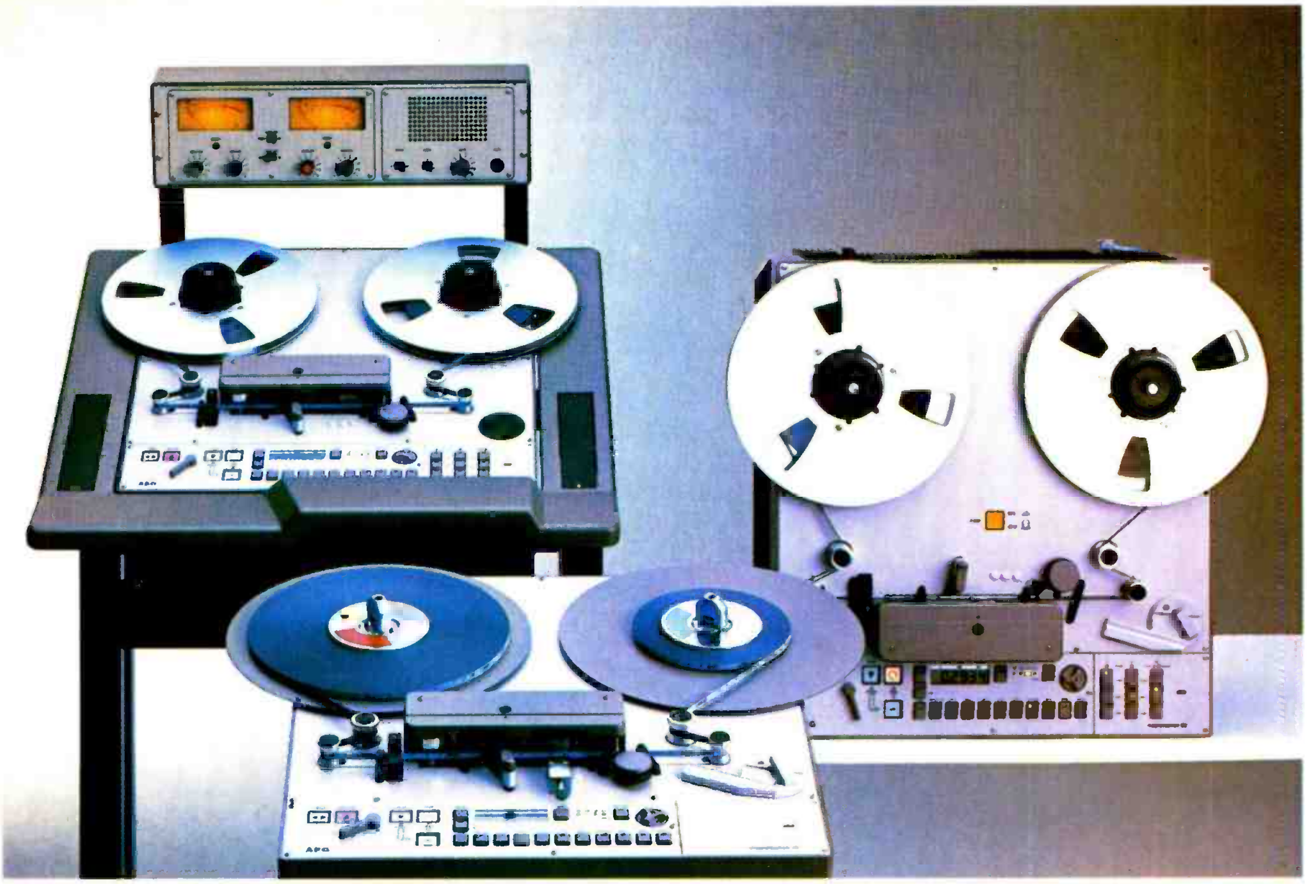
The decision of whether or not to design and fabricate custom sound equipment (whether monitor cabinets or wiring harnesses) to fit into a certain stage design concept has definite financial considerations for sound companies. If an artist wants floor monitors painted pink, or microphone cables that are gray, or a complex area-zoned, distributed sidefill wash loudspeaker configuration, will that particular project actually pay for the customer alteration of existing equipment? If new gear is purchased, will it be of the type that is useful to the sound company once that particular tour is over? Most companies find that the extra effort to please a long-term client is worth it.

Ultimately, the collaboration between custom stage designers and builders and concert sound companies leads to some innovative technical "fixes." Stage monitors that house low-frequency components beneath gridwork, and only HF components above stage level, or monitor-mix positions that are built right into rigid structures such as air cargo containers, can shave precious hours off the overall setup time required to get a major arena show out of the trucks and on the air.

When concert sound companies begin to collaborate with stage fabricators that are located in the same building complex or in adjacent neighborhoods, their working together on complex touring projects can create an atmosphere of cooperation and a synthesis of creative energy that can ultimately advance the state of the touring entertainment arts.

R·E/P

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



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Film Sound Today

By Larry Blake

As noted in my last two columns, only in the past decade has the process of securing high-quality sound effects recordings become SOP in the film industry. We have all learned a great deal about the requirements of effects recording: microphone and pre-amp selection, stereo techniques, editorial considerations, etc. (The early parts of our effects libraries are proof of how far we've come.) In addition, most of us record on ¼-inch tape, and we have more worn and noisy effects than we'd like to admit.

What we have here then is a group of people thinking the same thing: I want a library the size of my current one, but I want it *digital*. Well-recorded, no hiss, no wear: permanent.

What's stopping everyone, of course, is that such an effort is very expensive. (The other hurdle—a sensible field digital recording and editing medium—will be solved, I hope, with the introduction of R-DAT.)

The cost of building a 10,000-effect, 200-hour sound library from scratch (including a team of two recordists and two librarians) would be approximately $200,000, excluding expenses, computers, tape and equipment. It would also take at least a year.

A possible answer lies in a not-for-profit cooperative of studios, production companies, post-production facilities and sound editorial houses splitting the costs under the umbrella of a non-profit organization. This way almost everyone wins: We'd all have, in a short period of time, a sound effects library that would be beyond the means of any one company and would provide all of us with a foundation for the future. (No presumption is made that anyone will or should throw away old effects, nor that they will not record their own effects for in-house use only.)

The most pressing *raison d'etre* for such a sound library would be that it could document, in an orderly and quality fashion, the sounds of this planet. Every day something happens or disappears that cannot be recreated: What did the San Francisco streetcars sound like before they were remodeled? What sound was made when placing a long-distance call in 1930? 1950? How has the sound of Times Square on New Year's Eve, the floor of the New York Stock Exchange, Mardi Gras, etc., changed over the years? Restaurant walla, sports events...the list is huge and obvious.

Everything points to the fact that we should take nothing for granted, and today's sounds *must* be documented lest tomorrow's archives be incomplete. Of course, we should also place priority in obtaining best-generation recordings of the sounds of our past.

Therefore, it's almost mandatory that such a project be done worldwide. Having branches in the major cities of the world will not only improve the quality of the library (no one knows the sounds of a city better than a native), but it will also reduce the costs tremendously. Not to mention that such a project will bring together film sound communities worldwide, while providing everyone with access to fresh recordings of previously hard-to-get sounds.

We'd all have, in a short space of time, a sound effects library beyond the means of any one company.

It is hoped that having a single, unified front for sound effects personnel of the world will allow one-time access to usually restricted areas. I volunteer to do a complete series at the White House, including the presidential limo (fast passbys, with and without Secret Service men riding on the back) and Air Force One (which is soon to be replaced by a hot-rodded 747—see what I mean?).

What follows is the procedure I envision. The obvious candidates for branch offices are Los Angeles, New York, London, Paris, Stockholm, Moscow, Beijing, Tokyo, Toronto and Sydney. Each branch would be required to record, edit and cross-reference so many hours of sound each month. The original field tapes would be logged according to the recordist.

If this tape contained diesel train passbys, it would be added to the end of the "Trains—Diesel" edit master tapes for the New York branch. All pertinent information would be entered into the computer database, identical in field structure and lexicon in all branches. English would be used everywhere to avoid translation problems.

Updates of all edit master R-DAT cassettes that have been added to during each month are mailed to all other branches, which in turn, are responsible for local distribution to all "subscribing" companies. Updating is facilitated by each company owning two copies of all edit master cassettes: an in-house master for daily use and one that is stored at the local branch.

For example: the Los Angeles branch receives an update of New York's "Trains—Diesel" tape, which adds 10 minutes to the hour-long, in-house copy that the Los Angeles companies are currently using. A computer disk containing the revised New York database is also sent. Members also have "Trains—Diesel" tapes from all branches worldwide.

The Los Angeles branch will then copy the new material onto the end of the New York "Trains—Diesel" tapes that it holds in its vaults. Once the signatory companies receive their updates, they send their now outdated tapes back to the local branch.

The bulk of the tapes distributed by each branch would be recorded by people hired by the branch, and paid by the subscription fees. Signatory companies would not be required to share effects that they record on their own, although there would probably have to be a "brownie point" system to provide an incentive for companies to share and share alike.

Not only has R-DAT made this project a technical reality and practicality (high-quality, small and inexpensive storage costs, no-loss copying), but the era of personal computers will allow all companies to share not only the sounds but also the all-important data.

Smaller companies that perhaps cannot afford to subscribe can also have access to the sounds in the library by using a modem to search the current database, and then paying the local branch a fee for a copy of each effect used. (Or, a CD-ROM of the database could be updated annually.) This would be a one-payment buyout, in acknowledgement of the difficulty of policing each usage. (Could you imagine an ASCAP or BMI for sound effects?)

At first glance, this might seem like a pretty idealistic, pie-in-the-sky idea. In some respects perhaps it is, but I can't think of a more practical and inexpensive way for each of us to have acces to a comprehensive data bank of the sounds of Earth. If you would like to participate in such a project, please write to me directly at P.O. Box 288, Hollywood, CA 90078. R·E/P

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

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There are 18 different effect types in all, with up to nine parameters for each. So you can create precisely the sounds you want, while saying goodbye to outboard gear and their multiple data conversions. The DMP7's mixer has eight input channels, and its digital cascading feature lets you connect additional DMP7s to add more inputs (up to 32), as you require.

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The simplicity of the diagram below belies the many capabilities of the DMP7 at work in an automated multi-track mixdown/processing system.

Dry tracks from the multi-track recorder are fed into the DMP7. A SMPTE track is fed into the SMPTE/MIDI converter. The MIDI timing track controls the MIDI sequencer/recorder.

The MIDI control information for the DMP7 can now be recorded in sync with the music. The DMP7 then automatically mixes everything down into a two-track master or demo. Until now, this level of mixed automation was only possible with more expensive consoles.

The DMP7 doesn't just do the final mixing, but the final processing as well, to each individual track. With reverbs, flanging, delay, and stereo panning. Or whatever sweetening you need. And again, in sync with the music.

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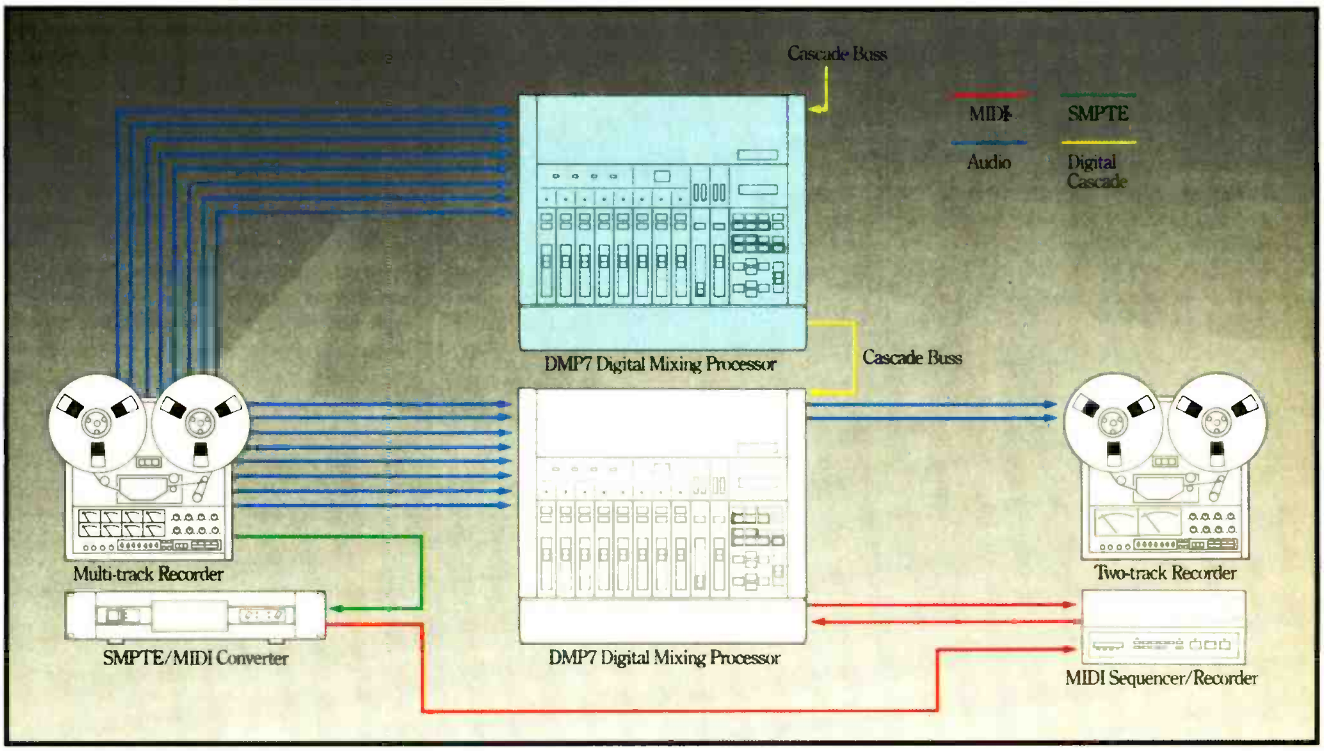

Living with Technology

By Stephen St. Croix

I write this month's column on a new hyper-powerful, full-screen, triple-drive, high-speed laptop computer: Living with Technology, I guess. I also feel that it's important to point out that I write this month's column as I sit on the ground in a 767 in a holding area off in the back 40 of Boon County Airport, Kentucky, for a surprise 3-hour wait (this does not count the previous 2-hour wait at the gate) due to an "ATC request for some unknown reason."

I live in Baltimore and am traveling to Los Angeles, so Boon County is not where I had planned to be tonight: Living with Technology, I guess. I could have caught the evening flight that departs Baltimore five hours after I left; it will probably arrive in LA slightly before I do. Technology is tricky stuff these days.

I could have spent those extra five hours out at the pool. I guess I actually *would* have spent the time fixing the three synthesizers that failed last week (repairing technology).

(As an interesting side note, on the return flight 10 days later, we sat on the ground for another nice wait because "the computer that takes care of weights and balances was down." The pilot had to make this announcement twice, by the way, because "the computer that takes care of announcements went down" in the middle of the first announcement. Really. I am not making this up.)

We use so much technology today that we are often immersed in 20 or more basic technologies at the same time. Many of these technologies were revolutionary when first introduced. Several were amazing a few very short years ago, yet we cannot imagine doing our jobs without them now.

There really is a danger in all this, although it is unavoidable for those of us who feel the need to compete: Loss of Touch. I used to find myself occasionally zooming through life on or with a chunk of technology, only to realize that I had no real idea of how I got to this point.

Let me attempt to clarify using a rather extreme example (one of my favorite things to do): Suppose they gave a war and everybody came. One big nuclear-generated electromagnetic pulse and technology as you know it would instantly cease to exist. Let's say that they didn't even give a war, just a few of these EMPs, carefully placed.

Again, all technology that you live with today and every day is gone. You, the land and the buildings would be there, but there would be zero electricity (not that it would matter; every electrical device would be smoke city). No automated 48-channel, SMPTE-locked, MIDI-chased, hands-off mixes. No tapes at all, no disks; no work.

Could you survive? Statistically, no. You would die because you had lost all touch with the foundation, with how the world really works; you would find yourself Living *without* Technology. (Actually, this scenario doesn't even take into account no phone, radio, cars, planes or lights. No pumps to get water out of wells; no way to ship food.)

We use so much technology today that we are often immersed in 20 or more basic technologies at the same time.

OK, I'll back off to a somewhat less severe scenario, and offer a more directly applicable example. Sometimes it seems to me that people forget what they are really working with, and sometimes what they are working for. I have heard a few too many tunes that sound like they exist solely because of and for technology. It seems that somehwere decisions were made for the use of some particular piece of monster gear, at the expense of the music itself.

I have also seen with my own eyes one situation in a well-known studio that brought this point home for me. At the audio "witching hour"—that magic time of day, on that magic day of the week, once every week, when you hope nothing breaks because you won't be able to get it fixed or replaced in time to finish the session—a sequencer loaded with a very impressive lead went to Mars, and took the lead along for the ride.

The artist, well-known that he is, became white and weak, sat down and hoped that the night tech on duty could resurrect both the machine and the music. It did not happen to turn out that way (which is the other part of the magic of the witching hour).

Faced with the concept of having to actually *play* the lead over again, live and in real-time direct to rolling tape, he lost it all, consumed the remainder of his recreational pharmaceuticals and actually cancelled the session.

We are not talking stage fright here, or even the newer phenomenon of "tape fright." I'm talking about the inability to function without technology. With the help of technology that this player had begun to take for granted as part of the process, he had been playing beyond his real capabilities for over a year. He had been using the sequencer as a safety buffer; to edit and repair sloppy playing (sort of like how I am using the word processor as I write this column).

I realized then that every time I sit down at a synthesizer keyboard, even in the privacy of my own studio, I make a decision about whether or not I can play this part directly to tape, or if I should record it into a sequencer first so that I can "fix it in the bits."

If you accept my premise that we sometimes use technology to produce a product that is dependent on that very technology—that is, the product would not exist without the technology, as opposed to merely being helped or produced faster because of the technology—then you can see that in some compositional situations, the creation process for music may not simply be aided, but in fact totally changed.

If this change is fully understood by the artist, and he remains in control at all times, that's great—we all get to hear the work weeks sooner.

If, on the other hand, the change is part of the work and the artist is not aware of this, then the piece has been modified to fit the constraints of, let's say, a software sequencing package. This can be a tragedy.

While it can be argued that the creation of music for all time has been subject to the constraints of actual musical instruments, a few rational humans would take the stand that a sequencer qualifies as such.

This is what I am pontificating about: the difference in using high-technology tools to help us in the creation of music, and the tools using *us*. As we become more dependent on these tools we may lose sight of the very musical idea that made us sit down at the computer in the first place.

Oh, did I tell you that the music that they played for us while we sat on the runway was sequenced? R·E/P

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

SPARS On-Line

By Gary Helmers

The state and health of educational opportunities for careers in entertainment audio have never been better. More programs than ever are now available from public and private institutions, and more are starting up each academic year. The Audio Engineering Soceity and various trade publications list directories of schools offering everything from short workshops and courses, to bachelor and masters degrees with emphasis in sound recording and music technology.

Ever since the first 4-year degree program in the United States was started by Billy Porter (now at CU-Denver), educators have used a modification of the European "tonmeister" program to wed music and technology studies into college degrees or certificate programs.

When Geoffrey Wilson edited the first AES Directory of Educational Programs in 1979, there were about 24 baccalaureate degrees in audio-related studies, and about 10 in music business studies worldwide. The 1984 AES Directory lists more than 40 technology programs, and another 40 business programs.

The majority of these are in music schools of public and private institutions or in trade schools. As the number of new programs increase, there is more growth shown among the private universities and colleges compared to public tax-assisted schools.

When course content of the 1979 program is compared to the content of 1984, an expansion of video, maintenance and computer applications become obvious. The study is becoming multidisciplinary in nature, and schools that adapt to changes in the industry with qualified instructors and with proper equipment continue to grow.

New course titles in the study program reflect the growing complexity of the audio field, including such courses as video production, recording studio calibration and maintenance, music and the personal computer, MIDI and others.

Very few 4-year engineering programs have taken up the challenge of audio education. The majority of the training programs are located in "Mass Communications,"—as at Middle Tennessee State University—or in schools of music, such as Berklee, SUNY-Fredonia, University of Miami, Lowell and CU-Denver.

The equipment needed to support these new courses is complex, expensive and sometimes hard to come by. Where a school offering audio courses used to get by with technical support common to a radio-production facility, it now has to equip on a scale equal to the best record and video sweetening studios.

Multitrack analog and digital control rooms, analog and digital synthesis studios and video sweetening studios are required to provide instruction and hands-on laboratory experiences. It has become increasingly difficult for a school to catch up—let alone keep up—with the growing complexity of the equipment that is common to the average audio studio.

In a young discipline feeding a young industry, audio education in this country has gone through some growing pains.

Finding and keeping qualified instructors for this curriculum has presented a new set of challenges to program directors. Professional experience, academic qualification and salary are constant competitors when gathering a faculty. The trend is to rely more than ever on part-time staff, where available. This solution offers the advantage of greater flexibility in pay and scheduling, and brings active professionals into the classroom, but leaves these teachers out of planning, advising and budgeting.

Typical programs now have found a mixture of full- and part-time instructors necessary to balance this need for instructional staff. The key resident faculty are still faced with promotion and tenure issues that vary in complexity, but the surviving programs have found ways to ensure that these important specialists are appropriately recognized in the academic community.

To attract a new generation of students, many music departments in public and private schools have added courses in audio. While many of these institutions do not grant a degree major in audio, they provide a good introduction to the technical complexity of the art, and interested students can then proceed to one of the major programs for later career training in audio.

Those schools that have not mounted major audio programs are also finding it difficult to provide equipment for instruction and labs. There seems to be a minimum level of program commitment to gain critical mass in equipment and faculty.

The AES has provided a forum for educators to meet and compare progress in educational issues. However, there has not been any organized effort to standardize curriculum or faculty preparation during the infancy of this teaching field.

When SPARS wrote its National Studio Exam, it placed an understandable emphasis on engineering and maintenance items. Most students in music schools found themselves weak in technical training when they took this exam.

The SPARS Internship Program has also become an important link from the classroom to entry-level jobs. Because some major employers, as recent as five years ago, were not interested in college graduates, the SPARS program has helped create new access to internships from the quality 4-year schools.

In addition, the growth of MIDI studios has provided professional quality, pre-production capability to more musicians than ever; such changes will also need to be reflected in the curriculum of 4-year programs.

An increase of industrial participation in the educational process seems logical. The high cost and quick turnover of pro-audio technology makes it appropriate that manufacturers work closely with quality programs. Long-term loans, grants and gifts will be necessary to assure state-of-the-art facilities for training.

In a young discipline feeding a young industry, audio education has gone through some growing pains; there still exists plenty of room for improvement and continued growth. A healthy attitude of cooperation exists between competing academic programs, and genuine help is available from the industry.

I would like to thank Roy Pritts, acting resident dean of the College of Music, University of Colorado at Denver, for his invaluable assistance in the preparation of this article.

(**Postscript:** Due to an oversight, I forgot to thank Dave Porter, owner of Music Annex Studios, Menlo Park, CA, for his essential input in the May *On-Line* column dealing with studio construction. He just completed a new facility in San Francisco, and generously shared his research and experience.) R·E/P

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

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Stereo Microphone Evaluations:

An Overview of Techniques and Subjective Assessments

By Lowell Cross

From the results of four previous evaluation sessions at The University of Iowa, what can we learn about stereo techniques and the perception of qualitative differences between professional microphones?

The series of microphone evaluations appearing in **RE/P** during the past three years began as a demonstration for my recording students. Not content with classroom discussions of the audible properties of our own collection of microphones, I decided to make multichannel recordings with them and with others on loan, all in stereo, from which A-B comparisons could be made.

The recording students, as well as members of our faculty and staff, have heard and learned to appreciate the subtle (and sometimes not so subtle) qualities of more than 40 studio models via the 24-track recording medium.

Lowell Cross is professor of music and director of recording studios at the University of Iowa, Iowa City, IA, and a regular contributor to **RE/P**.

However, in agreeing to publish the results of our evaluations, I have come full circle, and then some: I am still attempting to describe the "sound" of professional microphones, but now in print—and to a much larger and more diversified audience than the students in a recording class.

Inevitably, the person who learned the most from this process was the one who initiated it. It was a rare personal privilege to be able to make stereo recordings with the industry's most revered microphones, and to become acquainted or reacquainted with them first-hand. The masters from these sessions have been retained in our tape archive; they are exceptionally valuable resources for our audio recording courses. (With a little advance notice, **RE/P** readers traveling through the Midwest could obtain a cordial invitation to visit our studios in Iowa City for an audition session of A-B microphone comparisons.)

Foremost among my own learning experiences from these experimental sessions has been the discovery of the world of ambisonics. My impressions of the Soundfield system loaned for evaluation in May 1984 were so positive that I put one at the very top of our equipment "wish list." Eventually, we were able to place an order and the new microphone arrived in late May 1986, during the writing of this article.

Patch-panel rewiring and installation of the system in our main control room is now complete. That control room is equipped with four reference loudspeaker systems for monitoring horizontal sound fields. The construction of an 8-loudspeaker "full-sphere" or "periphonic" monitoring array and supporting electronics is planned for the very near future, allowing true 3-dimensional reproduction—including height information.

I cannot conceal my enthusiasm for the ambisonic technique, not just because of its advanced or exotic applications, but because it reveals so much to us about 2-channel stereo.

Our initial exposure to ambisonics in 1984 prompted another comparison beyond that of evaluating individual microphones: an examination of various stereo techniques. Just as all engineers have their own favorite microphones, each of us usually maintains a strong preference for one of the ways of recording in stereo. Advocates of the spaced-omni technique, for example, shun those engineers who favor coincident placements, and *vice-versa*.

I admit that my own biases about 2-channel stereo recording were quite well developed before having the opportunity to make direct comparisons. However, I gained a healthy respect for two methods that I had not tried before: ambisonics, and the artificial-head technique, that uses equalized pressure transducers for "ears."

Yet another learning situation presented by our evaluations concerns the matter of microphone "coloration." While I would agree that the pursuit of absolute accuracy in audio recording should remain a cherished ideal, I also believe that our field is enough of an "art" that there always will be strong motivations—and quite justifiable ones—for engineers to devise techniques for embellishment and heightened illusion. The forms of enhancement obtainable from certain microphones that we evaluated, including some of the older large-diaphragm models, were to my ears pleasant and ingratiating. The special attractions of these vintage units, for which no one would dare to make claims of great accuracy, have caused them to become collector's items.

Our subjective approach has been a major factor in all of these articles. Readers of the previous four reports, published in the April 1984, December 1984, February 1985 and December 1985 issues, will recall that the averaged responses received from our groups of listeners provided one method for "ranking" the review microphones. The individuals who participated in our playback sessions were members of my recording classes, other students in the School of Music, and faculty and staff colleagues. All participants have received some form of ear training, either in an audio or a musical sense.

But, as the author/perpetrator of this exercise, I have given the most emphasis to my own impressions. Except for instances in which stereo technique played a role in determining the perceptions of individual microphone qualities, as in the case of widely spaced omnidirectional or PZM units, the averaged ratings of our listening groups have been generally consistent with my own conclusions.

The central issue in these evaluations has been the perception of qualitative differences among professional microphones, as heard under actual studio conditions. Accordingly, specific makes and model numbers have received close scrutiny. In the long run, most of our findings have been reassuring rather than surprising. Those brands which consistently have received favorable ratings in our listening tests were already well known and highly regarded throughout the audio community. One measure of the present health of this profession is the fact that so many brands of fine microphones are now available worldwide.

So this is the overview that I wish to present here: a look at what has been experienced, learned anew, and reconfirmed during an extended period of living with many different brands and models of high-quality microphones. The topics to be addressed are ambisonics and stereo, colorations and accuracy, and subjective choices and preferences. All of these elements contribute to the very basic question, "How does one choose a microphone?" Which, of course, has no single answer.

Ambisonics and stereo

For the last 15 years, I have experimented with discrete 4-channel quadraphonic recording techniques. It became increasingly clear to me that these recordings, mostly involving front and rear pairs of microphones, did not measure up to expectations. They were characterized by an obvious time delay between front and rear, producing unnatural spatial effects that would never be heard in a concert hall.

I realize now that my basic premise in making them was just plain wrong—I was following the popular misconception that surround-sound reproduction over four spaced loudspeakers requires a recording from four spaced microphones, or some similar arrangement. This incorrect premise is dying a natural death, just as "quad" records already have.

Basic mathematical and acoustical research into the complex phenomena of multidimensional hearing—carried out notably in England, where Alan Dower Blumlein conducted his pathfinding stereophonic investigations in the Thirties—has led to ambisonics, an elegant solution to the problems of 2- and 3-dimensional sound recording and reproduction. Credit for this work goes to Michael Gerzon and Peter Fellgett, whose research has been supported since the Seventies by the National Research Development Corporation in the United Kingdom, and to earlier investigators such as the Japanese acoustician Y. Makita, Peter Scheiber (United States), and the team of Duane Cooper and Takeo Shiga (United States/Japan).[1]

The tangible realization of ambisonic recording is here today in the form of a well-known, but expensive and complex, microphone. Engineers who have used it and who have studied its properties are aware of the truly comprehensive range of possibilities that it offers.[2]

Ambisonic techniques have motivated us to re-evaluate what is meant by the expression "stereophonic sound." If it were not for the fact that 2-loudspeaker reproduction became known as "stereo" before the advent of ambisonics, then the latter could have assumed that name. (Our present-day descriptor, *stereo*, comes from the Greek word *stereos*, meaning solid, i.e., 3-dimensional.) Ambisonics is *not* extended stereo; the various coincident-stereo techniques are subsets of ambisonic technology.

Stereo, of course, means many things to many people, ranging from "something that comes over two loudspeakers or a Walkman," to the panpot techniques of multichannel recording (an example of "intensity stereo") to the intended results from specific microphone choices and procedures. The last includes the various classic techniques: spaced omnis, near-coincident placements (ORTF, NOS), and the coincident methods (which are also forms of "intensity stereo," including MS, cardioid XY and Blumlein).

Even the very "best" stereo techniques can only approximate a reconstruction of the original sound event, by creating an illusion that the event is taking place between and behind the loudspeakers (*not* in 3-dimensional or even 2-dimensional space; both possibilities are included within the ambisonic concept).

In his excellent paper, "Stereo Microphone Techniques: Are the Purists Wrong?," Stanley P. Lipshitz never quite arrives at a definition of "stereo" in unequivocal terms, but he does shed light on the matter, by stating the following:

"I believe that the best that stereo can do is to provide a credible illusion that between and beyond the pair of loudspeakers there exists another acoustic environment within which the musicians are located and performing. I am considering only musical events which have a genuine acoustic origin within an acoustic setting, a situation which pertains to all classical musical performances and a great many others besides. I accept that not all music originates under such conditions."[3]

Later in the article, Professor Lipshitz reaffirms the premise that ambisonics is the only available technique that can accurately reconstruct an acoustical event in a listening environment different from the original one. The following statements are extracted from the section of his paper entitled "What's wrong with stereo?":

"...*stereo setups usually use a subtended angle* [between the loudspeakers, at the listening position] *of less than 90°—60° is more common, leading to a summing error of just over 1dB.*

"This error can be eliminated if one has loudspeakers surrounding the listener, and is correctly handled in the ambisonic surround-sound system. Since in this system...both pressure (W) and velocity (X, Y, Z) signals are independently available, suitable loudspeaker feeds are synthesized so that the correct pressure-velocity relationship is regained at the listener's head. This requires the sympathetic behavior of all the loudspeakers for a center-front source....

"... *The ambisonic system can correctly handle both direct and reverberant signals whereas stereo cannot. And herein lies the basis for its extreme naturalness and precision in...imaging, depth and ambience.*"[4]

There is another surround-sound technique proposed by a leading Japanese manufacturer. This proprietary system, known as Q-Biphonics, is supposed to take into account the one ingredient missing from Gerzon and Fellgett's realization of ambisonics: arrival-time differences between the ears.[5] The information required to preserve inter-aural delays is obtained from a system employing two artificial heads, one forward-facing, one rear-facing, separated by a baffle.[6]

Q-Biphonics is a near-coincident system; ambisonics is coincident. In fact, great pains have been taken to make the Soundfield microphone as close to *absolutely* coincident as possible. The exact coincidence of the capsules is electronically synthesized, although they are, by necessity, slightly separated spatially.

Close up detail of microphones used during stereo evaluation sessions.

I would like to agree with Professor Lipshitz' prediction that ambisonics "is the way of the future." But we must recognize that most engineers and producers are going to continue to work in the 2-channel domain of stereo—at least for the present. Fewer than 200 Soundfield microphones have been sold worldwide; about one-third of these are in North America. The Q-Biphonics system has yet to achieve any popularity outside of Japan. So, to conclude this section, we return to the more universal topic of stereo.

Culminating in our May 1984 evaluation sesson, my experience has been that certain near-coincident and coincident techniques come closest to approximating the goals of "good" stereo. I have never favored spaced omnis (which, to my ears, produce an amorphous, diffuse and unfocused result) or coincident XY cardioids (because of their oppressive "center buildup" effects). Of course, for *any* "illusory reconstruction" to work, everything must be correct throughout the program chain: matched, in-phase components; acceptable recording and reproducing environments; and low distortion, low noise, broadband equipment.

The MS technique offers "safe" and predictable results. Because the side (S) component drops out when the resultant post-matrix left and right channels are summed, leaving only the middle (M), MS is properly mono-compatible. (This feature also explains why center buildup is not a problem in MS.) Furthermore, the stereo microphones associated with this technique have variable patterns for both capsules, allowing a wide range of options.

I still advocate near-coincident placements as a practical, satisfying approach to stereo recording, in spite of my recent immersion into the ways of ambisonics. Quite apart from the fact that near-coincidence can be a far less expensive method than any of the costly MS/XY stereo microphones, ORTF and similar techniques *can* yield excellent imaging and localization, perfectly acceptable mono compatibility and, in my opinion, a more satisfying illusion of depth perspective than certain MS and XY procedures.

In an earlier discussion in this series, I expressed my preference for near-coincident cardioid techniques over the purely coincident ones, especially MS and cardioid XY, because of this perceived "added dimension" of depth perspective. However, 2-channel stereo never can be more than a *1-dimensional* affair, defined by the line between the loudspeakers. The illusion of stereo depth exists, of course, but it comes from phase-difference and reverberation cues.

This illusion is comparable to the one discovered by Italian painters of the Renaissance, who learned how to achieve depth, or vanishing-point perspective, on a flat surface. (Stereo has often been called "auditory perspective.")

Stereo is definitely an improvement over mono, which is *zero-dimensional* (i.e., from a single point source). To achieve the goals of true 2- and 3-dimensional recording, ambisonics is the only technique now universally available. The Q-Biphonics system is conceived as only a 2-dimensional system, with none of the height information available from ambisonics.

Colorations and accuracy

It is a big step, and perhaps one in a backward direction, to jump from the topic of ambisonics to a discussion of the colorations present in some of the industry's favored microphones. The ambisonic system is, indeed, one of the most "uncolored" and natural-sounding available, by very deliberate design. Not only does its impressive array of electronic processors achieve such goals as absolute coincidence of the capsules, but also provisions are included for frequency-shaping circuits that take into account the psycho-acoustical criteria required to synthesize the original wavefronts at the ears of the listener.

Yet however successful recent advances have been in overcoming the frequency- and phase-response, noise and distortion problems of transducer design, microphone coloration remains an important tool available to the engineer.

Our evaluations confirmed that there are three manifestations of coloration: as little as possible, good and bad. ("Good" is not really *better* than "as little as possible," but it is surely better than "bad.") Even the most uncolored of microphones—in my experience, limited exclusively to a few condenser models—exhibit *some* coloration. But, whatever minute levels of coloration that such microphones may individually possess, they still may be characterized as having a "condenser sound."

I admit to fostering a preference for condenser microphones over all other types; the remarks that follow are observations from one who has very positive views about condenser principles, technology and products. The colorations

that I hear from "condenser sound" are partly electrical in origin; those heard from ribbon and moving-coil designs are purely mechanical by comparison. As noted, I prefer the former to the latter, but I still find even the *least* colored "condenser sound" to be a bit unnatural. I have heard only one or two condenser microphones that did not have some suggestion of high-frequency emphasis; the experience was so unusual that I suspected that something was wrong.

Capsule resonances, determined by the materials, dimensions and tensioning of diaphragms, as well as reflections in and around head screens and microphone bodies, produce the acoustical/mechanical components of "condenser sound." Added to these conditions are the workings of tube or solid-state circuity (including other capacitors, transformers and the requirements of polarizing voltages for the capsules).

It is the sum total of these elements which bring about the electrical sounding colorations, however subtle, that I am describing. At worst, the situation gets quite out of control, producing the gritty, spitty sound that accounts for the ongoing interest in the better ribbon and moving-coil designs.

Overly bright, strident sound, resulting from uncontrolled high-frequency resonances, exemplifies the "bad" category of colorations heard from condenser microphones. The mechanical resonances, associated with dynamic units (ribbon and moving-coil) occur at various points across the audio spectrum, often at several places simultaneously. The result is a roughness in response that has been variously described as "tinny," "brittle," "boomy" or "tubby."

The now obsolete RCA 77-DX ribbon microphone, in use for decades in the U.S. audio industry, can serve as an example of some of these problems. While it continues to command respect in certain quarters, the 77's rather pronounced resonances in the 200Hz and 2akHz-6kHz ranges—in combination with a rapidly decreasing output above 10kHz cause its colorations to be unpleasant to my ears, if not actually bad. (An appreciation of the challenges of designing microphones without coloration-producing resonances, whether condenser, ribbon, or moving coil, may be obtained by reading Dr. Gerhart Bore's *Microphones*, pages 56-62. He summarizes the associated problems of transient-response and offaxis colorations on pages 69-70[7].

If "as little as possible" coloration from the highest-quality condenser units still leaves a "suggestion of high-frequency emphasis," (my phraseology), and if "bad" coloration is the result of uncontrolled resonances, poor transient response and poor off-axis and pattern characteristics (or a combination of all of these problems), then what is "good" coloration? In my opinion, extremely few microphones possess such a quality—and I repeat my admonition that they are not necessarily "better" products than those with a *minimum* of coloration.

The ability of a microphone to provide enhancement is a more elusive property than even accuracy. "Musicality" may be another way of expressing this nebulous attribute. Again, the only microphones to which I can ascribe this distinction are certain condenser models. They are noted for a wide frequency range, professional-quality noise and distortion specifications, and either the capability of electrically variable patterns or a special polar characteristic that has been designed to change with frequency. Their appeal arises from an *atypical* condenser sound. They behave graciously in that part of the spectrum where many others are becoming increasingly more aggressive: the range of greatest human hearing sensitivity.

Accordingly, my choices for microphones with "good" colorations are those that approach the goal of accuracy, yet have smooth, non-aggressive (or even withdrawn) midrange and high-frequency qualities that are attractive to the ear. Even though they are condenser models, these are units that approximate, or even surpass,the ideal "ribbon sound"—with those longed-for qualities of smoothness and warmth.

Author Lowell Cross with microphones from the University of Iowa collection (left-to-right): RCA 77-DX (on boom), AKG C452EB, Shure SM81, Neumann U47 (tube), Neumann SM69fet and Calrec MkIV Soundfield. RTW 1260 stereo correlation meter and Calrec MkIV control unit are shown on the studio's Neve 5315/24 console.

Choosing a microphone: subjective preferences

There are many factors that determine the choice of a microphone, both at the time of purchase and prior to a recording session. Some of these elements relate to purely practical matters: cost, availability, reputation of the manufacturer and distributor, susceptibility to damage, physical appearance and/or ease of concealment, and complexity of operating features. Other considerations (which are more pertinent to this discussion) concern the intended application and the user's expectations for the final result.

The parts of the decision-making process that deal with applications and expectations are, of course, the most difficult, because they are invariably *subjective*. No amount of reading published specifications, studying reports of anechoic chamber and/or time-delay spectrometry tests, or even conducting such tests, will allow anyone to predict *exactly* how microphones are going to sound under the wide range of conditions imposed upon them in actual use.

I completely agree that empirical testing and reporting make up a very significant part of our collective learning processes. I also believe that first-hand experience makes up at least an *equally* significant part—and that is why I undertook all of these evaluations for myself and for my students. I urge **RE/P** readers to compare microphones on their own, so that they will become better informed before a final decision is required for a given application.

Our experimental sessions have been undertaken with a deliberate choice of "classical" repertoire. Music is clearly an end unto itself; the act of recording is merely a means to that end. Therefore, music is the single most important ingredient in our evaluations.

Classical music has been our vehicle, for several reasons. My colleagues and I are trained in it, it is readily available at our school of music, and the concert halls in which we record are suitable for its performance. It is also the most "neutral" music available for our purpose; other genres, be they jazz, rock, pop or country, have too many unknowns and connotative aspects in an electro-acoustical sense to be suitable. How are they supposed to sound *without* audio equipment?

Classical music has its primary, and original, existence outside of the recording medium; it is historically a performing art with established traditions, rather than one which requires reproduction or synthesis to come into being. We already *know* how it is supposed to sound.

Microphones that serve classical music well may not always be the appropriate choices for certain rock, pop or commercial requirements. Even though we have concentrated on vocal reproduction, we have not investigated any hand-held models. These are invariably associated with solo (mono) source material; our approach has centered on stereo.

But because pop vocal recording is such an important part of the daily routine of the industry, what have our evaluations revealed that may be useful in those applications? Certainly, studio microphones that are accurate, or which may "enhance" the voice to some extent, are worthy of serious investigation.

MS and ambisonic techniques for recording solo vocalists are very appealing—they create additional space around the voice, yet retain mono compatibility.

A microphone setup for vocal and piano recordings.

Finally, we should remember that certain models have actually *defined* the sound of today's recorded pop vocals. A few European, large diaphragm condenser units, both tube and solid-state, have had a sonic and an economic impact on the audio profession. Many engineers prefer to continue their personal successes with these famous microphones, especially since the clients themselves often demand them. They add their own characteristic "stamp" to vocal sound, in varying degress of increased presence and brightness (mid- and high-frequency emphasis).

I agree that these models are capable of enhancing vocal quality when such forms of emphasis are desirable, as in pop music. In this sense, they too exhibit the elusive properties of "good" colorations. R·E/P

Notes and references

1. For an account of the evolution from "quad" to Ambisonics, see: William Sommerwerck, "Ambisonics Comes of Age," Parts I, II, & III, *The Audio Amateur*, Nos. 3, 4, & 5 (1984). Letters from John Roberts and William R. Crawford and author's replies, *TAA*, No. 5 (1984) and No. 2 (1986).

2. References to Ambisonic recording are found throughout the *Journal of the Audio Engineering Society* in recent years. An important early article is Michael Gerzon's "Periphony: With-Height Sound Reproduction," *JAES*, Vol. 21, No. 1 (Jan.-Feb. 1973), pp. 2-10.

3. Stanley P. Lipshitz, "Stereo Microphone Techniques: Are the Purists Wrong?" Audio Engineering Society Preprint No. 2261 (D-5), p. 6.

4. *Ibid.*, pp. 37-38.

5. Sommerwerck, *op. cit.*, Part II, p. 39.

6. John Eargle, *Handbook of Recording Engineering*. New York: Van Nostrand Reinhold, 1986, pp. 108-110.

7. Gerhart Bore' *Microphones for professional and semi-professional applications* (transl. S.F. Temmer). Berlin: Georg Neumann GmbH, 1978. (This important publication is now available in limited quantities from Gotham Audio Corporation in New York.)

I wish to thank our audio engineers, David Muller and Peter Nothnagle, for their assistance in the preparation of this article.

All photography by James J. March and Linda Bourassa.

An Engineer's Guide to Studio and Live-Performance Microphones

By Bruce Bartlett

To achieve high-quality results in the studio and in live-performance situations, you need a basic understanding of the various types of available microphones.

Which type of microphone is best suited for recording a symphonic band? What's a good piano mic? Should the microphone be a condenser or dynamic model, with an omni or cardioid pick-up pattern?

Such questions can be better answered once you understand the various types of microphones and their specifications. After reviewing the definitions in this article, you should have a better idea of which microphone should be used in a particular application.

A microphone is basically a *transducer*—a device that converts one form of energy into another. Specifically, a microphone converts acoustical energy (sound) into electrical energy (the output voltage).

Transducer types

Microphones for recording and production applications can be grouped into two types, depending on their operating principle: *dynamic* or *condenser*. In a dynamic microphone, a metal wire moves through a magnetic field to produce an electrical signal. The two types of dynamic microphones are moving-coil and ribbon models.

A *moving-coil* microphone (more popularly called a dynamic mic) is shown in Figure 1. A coil of wire attached to a diaphragm is suspended in a magnetic field. When sound waves vibrate the diaphragm, the coil moves within the magnetic field, generating an electrical signal similar to the incoming sound wave.

In a *ribbon* microphone, a thin metal foil or ribbon is suspended in a magnetic field; sound waves vibrate the ribbon in the field and generate an electrical signal.

Bruce Bartlett is senior microphone development engineer and technical writer at Crown International, Elkhart, IN.

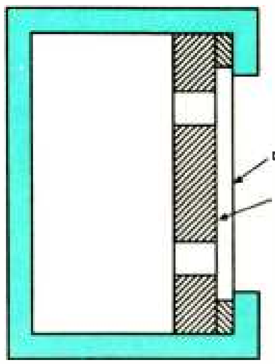

Figure 1. *Section of a typical dynamic or moving-coil microphone.*

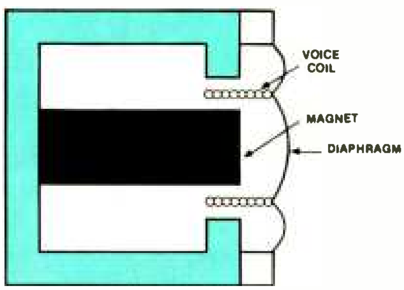

Figure 2. *Section of a typical capacitor or condenser microphone.*

In a *condenser* or capacitor microphone (Figure 2), a conductive diaphragm and an adjacent metallic disk (backplate) are charged to form two plates of a capacitor. Sound waves striking the diaphragm vary the spacing between the plates; this alters the capacitance and generates an electrical signal similar to the incoming sound wave.

The diaphragm and backplate can be charged either by an externally applied voltage, or by a permanently charged electret material in the diaphragm or on the backplate.

Because of its lower diaphragm mass and higher damping, a condenser microphone responds faster than a dynamic microphone to transients.

A condenser microphone generally provides a smooth, detailed sound with a wide frequency response. Working with a good condenser microphone, you can capture all the "ping" of the cymbals, or the plucking of each string in a strummed guitar chord. This clear, detailed sound quality makes the condenser microphone especially suitable for micing cymbals, snare drums, acoustic instruments and studio vocals.

A condenser microphone requires a power supply to operate, such as a battery or external phantom-power supply. Simplex phantom power is 12Vdc-48Vdc applied to pins #2 and #3 of the microphone connector through two equal-value resistors; the microphone receives phantom power and sends audio signals on the same two conductors. Many mixing consoles supply direct phantom powering at their mic input connectors.

By contrast, a dynamic microphone works without any power supply and provides a reliable signal under a wide range of environmental conditions. A well-designed dynamic mic is quite rugged and can accept very loud sound pressure levels without overloading, an ability that suits it for micing guitar amps and drums.

Because a dynamic mic generally has a slower transient response than a conden-

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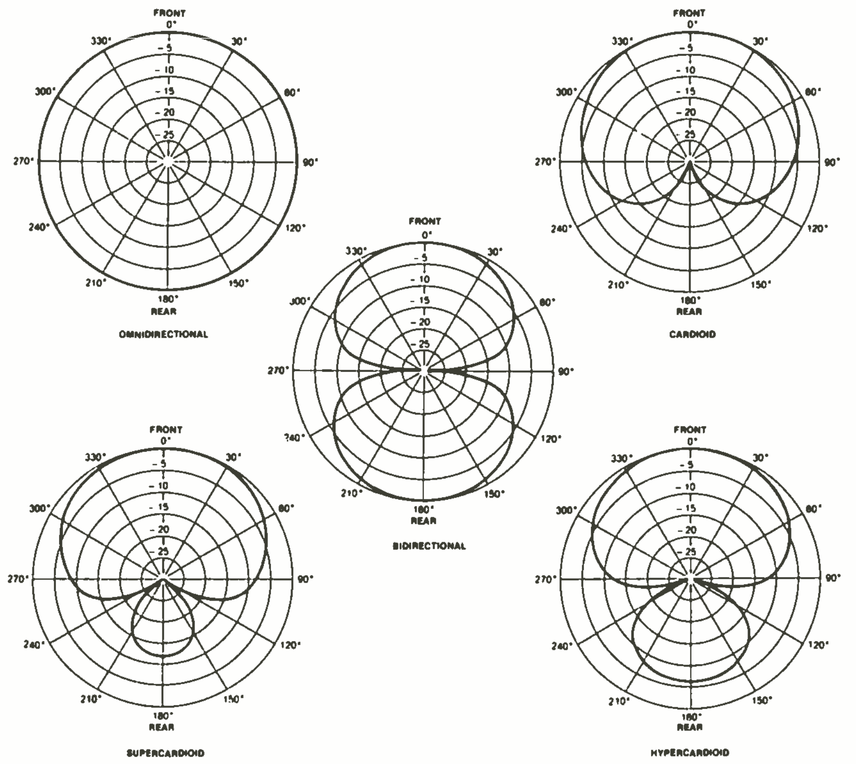

Figure 3. *Polar pick-up patterns of the five types of response curves offered by microphones.*

Omni or directional?

Omnidirectional microphones have one or two characteristics that make them especially useful for certain applications.

Use an omnidirectional when you need:

- *All-around pickup.*
- *Pickup of room reverb.*
- *Low sensitivity to explosive breath sounds.*
- *Low handling noise.*
- *No proximity effect.*
- *Extended low-frequency response (in condenser models).*
- *Lower cost in general.*

Use directional microphones when you need:

- *Selective pickup.*
- *Rejection of room acoustics, background noise and leakage.*
- *Up-close bass boost.*
- *Better gain-before-feedback in a sound reinforcement system.*

ser, it can be used to soften the fine detail the latter would pick up. A flat-response dynamic might be a good choice for woodwinds or brass, if you want to take the "edge" off the sound. Dynamics generally have a rougher response than condensers or ribbons, although moving-coil models of excellent quality are available.

Ribbon microphones, while usually considered to be more delicate than the dynamic type, are often prized for their warm, smooth tone quality. Typically, they are used on brass instruments to mellow the tone.

Polar pattern

Microphones also differ in the way that they respond to sounds coming from different directions. An *omnidirectional* microphone is equally sensitive to sounds arriving from all directions. A *unidirectional* microphone is most sensitive to sounds arriving from one direction—in front of the microphone—but discriminates against sounds entering the sides or rear of the microphone. A *bidirectional* microphone is most sensitive to sounds arriving from two directions—in front of and behind the microphone—but rejects sounds entering the sides. Figure 3 shows various polar patterns.

The unidirectional classification can be further divided into cardioid, supercardioid and hypercardioid. A microphone with a *cardiod* pattern is sensitive to sounds arriving from a broad angle in front of the microphone. It is about 6dB less sensitive at the sides, and about 15dB to 25dB less sensitive at the rear.

The *supercardioid* pattern is a 8.6dB down at the sides and has two nulls of least pickup at 125° off-axis. The *hypercardioid* pattern is 12dB down at the sides and has two nulls of least pickup at 110° off-axis. This pattern has the most rejection of leakage and room reverberation of the three types.

Because they discriminate against sound to the sides and rear, cardioids help to reject unwanted sounds such as room acoustics (reverb), feedback or leakage (off-mic sounds from other instruments). Cardioids are the most popular choice in the studio for this reason, because they provide good isolation or separation between recorded tracks.

Most unidirectional and bidirectional mics produce a bass boost when used within a few inches of a sound source. This proximity effect occurs in directional microphones that have a single distance between the front and rear sound entries.

The "warmth" created by proximity effect adds a pleasing fullness to drums. In most recording situations, however, proximity effect lends an unnatural "boomy" or "bassy" sound to the instrument or voice picked up by the mic. To minimize proximity effect, some microphones are specially designed, while others feature a bass-rolloff switch to compensate for the low-frequency boost. Alternatively, you can roll off the excess bass at the console until the sound is more natural. By cutting the bass response you will also reduce low-frequency leakage picked up by the mic.

Condenser or dynamic models are available with any kind of directional pattern. Ribbon mics, on the other hand, are either bidirectional or hypercardioid.

Figure 4 classifies the various microphones according to transducer type and polar pattern.

Frequency response

Frequency response is the range of frequencies that a microphone will reproduce at an equal level, and is normally quoted within a tolerance window, such as ±3dB. If an accurate or natural sound is desired, the mic's frequency response should cover the frequency range of the instrument. For example, a trombone radiates frequencies from about 80Hz to 8kHz. Select a mic with a response covering at least this range. Similarly, an orchestra produces a very wide frequency span ranging from about 40Hz (bass drum and bass viol) to 15kHz or higher (cymbals and other percussion).

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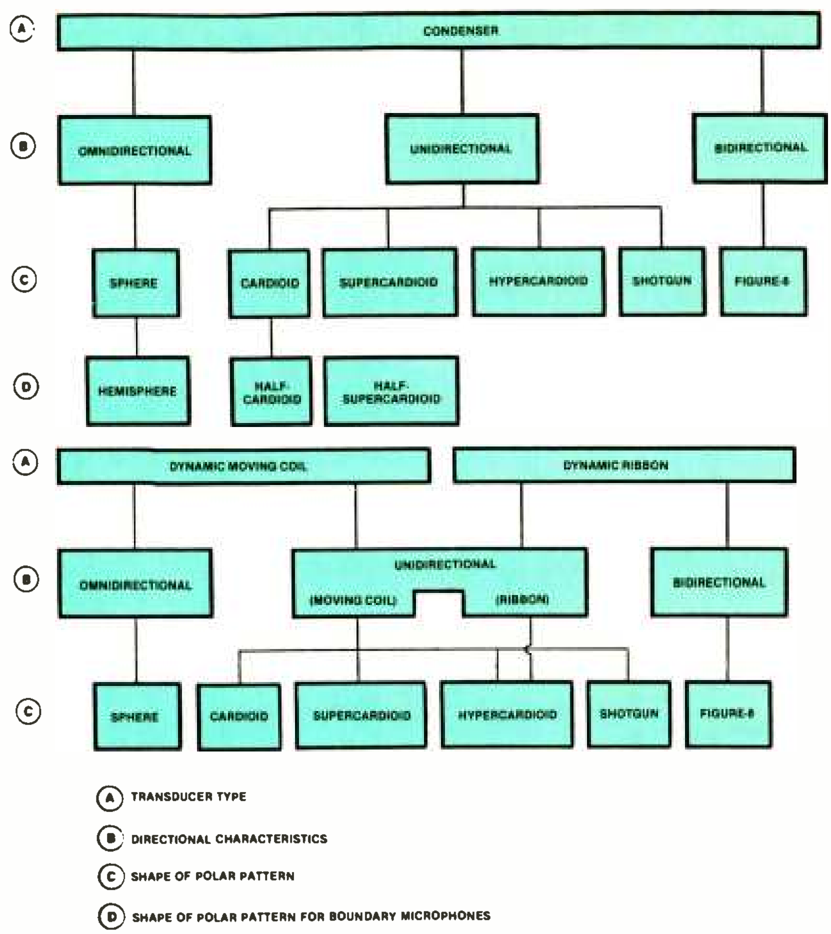

Figure 4. *Classification of various studio and live-sound microphones according to transducer type and polar-pattern responses.*

A frequency response from 80Hz to 15kHz is adequate for most instruments; a response from 40Hz to 9kHz covers the range of bass instruments; a high-end response up to 12kHz is sufficient for brass, voice and piano; and a response out to 15kHz or 20kHz is necessary for cymbals and some percussion instruments.

The low-frequency response of the microphone should be limited, if possible, to the lowest fundamental frequency of the instrument to be recorded. For example, the frequency of the low-E string on an acoustic guitar is 82.41Hz. A mic used on the acoustic guitar should roll off below that frequency to avoid picking up low frequency noise and room rumble. Some microphones provide a low-frequency cutoff switch for this purpose; alternatively, you can filter out the unneeded lows at the console.

A frequency-response curve is a graph of output level in decibels at various frequencies. For a microphone, the output level at 1kHz is placed at the 0dB line on the graph, and the levels produced at other frequencies are so many decibels above or below the reference level.

The shape of the response curve usually indicates how the microphone sounds at a distance of about 2 to 3 feet. For example, a microphone with a flat, extended response reproduces the fundamental frequencies and harmonics in the same proportion as the sound source. Thus, a flat-response mic tends to provide accurate, natural reproduction at that distance.

A microphone with a rising high-end or a presence peak around 5kHz to 10kHz emphasizes the higher harmonics as shown in Figure 5. The subjective effect is a crisp, articulate sound. (This type of response is sometimes called a "tailored" or "contoured" response and is popular for guitar amps and drums because it adds punch and emphasizes attack.)

Sensitivity

Sensitivity is a measure of a microphone's efficiency. A very sensitive microphone, for example, produces a relatively high output voltage for a sound source of a given loudness.

Sensitivity doesn't affect its sound quality; instead it affects the audibility of console noise. To achieve the same recording level, a low-sensitivity mic requires more gain than a high-sensitivity model. More gain usually results in more noise.

If you record quiet, distant instruments, such as a classical guitar or chamber music, you'll hear more console noise with a low-sensitivity mic than with a high-sensitivity model, all other factors being equal. With close-miced pop music, however, sensitivity matters little because the microphone signal level is well above the console noise floor.

Microphone sensitivity is often stated in "dB re:1 volt per microbar." The figure provides an indication of the voltage produced (in decibels relative to 1V) when picking up a 1kHz tone at a 74dB sound pressure level. Listed below are typical sensitivity specs for the three transducer types:

Condenser: –65dB (high sensitivity).

Dynamic: –75dB (medium sensitivity).

Ribbon or small-diaphragm dynamic: –85dB (low sensitivity).

Differences of a few decibels among microphones are not cricital.

The louder the sound source, the higher the signal voltage produced by the microphone. Extremely loud sources, such as kick drums or guitar amps, can cause a microphone to generate a signal strong enough to overload the console mic preamp—which is why input attenuators or pads are included in mixers to reduce mic signal level.

Impedance

A mic's impedance is its effective output resistance at 1kHz. An impedance between 150Ω and 600Ω is considered low; 1kΩ to 4kΩ is medium and above 25kΩ is high. Low-impedance microphones are preferred for recording because they allow long cable runs to be used without hum pickup or high-frequency loss. Nearly all consoles are designed to accept low-impedance mics.

Maximum sound pressure level

Another specification is maximum sound pressure level (SPL), a measure of the intensity of a sound. The quietest sound we can hear—the threshold of hearing—measures 0dB SPL. A normal conversation at 1 foot measures about 70dB SPL; painfully loud sound is above 120dB SPL.

Maximum SPL is the point at which a microphone's output signal starts to distort; usually the SPL at which the microphone produces 3% total harmonic distortion. If a microphone has a maximum SPL spec of 125dB, it means that the microphone starts to distort audibly when the sound pressure level produced by the source reaches 125dB SPL.

A maximum SPL spec of 120dB is

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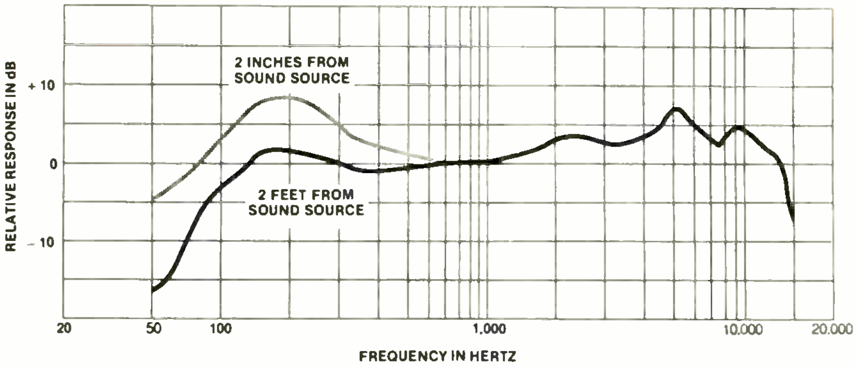


Figure 5. *An example of the way in which a microphone's frequency response can be affected by proximity effect and presence peak.*

Table I. *Microphone selection chart.*

Requirement	Recommended mic characteristic
Natural, smooth tone quality	Flat frequency response
Bright, present tone quality	Rising frequency response
Extended lows	Omnidirectional, multiple-D cardioid or single-D cardioid with bass rolloff
Extended highs (detailed sound)	Condenser
Reduced "edge" or detail	Dynamic
Boosted bass up-close	Single-D cardioid
Flat bass response up-close	Omnidirectional, multiple-D cardioid, or single-D cardioid with bass rolloff
Reduced pickup of leakage, feedback and room acoustics	Unidirectional, or omni up close
Enhanced pickup of room acoustics	Omnidirectional, or unidirectional farther away
(1) Micing close to a surface (2) Even coverage of moving sources or large sources (3) Inconspicuous mic	Boundary mic or miniature mic
Coincident or near-coincident stereo	Unidirectional mic or stereo mic
Extra ruggedness	Dynamic
Reduced handling noise	Omni or unidirectional with shock mount
Reduced breath popping	Omni or unidirectional with pop filter
Distortion-free pickup of very loud sounds	Condenser with high maximum-SPL spec, or dynamic
Noise-free pickup of quiet sounds	Low self-noise, high sensitivity

good, 135dB is very good and 150dB is excellent. Any well-designed dynamic microphone can handle SPLs in excess of 150dB SPL.

Self-noise

Self-noise or equivalent noise level is the electrical noise that a mic produces, equivalent to what a sound source would produce in dB SPL. This figure is usually A-weighted, meaning that the noise was measured through a filter that rolls off low and high frequencies to simulate the ear's frequency response. A self-noise spec of 20dB SPL or less is excellent; a spec around 30dB SPL is good and a spec around 40dB SPL is fair.

Polarity

Most mics produce a positive voltage at pin #2 with respect to pin #3 when the sound pressure pushes the diaphragm inward (a positive pressure). If a microphone is wired in the opposite polarity, and combined to the same channel, low frequencies in the sound pickup are attenuated or completely cancelled.

To prevent this from happening, check that all your studio or PA mics are wired identically, as follows:

1. Choose one microphone as a polarity reference.
2. Plug it into your console and talk into it from about 3 inches away, and set the meter to peak around zero VU.
3. Do the same with a second microphone and cable-plugged into another input channel.
4. With both microphones mixed to the same output bus, hold the mics together and talk into them again at a distance of 3 inches.
5. If the meter reading is lower, the polarity of the second mic or cable is reversed with respect to the reference.

In that case, remove the connector shell from the second mic's cable and reverse the connections to pins #2 and #3 (in one connector only). Use only that cable with that microphone, and label it.

If you can remove the connector in the microphone itself, reverse the connections to pins #2 and #3 for mics that are opposite in polarity to the reference. (Check a few mics before doing this to make sure the reference mic itself isn't wired backward!)

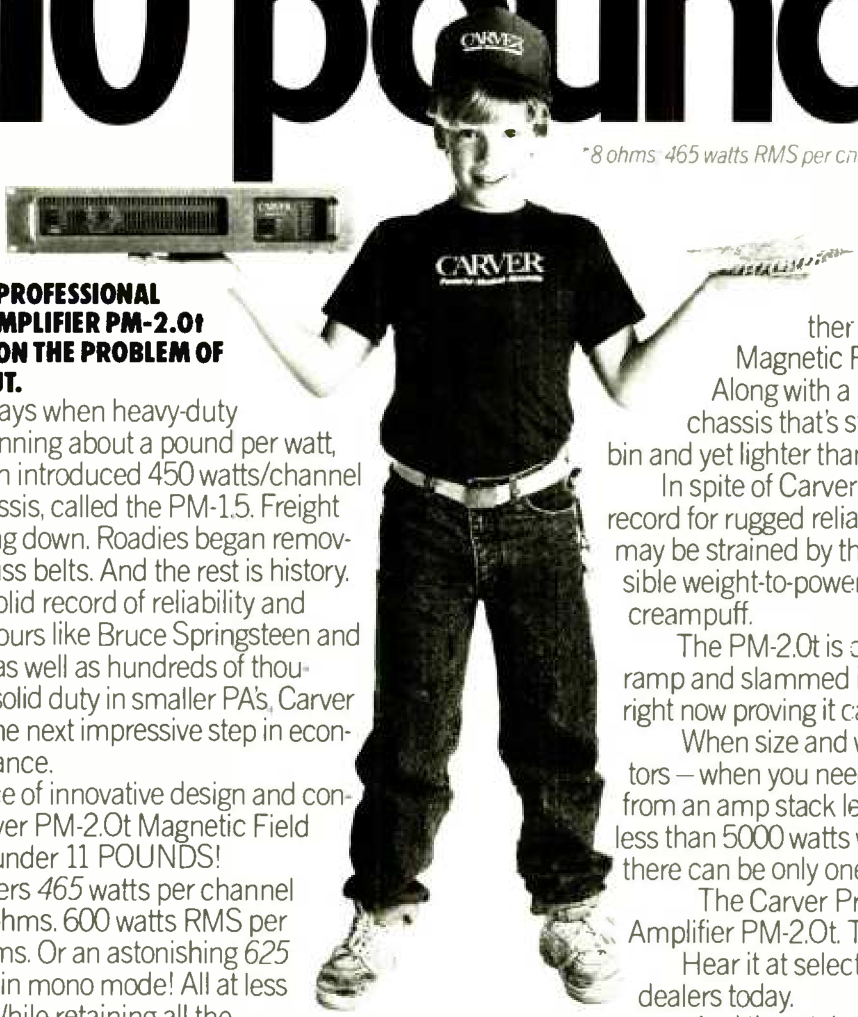
Special microphones

The following section describes three types of microphones used for special purposes. These are boundary microphones, miniature microphones and stereo microphones.

A *boundary microphone* is designed to be used on a surface such as a floor, wall, table, piano lid, baffle or panel. As can be seen from Figure 6, a typical boundary

465 watts 10 pounds.*

*8 ohms: 465 watts RMS per channel. 10 lbs., 12 oz.



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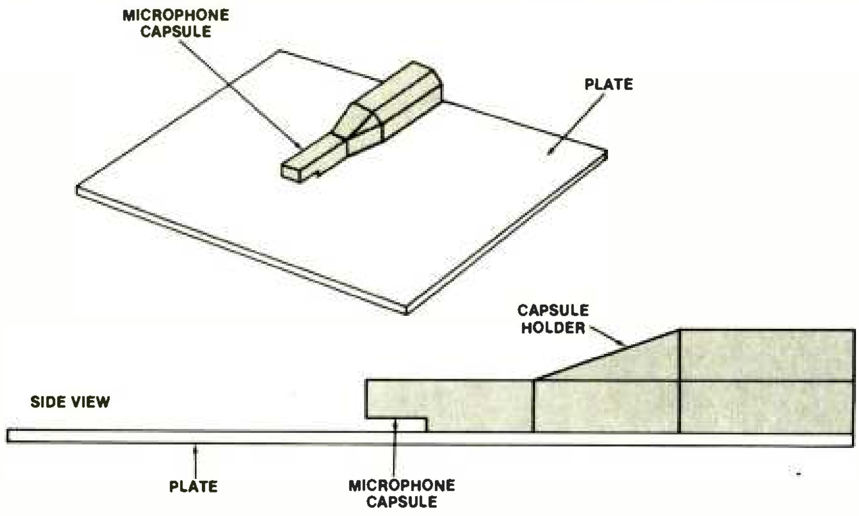


Figure 6. *Construction of a typical boundary microphone.*

microphone includes a miniature electret condenser capsule mounted face-down next to a sound-reflecting plate or boundary. The mic diaphragm now receives direct and reflected sounds in-phase at all frequencies, avoiding phase interference between them.

The claimed benefits of such mics are a wide, smooth frequency response free of phase cancellations, excellent clarity and "reach," a hemispherical polar pattern and uniform frequency response anywhere around the microphone.

In the studio, a boundary microphone is typically taped to the underside of a piano lid or to the wall for pickup of room ambience. It can be used on hard baffles between instruments, or on a panel to make it directional.

Boundary microphones are also available with a unidirectional polar pattern. They offer the benefits of both boundary mounting and the unidirectional pattern, and are well suited for lecterns, news desks and stage-floor pickup of drama or musicals.

Another special type of microphone is the *miniature condenser* type, which can be attached to drum rims, flutes, horn, guitars and so on. With miniature units, you can mic a live band for recording without cluttering up the stage with boom stands. In some instances, two or three mics can cover the small drum set.

A *stereo microphone* combines two directional mic capsules in a single housing for convenient stereo recording. The mic is placed about 10 feet to 15 feet in front of a band, choir or orchestra. Because there is no spacing between the mic capsules, there will be no delay or phase shift between their output signals. Consequently, stereo microphones are mono-compatible: the frequency response is the same in mono and stereo.

Taking into consideration the various microphone characteristics mentioned previously, Table 1 comprises a microphone selection guide based on specific requirements.

Let's run through some specific examples to see how Table 1 is used. Suppose you want to record a grand piano playing with several other instruments. You need a microphone that reduces sound leakage, so you find this requirement in the left column. The chart recommends a unidirectional mic or an omni mic up-close.

For this particular piano, you want a natural sound, for which the chart suggests a mic with a flat frequency response. You also want a detailed sound, so a condenser mic is the choice. A microphone with all these characteristics is a flat-response, uni condenser mic. If you are micing close to a surface (the piano lid), a boundary mic is recommended.

Now suppose that you're recording an acoustic guitar on stage, where the guitarist moves around. For a moving sound source, the chart recommends a miniature microphone attached to the guitar. Since you're micing close, feedback and leakage are not a problem; so you can use an omni mic.

You try an omnidirectional condenser mic. On this particular guitar, it sounds too detailed (too much pick noise and string noise). You want a less-detailed sound, so you finally choose a miniature omnidirectional dynamic mic, which represents a good choice for this particular application.

There is no one correct microphone to use on any particular instrument; you should choose the microphone that sounds best to you. Quality recordings, however, always require quality microphones with a smooth, wide-range response, low noise and low distortion. R·E/P

Microphone accessories

• ***Pop filters*** *are a necessary accessory for a vocalist's microphone. When a vocalist sings a work emphasizing "p", "b" or "t" sounds, a turbulent puff of air is forced from the mouth. A microphone placed close to the mouth is hit by this air puff, resulting in a thump or pop; the use of a windscreen or foam pop filter reduces this problem. Some microphones have pop filters or ball-shaped grills built into the mic.*

Pops are also reduced by placing the vocalist's mic above or to the side of the mouth, or by using an omnidirectional microphone.

• ***Stands and booms*** *are adjustable devices that hold the mic and let you position them as desired. A microphone stand has a heavy metal base that supports a vertical pipe. At the top of the pipe is a rotating clutch that lets you adjust the height of smaller telescoping pipe inside the larger one. The top of the small pipe has a standard 5/8-inch-27-inch thread, which screws into a microphone stand adapter.*

A boom is a long, horizontal pipe that attaches to the small vertical pipe. The angle and length of the boom are adjustable. The end of the boom is threaded to accept a microphone stand adapter, and the opposite end is weighted to balance the weight of the microphone.

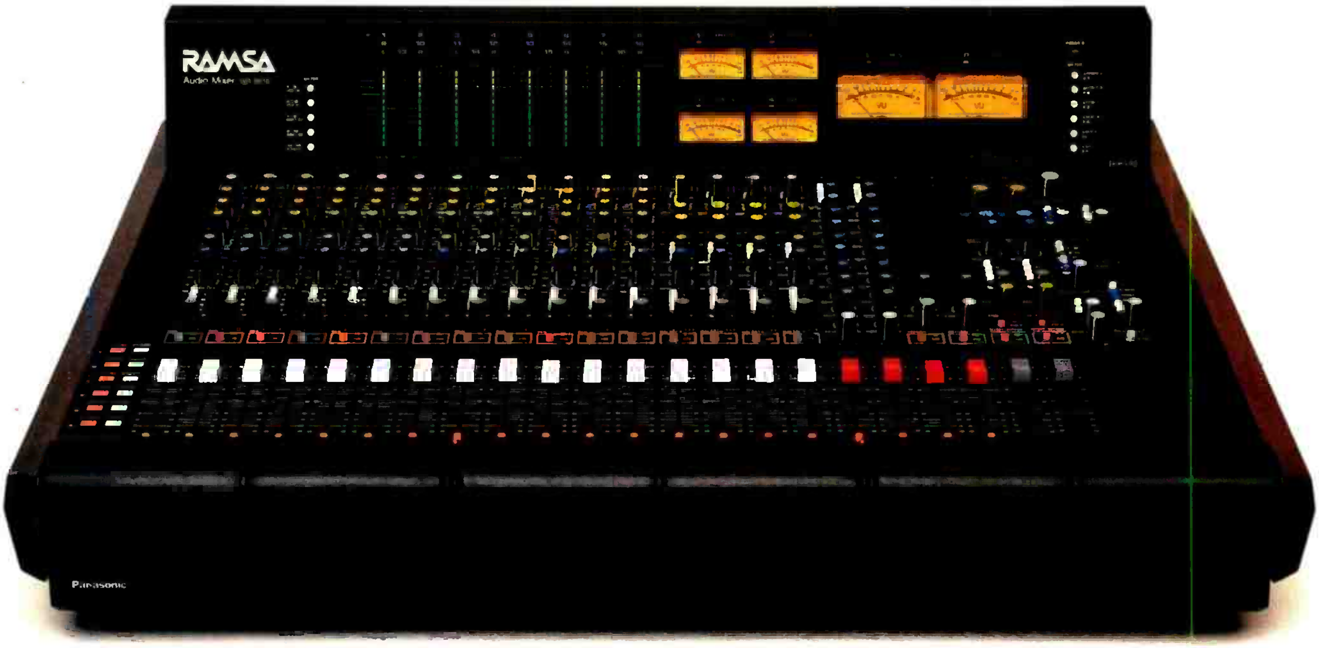
• ***Shock mount*** *is a device that mounts on a mic stand and holds it in a resilient suspension to provide isolation from mechanical vibrations, such as stand and floor thumps. Many mics have an internal shock mount that isolates the capsule from its housing; this reduces handling noise as well as stand thumps.*

• ***Cables and connectors*** *carry the electrical signal from the mic to the console or tape machine. The cable is made of one or two insulated conductors surrounded by a fine-wire shield to keep out electrostatic fields that can cause hum.*

• ***Junction boxes and snakes*** *reduce the number of individual cables running from many microphones to a console. Instead, you can plug all studio or on-stage mics into a junction box with multiple connectors. A single multiconductor cable (the snake) carries the signals to the mixer, where the cable divides into several connectors.*

• ***Splitters*** *are important if you are recording a live band, and you need each mic to feed a signal simultaneously to your recording console and the band's PA mixer. A transformer splitter has one input for each mic, and two or three isolated outputs to feed each mixer.*

This article has been excerpted from the author's book, **Introduction to Professional Recording Techniques**, and is reproduced with permission of the publisher, Howard W. Sams & Co. Copyright 1987.



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The Art and Science of Close-Field Monitors

By John Eargle

Practically every recording and production studio is equipped with close-field monitoring to allow an engineer to determine how the resultant mix will sound on domestic-style speakers, and to provide a detailed analysis of the stereo balance. What selection criteria exist for such monitors, and where should they best be located in the control room?

Most studio control rooms have elaborate built-in monitor systems that are bi- or tri-amplified and carefully equalized to match a given response contour. As good as these systems are, they are often turned off in favor of a pair of small, 2-way loudspeakers, usually mounted on the console's meter bridge.

The large monitor loudspeakers may be the reference during original recording and overdubbing but, when it gets down to the fine detailing of the final stereo mix, it is a good bet that the little ones will be used. The reason for this is simply that, in a highly competitive music marketplace, neither producer nor engineer wants to leave any detail unchecked when it comes to the actual balances that the public will hear.

The practice is known by several names, including *near-field monitoring*, *close-field monitoring*, *free-field monitoring* and *direct-field monitoring*. Because Ed Long and Associates has a trademark on the term Near Field, we will use the term close-field in describing this kind of monitoring.

Some definitions

What do these terms mean in the first place, and why do engineers like to monitor a mix in this way?

The term *near-field* has a precise meaning in acoustics, and has been used for many years. Imagine any kind of acoustical source in an environment completely free of reflections. An anechoic chamber will do, or we can simply locate the source on a tall pole outdoors.

John Eargle is president of JME Consulting Corporation and a regular contributor to **RE/P**.

As we approach the source from a sufficient distance, we will observe that each time we halve our distance, the sound pressure level increases 6dB (inverse square law).

But, the closer we get to the source, we begin to notice that the level does not quite double with each halving of distance. Instead, it varies in a rather unpredictable manner. At that point, we are in the near-field of the source.

The companion term is *far-field*, which describes the range over which inverse square relationships are applicable. As a practical matter, we can state that if the observer is located at a distance more than four times the longest transducer array dimension of a loudspeaker, the listener will be effectively in the far field of that loudspeaker.

Thus, for a single 5-inch cone loudspeaker, an engineer located three feet away will be well into the far field. But, if the loudspeaker is a 2-way design with an 8-inch woofer and a dome tweeter, the longest transducer array dimension is about one foot, and the listener will be in the transition region between near fields and far fields.

With normal console distances, it is clear that near or close field, whichever term is used, may or may not apply, depending upon the size of the loudspeaker in question. In some cases, the large soffit-mounted monitors may occupy so much surface space that engineers are really located in *their* close field when seated at the console!

Technically speaking, a more accurate term may be free-field monitoring, because that term defines a region, independent of near and far fields, in which direct sound from the loudspeaker predominates over reflected sound. This condition is probably the one that most engineers would agree is ideal. However, let's stick with close field as our operant term.

Advantages of close-field monitoring

An obvious advantage of playing a trial mix over a small pair of close-in loudspeakers is that those loudspeakers are probably limited in how loud they can play. As a result, the engineer and producer are forced to monitor their product at a lower level, and certain loudness level spectral changes may become significant. (Recall the Fletcher-Munson or Robinson-Dadson equal loudness contours.)

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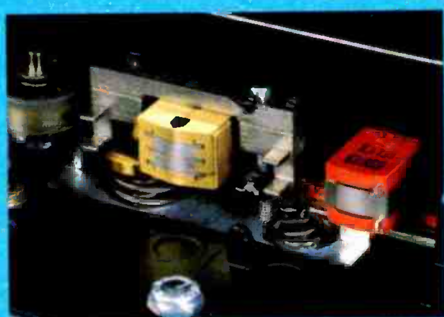
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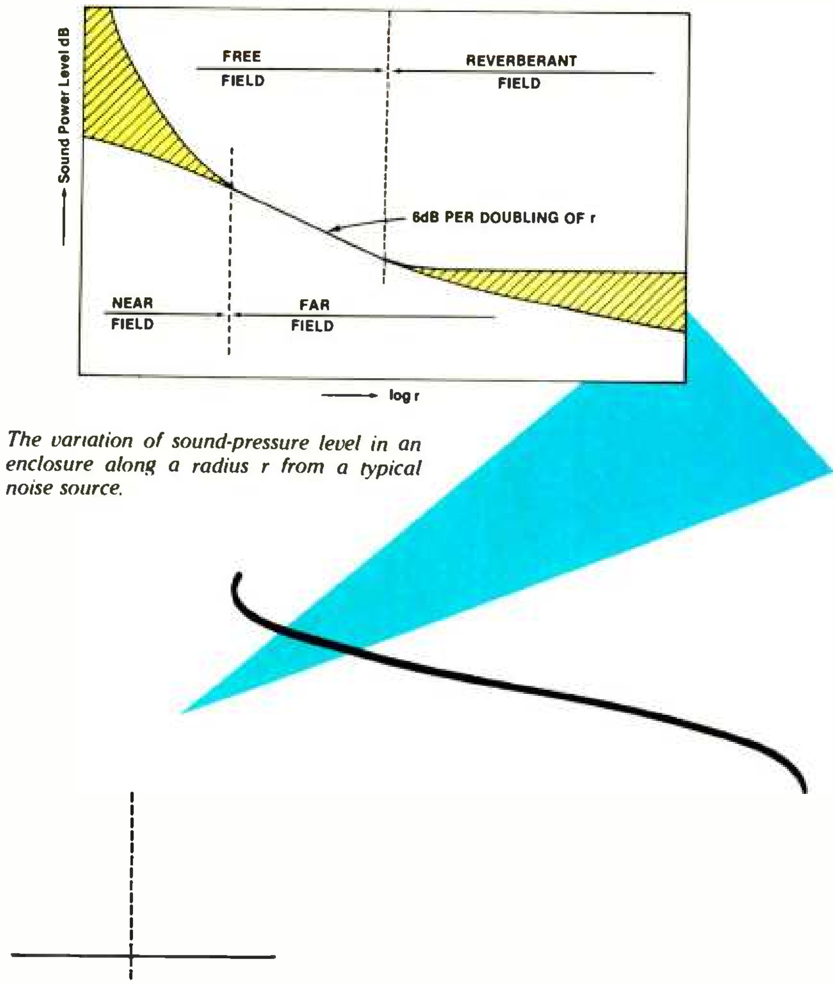
More on sound fields

In the accompanying illustration, the relation between near and far fields is shown in the lower legend. As stated in the main text, when the observer is more than about four times the longest transducer array dimension away from a loudspeaker, the observer is moving into the far field. That transition point is shown by the vertical dashed line to the left in the drawing.

The far field extends from that point outward, and the transition region between near and far fields is strictly a function of the dimensions of the sound source.

The upper portion of the figure shows the familiar relationships between the free (direct) field and the reverberant field. The transition region between these two fields is shown by the vertical dashed line to the right in the drawing, and that region is defined by both source directivity and room absorption.

Most control rooms do not have a reverberant field, statistically speaking. However, there can be considerable indirect sound from the big monitors at the engineer's position, by way of side- and back-wall reflections. In some control rooms, the reflected sound may be just about equal in acoustical level to the direct sound from the big monitors, and if those monitors are large enough, the listener might find himself in both the near field of the loudspeaker and in the reflected field of the room. Things are not always simple. (Diagram from Beranek, Noise and Vibration Control, *page 141. McGraw-Hill, New York. 1971.)*



The variation of sound-pressure level in an enclosure along a radius r from a typical noise source.

larger, soffit-mounted monitors, but that might not tell the whole story. Small, close-field monitors will undoubtedly be bass-shy compared with the larger ones, and that will contribute to bass imbalances too.

In any event, the producer and engineer will gain a better idea of how their product will sound over small sets and auto stereo systems in the field. This is particularly important to know, especially with regard to apparent bass and vocal balances.

Another important aspect has to do with the complexity of the mix. With today's digital recorders, superb microphones and high-resolution monitor loudspeakers, a very complex mix can be produced that sounds excellent over the large monitors. But, when reproduced over the smaller ones, it might become apparent that certain inner details in the music have become lost, due to the lower resolution of many small loudspeakers.

The engineer and producer then have the option of going back and making a mix that is really tailored to the smaller loudspeakers, and which will, all other factors being equal, probably sound better on lower-resolution players used by typical consumers.

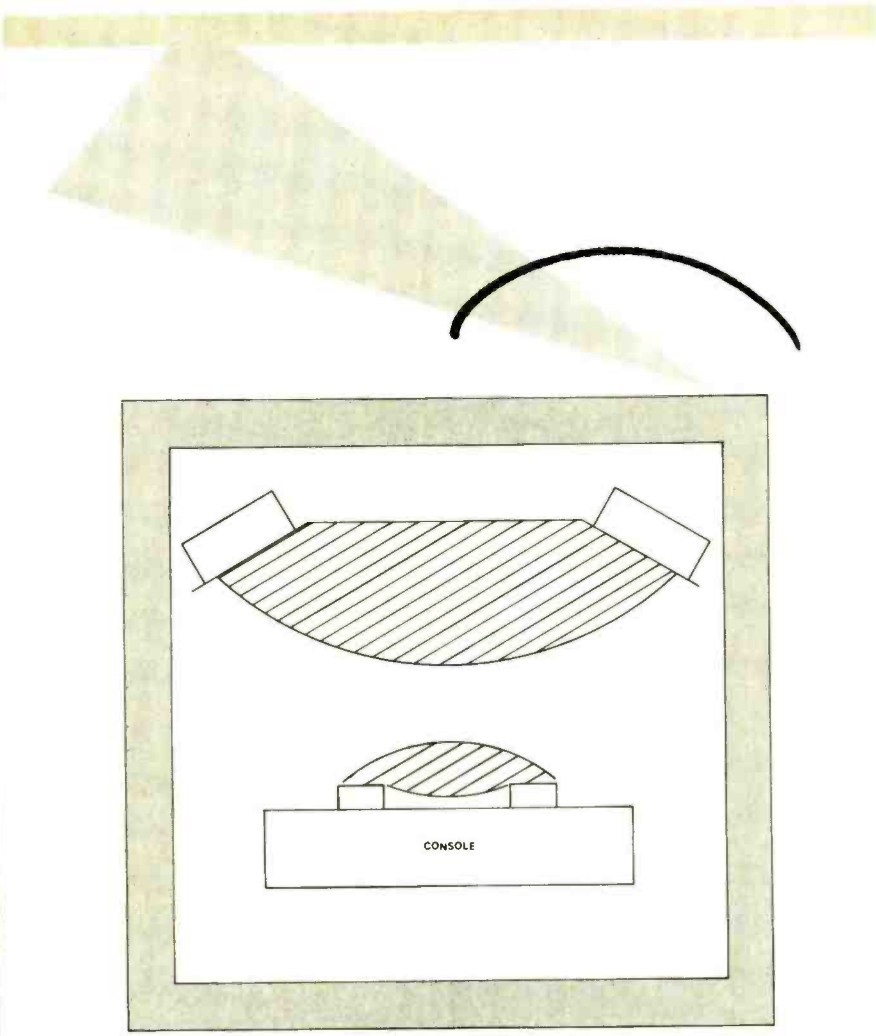
Performance parameters for close-field monitors

Close-field monitoring has been around for several years, but has become a vital link in the production chain only in the last half decade or so. Early loudspeakers used for the purpose were apt to be rather choppy in response. In time, engineers and producers demanded smoother response, and a family of 2-way systems with 8-inch woofers were developed by many manufacturers for the purpose.

Among the characteristics of a good close-field monitor are the following:

- *Frequency response:* Uniform from about 70Hz to 20kHz. The response through the midrange should be especially smooth.
- *Array size:* A 2-way vertical array is preferred because the longest array dimension can be held to about one foot if an 8-inch woofer is used. Under these conditions, the engineer will be in the transition region between the near and far fields, and will not readily perceive the sound as coming from both high- and low-frequency sources.
- *Sensitivity and power handling:* Most of the units in favor today have basic sensitivities in the range from about 87dB to 92dB, 1W at 1m. The actual sensitivity is not too important, as long as the model has enough powerhandling capability—and available amplifier power—to reach the required levels cleanly. Tastes vary all over the place, but a pair of close-field monitors should be capable of reaching levels of 95dB at the engineer's position with no distress.
- *Dispersion:* While controlled horizontal dispersion is an attribute in any loudspeaker, it is relatively unimportant in this application, provided that the on-axis response is smooth. Because the monitoring setup is usually optimized for one listening position, the principal axis is aimed at that desired location.

The vertical arraying of high- and low-frequency elements is the preferred orientation, in that it produced the most accurate and stable stereo imaging. However, some engineers prefer to place such loudspeakers on their sides, with



Stereo imaging with close-field monitors

While subjective observations will vary considerably, most engineers would probably agree that with large soffit-mounted monitor systems—especially if they are of the compression driver variety—phantom images tend to locate themselves slightly forward of the plane of the loudspeakers.

Why this is perceived is not well understood, but it may have to do with the generally up-front voicing of these systems, as well as with visual cues—the fact that there is a wall between the two loudspeakers which the listener doesn't normally want to "hear into."

By comparison, a pair of free-standing monitors placed on the meter bridge will often convey the illusion of loudspeakers, again possibly because of visual cues—the fact that there is no wall at all.

In both cases, fore and aft localization will be influenced by the ratio of direct-to-reverberant sound cues in the program mix itself.

In any event, engineers and producers will appreciate the added stereo perspective afforded by the smaller monitors.

the tweeters in-board. This orientation has the disadvantage of producing response lobing in the horizontal plane, making the ideal listening spot very critical.

- *Time-domain response* is just as important here as in any other application. Usually, there is no problem with small bookshelf systems, since they normally satisfy the Blauert and Laws criteria for acceptable response group delay.

Control room installations

Close-field monitors should never appear to be an accommodation or an afterthought. In fact, they have become an essential part of recording processes, and should be implemented in a professional manner.

A set of sliding platforms should be made for the console meter bridge, so that the loudspeakers can be easily located for either engineer or producer.

Reasonable gauge wire should be selected to hook them up, using professional connectors. A separate amplifier to drive the close-field monitors should be chosen to deliver the peak power for which the systems are rated. Be prepared to replace burnt-out monitors quickly—have a backup pair on hand.

Different models should be made available for quick changes. Always say yes when a producer suggests something you don't already have, and be ready to accommodate whatever a producer or outside engineer might bring.

Electrical switching between the close-field and main monitors should be positive and easily done. (Some studios have gone so far as to make sensitivity matches between the close-field and main monitors, for the convenience of producers and engineers who don't want to be blown out of the room when the switch is made to the big monitors.)

An important exception to the rules

On-location classical recording usually means the use of quickly installed monitoring set-ups in less than ideal spaces. The monitors usually chosen for this job are 3-way designs with a 10- or 12-inch woofer, located at a distance of about six or seven feet from the engineer and producer. It is essential that both engineer and producer perceive good imaging, which means that horizontal off-axis response must be quite uniform. This requirement implies a vertical transducer array.

The loudspeakers should be no farther away from the engineer and producer than necessary to satisfy their mutual demands for good imaging. Otherwise, the loudspeakers should be as close as possible to maximize their direct fields and thus minimizes room reflections.

Bandwidth should extend down to at least 35Hz for the recording of orchestral or organ music; this usually means that the systems will have sensitivities in the range of 87dB to 90dB, 1W at 1m. Generous amplifier power should be provided.

Because close-field monitoring is an important step in the production chain, it deserves more attention in implementation than most studio management personnel have traditionally given it. Too often, it is accorded the same casual treatment that headphone monitoring receives—and we all know what kind of trouble that can be! If you haven't already implemented the advice given here, it may be time to clean up your act. R·E/P

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"Live sound has to be intimate and real yet deal with the realities of gain before feedback, temporary set-ups, less than optimum placement, etc.

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Circle (28) on Rapid Facts Card

Beyer MC740 Condenser Microphone

By Lowell Cross

The Beyer MC740 studio condenser microphone is available in two models: the model MC740N(C), a standard 48V phantom-powered version with switchable pattern selection; and the MC740N(C/5), which offers remotely controlled pattern selection when used with the MSC740 power supply. The remote-controllable version (requiring 5-conductor cables and connectors) is intended for fixed installations in studios and concert halls, and permits the pattern selection to be made more conveniently in the control room.

The microphone is supplied with the EA-740 elastic suspension in a foam-lined plastic carrying case. Finish on the microphone body and suspension is anodized matte black. Both are attractively styled and elegantly assembled. The elastics of the suspension mount, however, are quite delicate for a studio model of this size and weight, which may mean that they could need frequent replacement if the assembly is subjected to anything other than the most careful handling. (Fortunately, a spare set of mounts is supplied.)

An examination of the circuit diagram, which is included with each microphone, revealed that a tantalum capacitor is used in the signal path (somewhat discouraged in today's audiophile circles). Other circuit features include the use of zero-volt dc potentials on the exterior capsule surfaces to minimize problems caused by humidity and the attraction of dust particles, and a dc-to-dc converter to provide internal capsule polarizing voltages that are higher than 48V sources (DIN standard 45 596) supply.

Lowell Cross is professor of music and director of recording studios, University of Iowa, Iowa City, IA, and a regular RE/P contributor.

Like certain other expensive German studio models, the MC740 offers five polar patterns: omnidirectional, "wide" cardioid, standard cardioid, hypercardioid and bidirectional figure-eight. The MC740 also has a switchable 10dB attenuator pad and, according to Beyer, allows uses up to 144dB SPL while maintaining less than 0.5% total harmonic distortion. There are two low-frequency rolloff characteristics (−3dB at 80Hz or 160Hz), in addition to the "flat" response setting. Optional accessories include a choice of wind or pop screens (not included with the pair sent for evaluation) and the usual complement of cables and power supplies. For our evaluations, the mics were 48V powered from the school's Neve console, an arrangement that worked properly without complications.

We evaluated the stereo pair of MC740s in the recording of a contemporary jazz group in a close-mic studio session and in the recording of orchestral music in a reverberant concert hall.

Evaluation sessions

To maintain uniformity with our previous evaluations, the microphones were first compared to our reference microphones during the Dec. 10, 1986, recording of a concert by the University Symphony Orchestra and Choral Groups in the University of Iowa's 2,600-seat Hancher Auditorium.

The cluster of microphones was suspended 13 feet above the stage, 10 feet downstage from the conductor's podium, which was on the center line of the auditorium. Our own 5-pattern German reference microphones were set up as a near-coincident cardioid pair, 7 inches each at a 45° angle to the center line. The MC740s were angled in the same manner, immediately adjacent to the respective left and right microphones of the reference pair, and required a slightly wider spacing of 14 inches. All of the four microphones were set for the standard cardioid pattern.

In the center of this arrangement was our other reference unit, the well-known ambisonic microphone. Both its B-format and stereo outputs were used; the latter was adjusted to yield the equivalent of two coincident cardioid patterns but with a somewhat wider angle than 90°.

The audible differences between the near-coincident cardioids and the coincident stereo output of the ambisonic microphone prompted this change. Our intention was to concentrate on *microphone* quality rather than *stereo* quality; we minimized the differences in stereo quality by widening the stereo spread of the ambisonic microphone.

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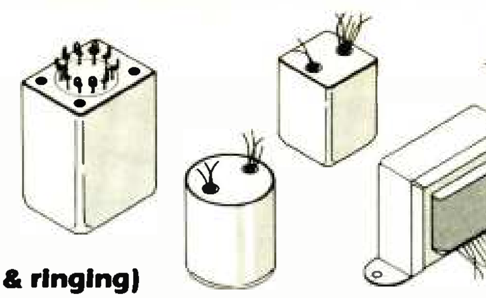
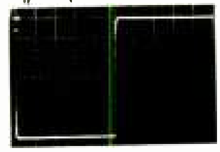


REVISED SEPTEMBER 1986



JE-16A
2 kHz Square Wave

INPUT TRANSFORMERS AND SPECIAL TYPES

Model	Application	Impedance Ratio Pri-Sec	Turns Ratio Pri:Sec	20Hz Max Input Level[1]	Typical THD Below Saturation (%) 20 Hz / 1 kHz	Frequency Response (dB ref. 1 kHz) 20 Hz / 20 kHz	Band-Width[2] −3 dB @(kHz)	20 kHz Phase Response (degrees)	Over-Shoot (%)	Noise Figure (dB)	Magnetic Shield[4] (dB)	Number of Faraday[4] Shields	Package[5]	PRICES 1-19	PRICES 100-249	PRICES 1000
MICROPHONE INPUT																
JE-16-A JE-16-B	Mic in for 990 opamp	150-600	1:2	+8	0.036/0.003	−0.08/−0.05	230	−8	<1	1.7	−30	1	A=1 B=2	75.42 82.89	49.87 54.81	34.40 37.81
JE-13K7-A JE-13K7-B	Mic in for 990 or I.C.	150-3750	1:5	+8	0.036/0.003	−0.09/−0.21	85	−19	<2	2.3	−30	1	A=1 B=2	75.42 82.89	49.87 54.81	34.40 37.81
* JE-115K-E	Mic in for I.C. opamp	150-15K	1:10	−6	0.170/0.010	−0.50/+0.10	100	−16	<7	1.5	−30	1	3	54.81	36.24	28.39
LINE INPUT																
JE-11P-9	Line in	15K-15K	1:1	+26	0.025/0.003	−0.03/−0.30	52	−28	<3		−30	1	1	122.22	80.82	55.75
JE-11P-1	Line in	15K-15K	1:1	+17	0.045/0.003	−0.03/−0.25	85	−23	<1		−30	1	3	52.32	34.59	27.10
JE-6110K-B JE-6110K-BB	Line in bridging	36K-2200 (10K-600)	4:1	+24	0.005/0.002	−0.02/−0.09	125	−12	<1		−30	1	B=1 BB=2	73.95 85.59	48.90 56.59	35.88 39.04
JE-10KB-C	Line in bridging	30K-1800 (10K-600)	4:1	+19	0.033/0.003	−0.11/−0.08	160	−9	<2		−30	1	3	53.17	35.16	24.53
JE-11SSP-8M	Line in/repeat coil	600/150-600/150	1:1 split	+22	0.035/0.003	−0.03/−0.00	120	−9	<3.5		−30	1	4	194.63	128.69	88.78
JE-11SSP-6M	Line in/repeat coil	600/150-600/150	1:1 split	+17	0.035/0.003	−0.25/−0.00	160	−5	<3		−30	1	5	98.39	65.06	44.88
SPECIAL TYPES																
JE-MB-C	2-way[3] mic split	150-150	1:1	+1	0.050/0.003	−0.16/−0.13	100	−12	<1		−30	2	3	44.85	29.65	23.24
JE-MB-D	3-way[3] mic split	150-150-150	1:1:1	+2	0.044/0.003	−0.14/−0.16	100	−12	<1		−30	3	3	76.19	50.37	39.42
JE-MB-E	4-way[3] mic split	150-150-150-150	1:1:1:1	+10	0.050/0.002	−0.10/−1.00	40	−18	<1		−30	4	1	114.40	75.64	52.18
JE-DB-E	Direct box for guitar	20K-150	12:1	+19	0.096/0.005	−0.20/−0.20	80	−18	<1		−30	2	6	54.56	36.07	28.23

1. (dBu) Max input level = 1% THD; dBu = dBv ref. 0.775 V
2. With recommended secondary termination
3. Specifications shown are for max. number of secondaries terminated in 1000 ohm (typical mic preamp)
4. Separate lead supplied for case and for each faraday shield
5. Except as noted, above transformers are cased in 80% nickel mu-metal cans with wire leads.

PACKAGE DIMENSIONS:

	W		L	H
1 =	1 5/16" Diam.			× 1 9/16"
2 =	1 3/16"	×	1 3/16"	× 1 5/8"
3 =	1 1/8" Diam.			× 1 1/16"
4 =	1 1/2"	×	1 3/4"	2 1/2" w/solder terminals
5 =	1 5/8" Diam.			× 1 3/4"
6 =	1 1/8" Diam.			× 15/16"

NICKEL CORE OUTPUT TRANSFORMERS[6]

Model	Construction	Nominal Impedance Ratio Pri-Sec	Turns Ratio Pri:Sec	20 Hz Max Output Level[7] (dBu)	across (n) windings	600 Ω Load Loss (dB)	DC Resistance per Winding	Typical THD Below Saturation (%) 20 Hz / 1 kHz	Frequency Response (dB ref. 1 kHz) 20 Hz / 20 kHz	Band-Width −3 dB @ (kHz)	20 kHz Phase Response (degrees)	Over-Shoot (%)	Package	PRICES 1-19	PRICES 100-249	PRICES 1000
JE-11-BMCF	Bifilar 80% nickel	600-600	1:1	+26	1	−1.1	40 Ω	0.002/0.002	−0.02/−0.00	>10MHz	−0.0	<1 [9]	7	81.55	53.92	37.76
JE-11-DMCF	Bifilar 80% nickel	600-600	1:1	+21	1	−1.0	38 Ω	0.004/0.002	−0.02/−0.00	>10MHz	−0.0	<1 [9]	8	56.32	37.24	25.69
JE-123-BLCF	Quadfilar	600-600 150-600	1:1 1:2	+32	2	−1.1	20 Ω	0.041/0.003	−0.02/−0.01	>450 170	−1.9 −4.0	<1 [8]	7	73.85	43.14	29.76
JE-11SS-DLCF	Bifilar split/split	600-600 150-600	1:1 1:2	+27	2	−1.0	19 Ω	0.065/0.003	−0.02/−0.01	>10MHz 245	−0.0 −2.5	<1 [8]	8	53.62	35.45	24.46
JE-11-ELCF	Bifilar	600-600	1:1	+23.5	1	−1.1	40 Ω	0.088/0.003	−0.03/−0.00	>10MHz	−0.0	<1 [9]	9	36.36	24.04	16.59
JE-11-FLCF	Bifilar	600-600	1:1	+20.4	1	−1.6	58 Ω	0.114/0.003	−0.03/−0.00	>10MHz	−0.0	<1 [9]	10	27.36	18.09	12.48
JE-112-LCF	Quadfilar	600-600 150-600	1:1 1:2	+20.4	2	−1.6	29 Ω	0.114/0.003	−0.03/−0.01	>450 205	−1.2 −3.2	<1 [8]	10	32.80	21.69	14.96
JE-123-ALCF	Quadfilar	66.7-600	1:3	+26.5	3	−1.3	8 Ω	0.125/0.003	−0.04/+0.06	190	−4.6	<6 [8]	8	50.96	33.69	23.24
JE-11S-LCF	Bifilar w/split pri.	600-600 150-600	1:1 1:2	+30	1 (sec)	−1.7	63 Ω	0.058/0.002	−0.02/+0.01 −0.02/−0.05	>10MHz 155	+1.1 −4.1	<1 [8]	8	50.96	33.69	23.24

6. Multifilar construction has no faraday shield: cannot be used as input transformer. All specifications are for 0 Ω source, 600 Ω load.
7. Max output level = 1% THD; dBu = dBv ref. 0.775 V
8. Source amplifier −3 dB @ 100 kHz
9. Source amplifier −3 dB @ 200 kHz
10. Output transformers are horizontal channel frame type with wire leads, vertical channel frames available. PC types available.

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PACKAGE DIMENSIONS:

	W		L		H	Mounting Centers
7 =	1 1/2"	×	2 5/16"	×	1 15/16"	2 13/16"
8 =	1 5/16"	×	1 15/16"	×	1 5/8"	2 3/8"
9 =	1 1/8"	×	1 11/16"	×	1 3/8"	2"
10 =	1 1/16"	×	1 7/16"	×	1 3/16"	1 3/4"

These charts include the most popular types which are usually available from stock. Many other types are available from stock or custom designs for OEM orders of 100 pieces or more can be made to order. Certified computer testing is available for OEM orders. Call or write for applications assistance and/or detailed data sheets on individual models.

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Circle (23) on Rapid Facts Card

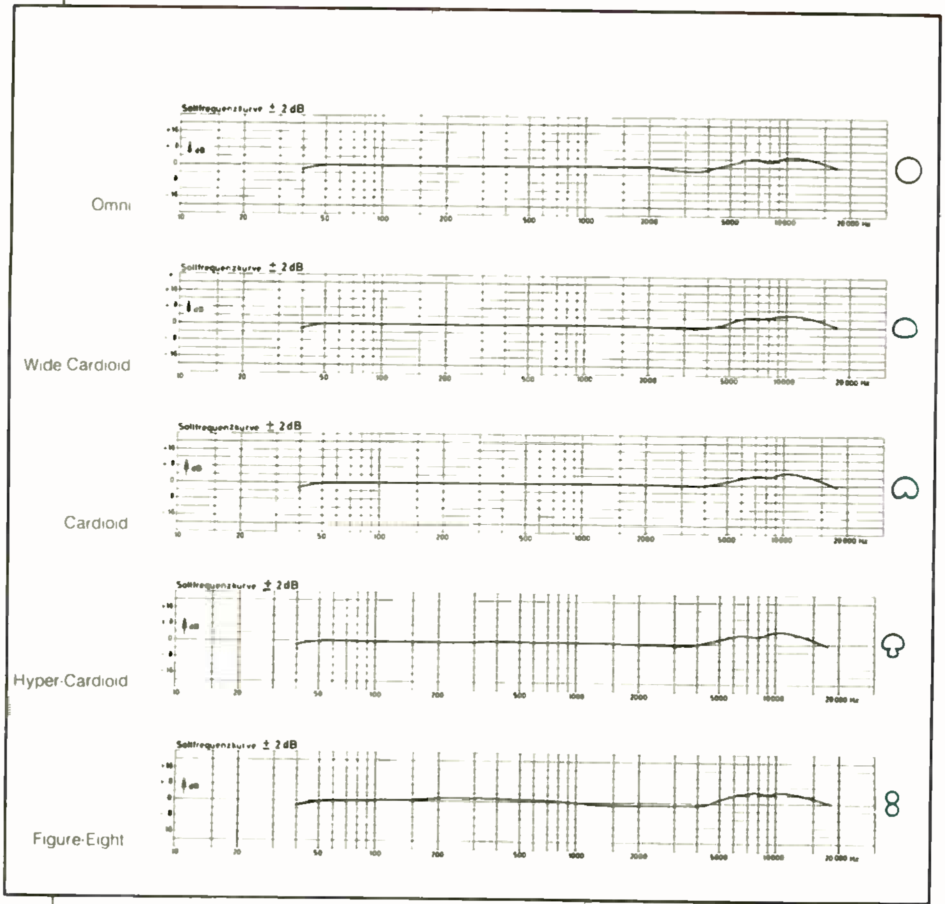

Figure 1. *Frequency response curves of the MC740's five switchable polar patterns. It should be noted that this data has been taken from the manufacturer's product literature, and has not been verified by the article's author or **RE/P** staff.*

Input-gain settings on the Neve model 5315/24 console were 45dB for both the reference pair and the MC740s. The settings were 0dB for the ambisonic system, which has line-level outputs plus separate gain and fader settings on its control unit. However, there was a noticeable level difference between the two Beyers; serial number 015097, our left-channel unit, has 2dB to 3dB less output than serial number 015029.

We listened to the ambient background noise of the quiet, empty concert hall after completing the installation. The hall's air-handling system, which has been measured in the NC15 to 20 range, completely masked the self-noise from any of the five microphones.

The concert was recorded at 15ips onto 2-inch 3M 226 mastering tape, using several channels of a Studer A80/VU-24C MkIII equipped with ANT telcom noise reduction. Tracks 3 through 6 comprised the B-Format outputs of the Ambisonic microphone; 13 and 14, its post-matrix stereo outputs; 15 and 16, the two MC740s; and 17 and 18, our German reference microphones. Console outputs were configured so that monitoring over the four K+H model 092 loudspeakers could be accomplished "live" in all cases. We could conveniently switch among the three stereo outputs, as well as a decoded surround-sound (4-channel) output, without disturbing the recording process.

These monitoring techniques enabled us to make direct comparisons without resorting to playback via the tape medium. However, the "listening panel" of studio engineers—Peter Nothnagle, David Muller and I—spent some time a few days later comparing the three microphones from the multitrack recording.

In this part of the evaluation, the MC740s sounded very similar to the more expensive reference pair. The principal differences were in the upper midrange and high-frequency regions, as is usually the case with various brands of high-quality condenser microphones. By comparison, the MC740s had slightly more HF output above 5kHz. It would be practically impossible to describe the differences in the mid- and low-frequency parts of the spectrum, which were excellently reproduced by all of the microphones.

Technical Specifications

Transducer type: *condenser.*
Operating principle: *pressure gradient.*
Frequency range: *40Hz-20kHz.*
Polar pattern: *switchable, omni, wide cardioid, cardioid, hyper-cardioid and figure-eight.*
Open circuit voltage at 1kHz: *10 Mv/Pa.*
Output level: *−40dB (0dB Δ 1 mW/Pa).*
EIA sensitivity rating: *−133dBm (0dBm Δ 1 mW/2 × 10^{-5} Pa).*
Nominal output impedance: *150Ω.*
Load impedance: *≥ 1000Ω*
Maximum SPL for THD<0.5%: *134dB (144dB w/attenuation).*
Weighted noise voltage: *3.2muV.*
S/N ratio: *70dB.*
A weighted equiv. SPL: *approx. 17dB.*
Bass attenuation: *−3dB at 80/160Hz (switchable).*
Phantom power: *48V (±4V).*
Current consumption: *1.4mV.*
Net weight (less cable): *13.8 oz.*

Circle 175 on Rapid Facts Card

The stereo output of the ambisonic microphone was perceptibly "brighter" by a small, but obvious amount, when compared to both of the two near-coincident cardioid pairs.

All three systems produced stereo recordings of satisfying musical quality. Members of our listening panel agreed that the MC740s compared favorably to both of our reference units. And, in my opinion, they can be recommended without reservation for cardioid-pattern stereo applications in concert recordings.

Jazz recordings

On Dec. 15, 1986, we recorded a jazz ensemble as a class project for University of Iowa recording students. The MC740s were compared to other well-known microphones in a close-mic studio setting. The group, organized by Ed Sarath, consisted of piano, bass (electric as well as amplified upright acoustic), a conventional drum set, Sarath on fluegelhorn and vocalist Sue Werner.

The first number recorded, Sarath's own composition "Desert Song," was entirely instrumental. We compared the MC740s (cardioid pattern), as used over the drum set, to a pair of miniature fixed-cardioid German condenser units. These two near-coincident arrangements were positioned as identically as possible for this part of the evaluation and were assigned to four channels of our A80 24-track. They were operated at 15ips with 3M 226 tape and telcom noise reduction. Other tape tracks were occupied with various mics used on the kick drum, bass amp, piano, Sarath's solos and as an overall MS stereo pickup.

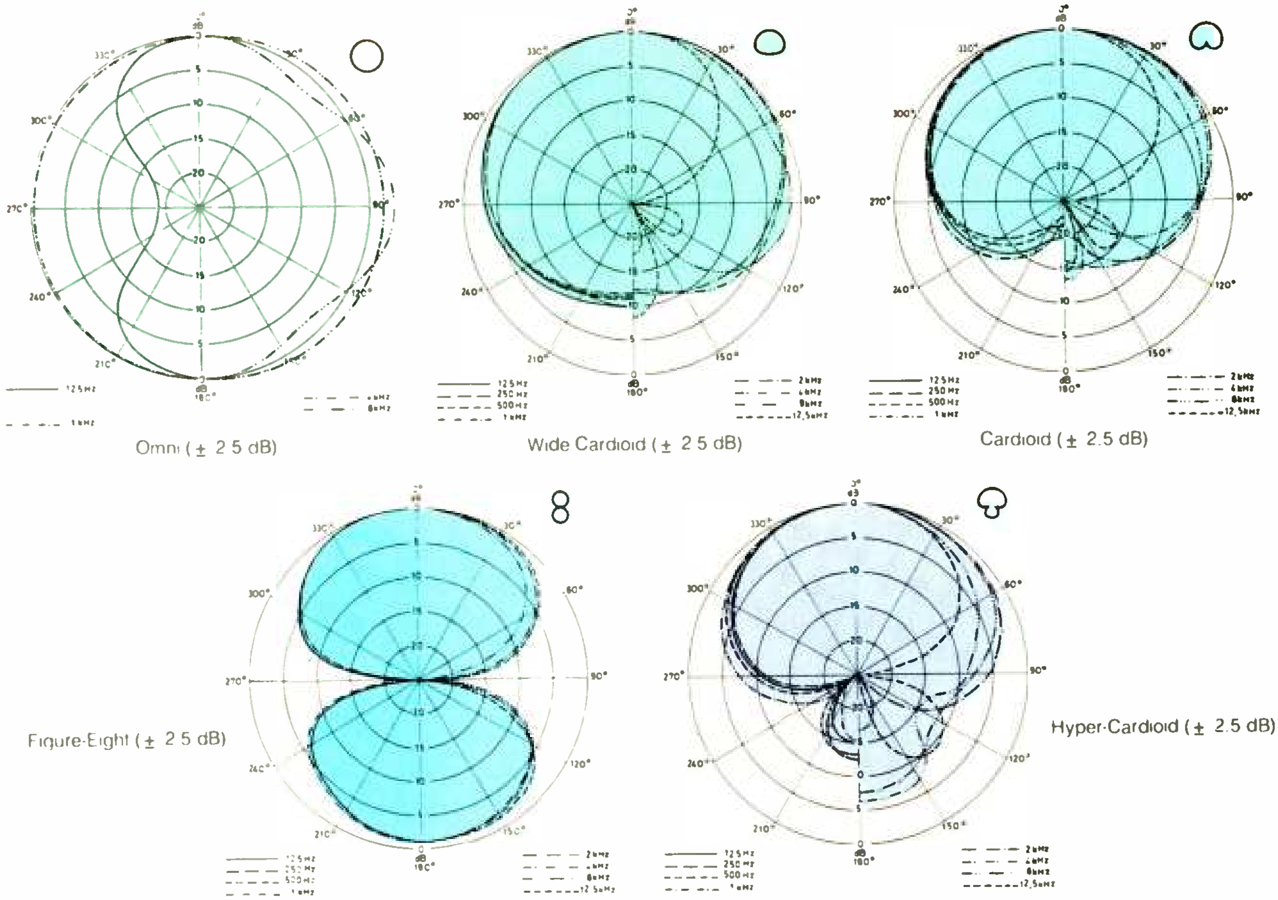

Figure 2. *Polar frequency response curves for the MC740's five switchable polar patterns, extracted from manufacturer's product literature.*

Then we recorded Sarath's arrangement of the classic swing tune, "Green Dolphin Street," in which his and vocalist Sue Werner's solos were featured. To learn how the MC740s performed with a brass instrument and vocals, a setup change was made: one of the MC740s was suspended immediately above a large-diaphragm, 3-pattern German studio model, a type frequently used for recording solos in multitrack sessions.

The latter microphone and the MC740 each fed separate tracks so that A-B comparisons could be made easily. The vocalist was within one foot of the two units; the bell of the fluegelhorn was slightly angled away from the capsules and maintained at a distance of about three feet. No low-frequency rolloff or attenuation switches were activated on either microphone; which were both set for cardioid patterns.

As in the concert recording, monitoring of the pertinent channels in the jazz session was accomplished both live and during subsequent playback of the multitrack tape. We discovered that the MC740s have a distinctive "high-end rise" when used in a close studio situation. Their brightness, in comparison to the other microphones, confirms information in the mic's brochure, including the cardioid frequency response curve. In all other respects, the drum set, brass and vocals were easily recorded by the MC740s, which provided clear and sharply detailed reproduction.

An inquiry about the published frequency response curves, shown in Figure 1, brought the following response from the factory in Germany: "...this measurement has been made under particular circumstances, namely 1m distant from the sound source in a plane wave [anechoic chamber]. The frequency response in the diffuse sound field which is not published, is decisive, as it also takes into consideration the other wave angles."

This information combined with our perception of relative high-frequency uniformity in the concert recording—*less* brightness or "high end rise"—could suggest that the diffuse-field (distant or reverberant) and free-field (close or anechoic) frequency response curves for the MC740's cardioid pattern may become different at frequencies above 5kHz. In this sense, the cardioid setting of the microphone, shown in Figure 2, has similar high-frequency characteristics to a pressure transducer (omni).

These properties will not be detrimental to the effectiveness of the MC740 in many cardioid applications; they simply indicate that the microphone will sound *brighter* when it is used *closer*. On the other hand, it is important to keep in mind that the most accurate directional microphones have nearly identical HF response curves in both the diffuse and free fields.

The Beyer MC740 is an excellent multipattern studio microphone, and has a price range between its Austrian and German competition. As a concert hall microphone, it exhibits a smooth, widerange response, with virtually no selfnoise, distortion or unpleasant colorations. When used in close-mic studio recording, its cardioid pattern offers a characteristic brightness reminiscent of many of the large-diaphragm condenser models of the recent past. Engineers who are interested in investing in a microphone that combines these elements should give the MC740 serious consideration. R·E/P

Evaluating Low-Frequency Driver Performance

By Clifford A. Henrickson

When selecting an LF transducer for live-sound applications, there may be more to look for in the performance specifications than a large coil diameter.

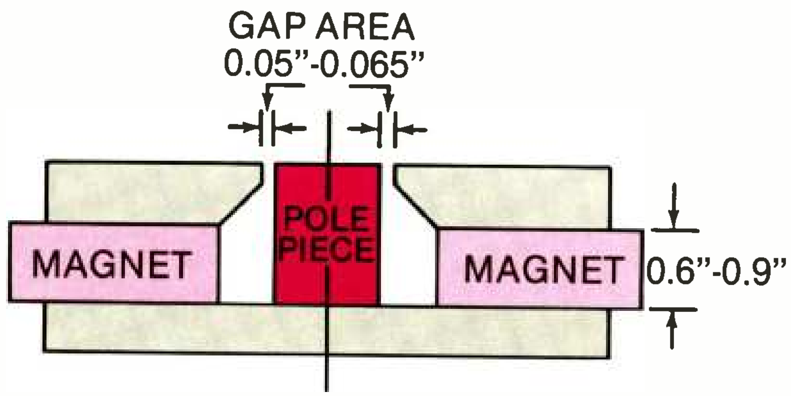

Figure 1. Cross section of a typical low-frequency driver. Between 0.6-inch and 0.9-inch of magnetic structure can support 0.05-inch to 0.65-inch of air in the gap space.

Why various manufacturers adopted specific coil sizes for low-frequency drivers is a mystery. Whatever the arbitrary reasons, manufacturers have retained certain coil sizes from the beginning and made them work within their respective manufacturing systems. Choosing a given company's product, especially woofers, usually means choosing a coil size and choice of a resulting engineering and design philosophy.

Magnet and air gap sizes

Loudspeakers intended for a given use in a market area tend to be cost-competitive. For this to be true, the (relatively expensive) magnet and magnetic circuit parts must be similar in cost. Magnet thickness is typically the same in most comparable products; ¾-inch to ⅞-inch, which is about the practical thickness limit imposed by the manufacturing process used to make ferrite magnets, now an industry standard. This being true, the magnetic *voltage* or magnetomotive force (measured in oersteds or ampturns) provided by these magnets is a relative constant. As a result, the magnetic gap spacing or length of gap that can be supported by the magnet force must be fixed, usually at about 0.050 to 0.065 of an inch (Figure 1).

Making the air gap substantially wider or narrower than this will change the magnetic bias or operating point of the magnet and will give an unacceptably wasteful use of the magnet. Table 1 lists the quoted efficiency, sensitivity and motor strength for nine representative low-frequency drivers.

The other aspect of the magnet's operating point is the flux density through it, which must be a certain value (about 2.2kG [kilogauss] for ferrite is optimum). For the same size magnet and intended efficiency, the larger coil diameter must have a shorter gap height to maintain the same flux density in the gap and in the magnet (Figure 2).

The consequence is that gap volume tends to be a constant for competing, similar-performing products. This is the same as saying that the product of coil volume or mass and flux density is con-

Clifford Henricksen is a consultant with U.S. Sound, Tuckerton, NJ.

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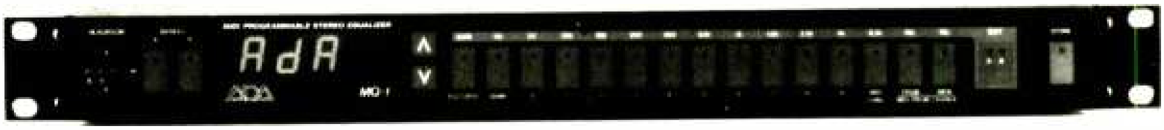

Circle (29) on Rapid Facts Card

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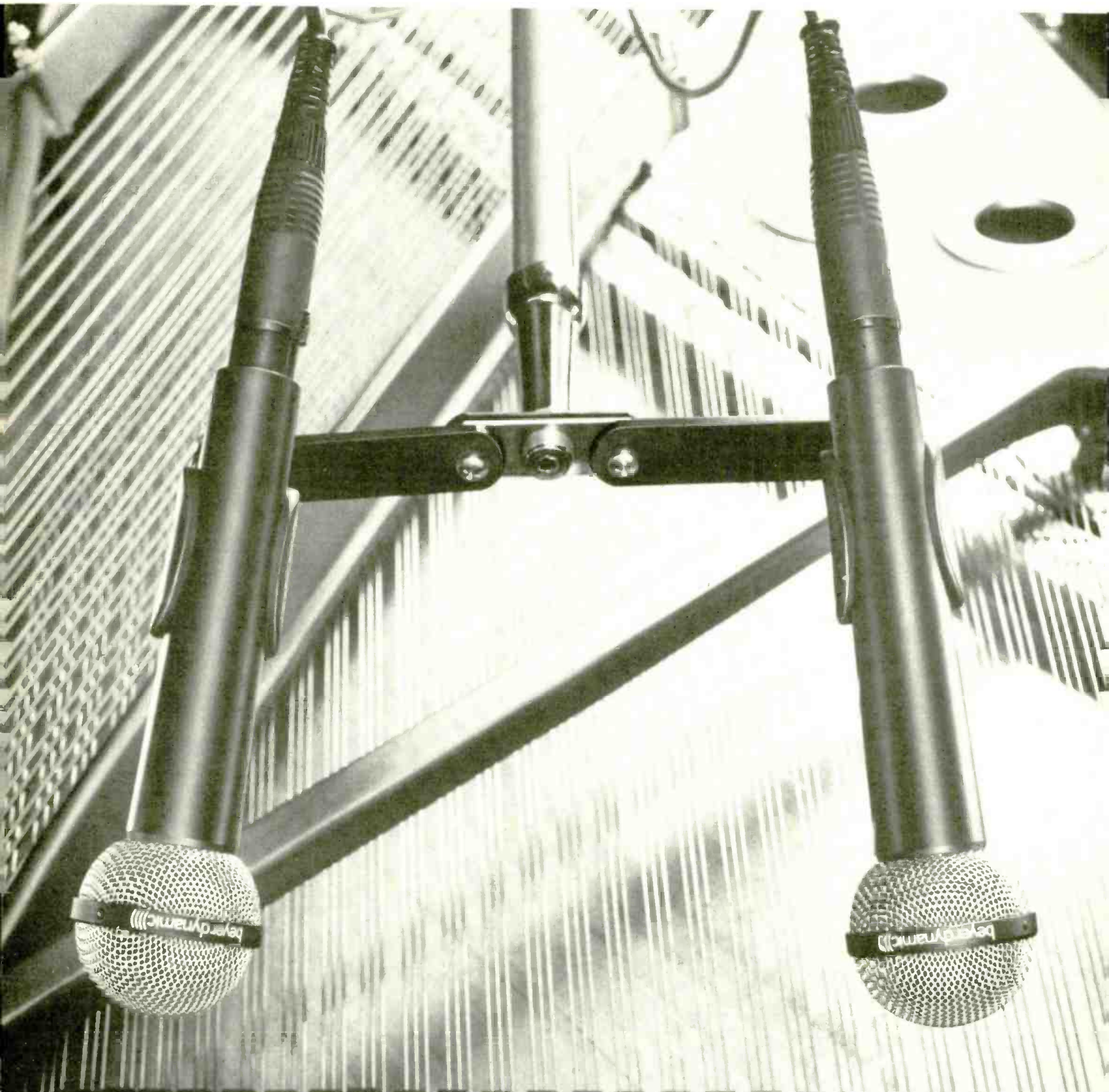


THE DYNAMIC DECISION

Table 1. Comparison of size, efficiency and motor strength (BL) for representative low-frequency drivers.

Model	Size (Inches)	Efficiency (%)	dB/SPL (1W, 1m)	Motor strength BL* in teslameter
A	15	8.7	101.0	22.3
B	15	6.2	100.1	18.0
C	15	3.5	97.6	23.0
D	15	5.0	99.2	22.5
E	15	1.3	93.3	20.5
F	15	2.7	96.5	24.5
G	18	2.1	95.4	21.0
H	18	2.9	96.8	24.5
I	18	5.0	99.2	25.0

*BL is the product of the flux density, B, times the length of the voice coil in the gap, L.

Table 2. Comparison of magnetic assemblies for units listed in Table 1.

Model	Top plate thickness (Inches)	Voice coil length (Inches)	Area of coil in gap (in^2)	*Peak excursion coil still in gap	Maximum excursion rated at 10% THD (inches)
A	0.35	0.28	3.51	0.035	0.12
B	0.43	0.42	3.38	0.005	0.13
C	0.28	0.630	3.51	0.175	0.20
D	0.43	0.60	3.38	0.085	0.16
E	0.28	0.75	3.51	0.235	0.33
F	0.43	0.80	3.38	0.185	0.22
G	0.35	0.96	4.39	0.305	0.375
H	0.43	0.80	3.38	0.185	0.22
I	0.35	0.75	4.39	0.20	0.22

Note that this table considers only the mechanical relationships and does not account for fringing flux.

*The peak excursion is defined as the absolute value of the difference between the voice coil length and the top plate thickness divided by two.

stant between the larger and smaller coil, because the volume of air in the air gap must remain approximately constant (Figure 3).

There are possible problems when using a larger coil, based on flux leakage. The larger coil expands more radially when it heats up and is harder to keep round. Therefore, clearances, especially between the top plate and coil outer diameter, must be a bit more generous for the larger coil.

This results in slightly more magnetic *waste* needed for large-coil gap clearances. A larger gap diameter will lead to more flux leakage or waste, in total magnetic lines. The shorter gap height leads to a magnetically less advantageous gap aspect ratio, which promotes more leakage. Also, for the same gap flux density, the larger perimeter yields a proportionally larger total leakage. This is of greater importance to high-efficiency, high-flux density designs.

Because the air gap diameter is larger, the hole for the pole piece through the center of the magnet is larger, thereby requiring the outside diameter of the magnet to be larger to maintain the same cross-section area of magnet (Figure 4). This in turn leads to even more flux leakage; in this case, around the magnet's outer perimeter. Again, the leakage is proportional to the magnet diameter. It leads to an even larger magnet diameter being necessary to achieve the same motor strength. The conclusion is that a bigger coil requires a heavier magnet for the same magnetic performance.

Extended-coil designs

The previous discussion is true if you require the same gap energy and mass of conductor in that gap. As power is applied to the speaker at low frequencies, the coil/diaphragm assembly begins to move and the coil moves out of the air gap. When the coil leaves the gap, there is no more magnet to push against and we reach a point similar to hard clip in amplifiers, only in this case the power supply rails are magnetic.

If we want the speaker to operate at

high excursions, our best approach is to make the coil longer than the gap; the popular overhung coil. The longer we extend the coil, the greater the passive proportion of the coil that is out of the gap. Therefore, the greater the extra dc resistance there will be to degrade the motor strength, which is inversely proportional to dc resistance. This extra dc resistance is, of course, directly proportional to coil diameter.

For the same magnetic excursion linearity, the larger coil naturally needs more motor strength to compensate for the extra resistance of the passive portion of the core. In commercial loudspeakers, larger-coil designs tend to have larger magnets for the same overall operating characteristics for all the previous reasons. Therefore, for equivalent-performance, cost-competitive designs, larger-coil extended-excursion woofers need larger magnets. (See Table 2.)

Heat transfer

Heat transfer and the amount of heating of a voice coil is controlled by a thin film of air between the coil surfaces and the magnetic circuit: the pole piece and the top plate. The effectiveness of this air film can be determined by the

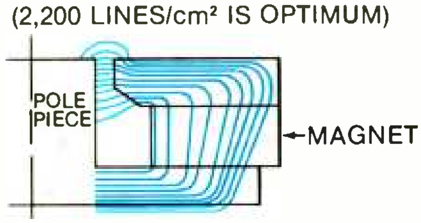


Figure 2. Partial cross section of a low-frequency driver showing the magnetic lines of force generated by a typical configuration.

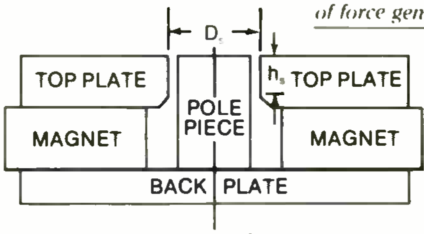


Figure 3. Cross section showing small diameter voice coil design.

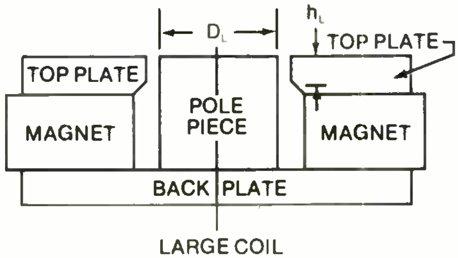


Figure 4. Cross section showing large diameter voice coil design.

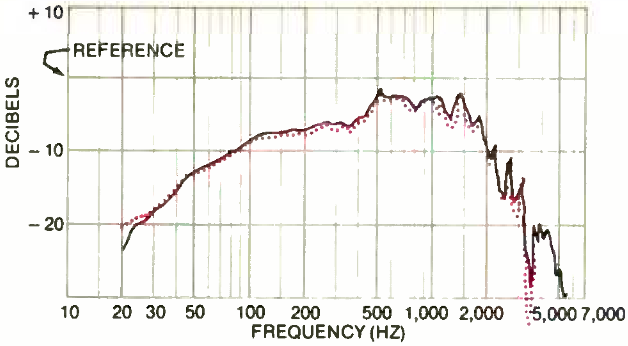


Figure 5. The 100W thermal compression curves for low-frequency driver E. Representing 1W at 20dB, 1.5% efficient 15-inch long-excursion woofer with 4-inch diameter coil.

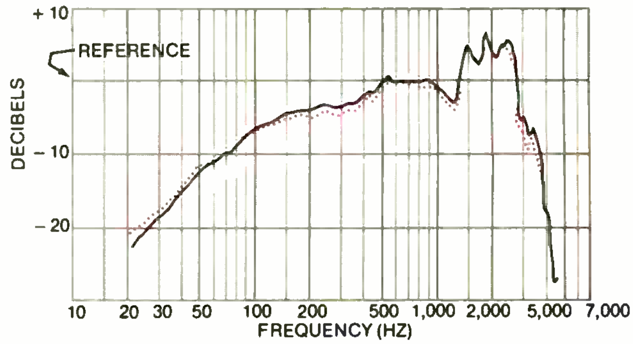


Figure 6. The 100W thermal compression curves for driver F. Represented by 1W at 20dB, 2.7% efficient, 15-inch, long-excursion woofer with 2½-inch diameter coil.

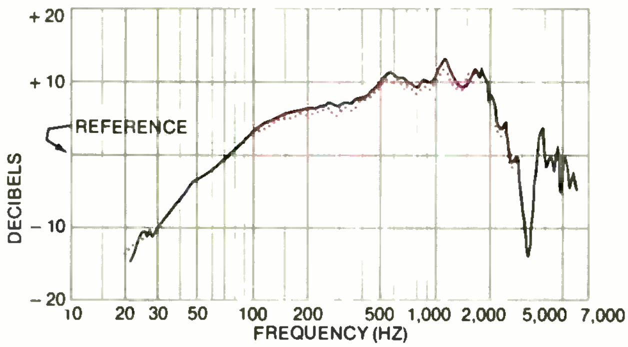


Figure 7. The 100W thermal compression curves for driver G. Represented by 1W at 20dB, 2.5% efficient, 15-inch long-excursion woofer with a 14-inch diameter coil.

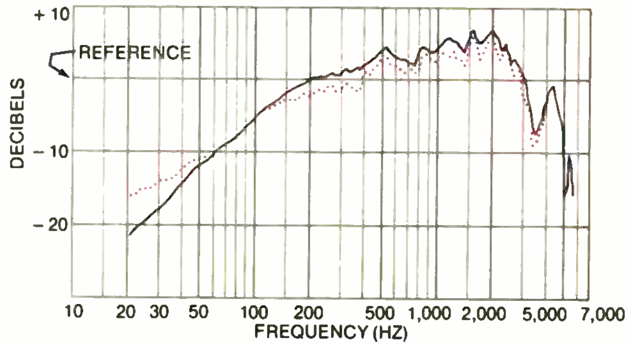


Figure 8. The 100W thermal compression curves for driver D. Represented by 1W at 20dB, 5% efficient, 15-inch, medium-excursion woofer with 2½-inch diameter coil.

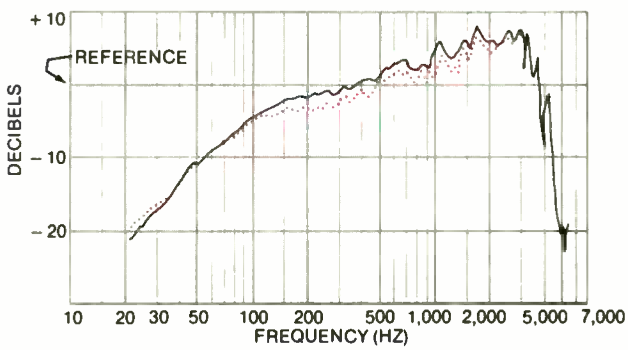


Figure 9. The 100W thermal compression curves for driver A. Represented by 1W at 20dB, 8% efficient, 15-inch, short-excursion woofer with 4-inch diameter coil.

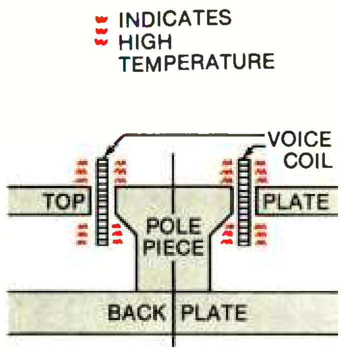


Figure 10. A cross section of the area around the voice coil showing the undercut pole piece configuration that would probably provide the least thermal protection to the coil.

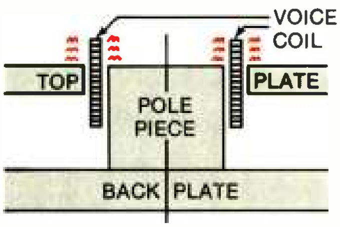


Figure 11. A cross section showing the area around the voice coil and the solid pole piece configuration. This design permits the top of the coil to heat up substantially.

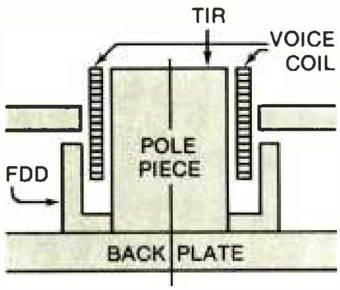


Figure 12. The voice coil area from a proprietary design shows the thermal inductive ring (TIR) and flux demodulating device (FDD).

same method we use to determine the dc resistance of wire. Thermal resistance is directly proportional to surface area, inversely proportional to the air spacing. Thus, large surface areas and tight air gaps contribute favorably to voice-coil cooling in the gap area.

Thermal resistance can be calculated by the following equations:

- $R_T = L/AK$;
  where
- L = thickness or length of air path;
- A = area of air path; and
- K = thermal conductivity of air.

In high-efficiency designs where coil length is equal to gap length (equal-height coil), the surface areas of the gap surfaces tend to be equivalent for equal competing products: larger diameter coils are shorter and vice versa. This surface area directly determines the heat transfer coefficient thermal resistance (°C/watt) of the coil. Thus, for commercially competing designs using high-efficiency, equal-height coils, heating is equivalent per watt of input.

For high-excursion, extended-coil designs, larger diameter coils will not have proportionately larger heat-radiating surface areas, because the air gaps are disproportionate in size. For a given coil length overhang, the heat-transferring gap areas are about the same, thus the larger-diameter coil will have a greater portion of its length outside the gap.

Because the air gap controls all heat transfer, larger-diameter coils may actually heat up more, but certainly not less. [For an opposing viewpoint, refer to the accompanying sidebar—*Editor.*] True, a larger diameter coil for the same excursion will have more total surface area, but not necessarily for heat transfer.

So far we have seen that for competing

designs, coil size is not necessarily a factor in determining heating of the coil. We will also see that the high-efficiency designs, which have light, equal-height coils with actively small surface areas, tend to have higher thermal compression than lower-efficiency, long-throw designs. Figures 5 to 9 demonstrate this phenomenon.

Figure 5 is a 1.5% efficient long-throw woofer for our 100W compression curves, 20dB subtracted from the 100W curve, which shows about 1dB of compression overall.

Figure 6 shows a 2.7% efficient loudspeaker. It compresses about the same as the unit shown in Figure 7, which is about 3% efficient. The woofer shown in Figure 8 has about 1.5dB to 2dB compression and is about 5% efficient.

Lastly, the unit shown in Figure 9 is a very efficient speaker and compresses the most on the slow-sine sweep test. The latter has a light coil and high gap flux; typical for high-efficiency designs.

The conclusion is that high-efficiency limited-excursion designs have light coils and low surface area and tend to compress the most. The reverse is true for inefficient, high-excursion designs. This is relatively independent of voice-coil diameter.

Total thermal engineering

We have seen that the air gap spacing between the coil's inner and outer surfaces and the area of magnetic paths, in the vicinity of the coil *only*, determine the cooling of the coil. There are few experienced sound reinforcement practitioners who will not tell you that the exposed or passive part of the coil is the weakest thermal link. Overpowered, long-coil woofers are always burned or charred in this exposed region.

The actual configuration of the coil and air gap area can have a great bearing on this. Figure 10 shows a configuration using a stepped pole, which provides the minimum heat transfer area to the coil. Both the upper and lower portions of the coil are exposed or have no direct heat transfer path. Figure 11 shows a solid pole, which protects the lower portion of the coil from heating, although the upper portion is still exposed.

For a recent design that uses special aluminum magnetic components to provide for heat transfer from all portions of the coil, a thermo-inductive ring allows heat generated in the upper portion of the coil to flow into the aluminum heat path (Figure 12).

The lower coil section heat transfer ring, while serving as an effective shorting ring for improvement in midband hysteretic distortion, also provides an additional heat path for the lower portion

Reducing heat buildup in LF transducers

By Mark Gander

Although the phenomenon of dynamic compression in low-frequency transducers has been observed for years, it has not been discussed to any extent in the literature and manufacturers do not directly refer to it when listing specifications. Many live-sound engineers may have heard of it, but few are likely to have any idea of its magnitude.

To most users and specifiers of low-frequency transducers, the manufacturer's published maximum thermal input rating, along with the published sensitivity rating, give a measure of how loud the device will play. As we shall see, such may not be the case.

Duty cycles and voice coil heating

Most sturdy transducers can withstand momentary power input surges up to 10 times their rated power handling, as long as frequencies are high enough that the voice coil does not try to move beyond safe mechanical limits. The key here is duty cycle.

For many kinds of program material, the duty cycle may be nearly a continuous one. Under such operating conditions, the voice coil heats up and its resistance begins to rise.

Figure 13 shows what happens when a 380mm (15-inch) transducer is swept over the audible range at power inputs of 1W and 100W, the traces superimposed and then displaced by 20dB. The degree of dynamic compression is about 1.5dB over the entire frequency range.

The increase in voice coil resistance is given by the following equation:

$$R_T = R_t [1 + \alpha (T_T - T_t)]$$

- T_T = *some elevated temperature in degrees Celsius;*
- T_t = *room temperature, normally 20°C;*
- α = *the temperature resistance coefficient; and*
- R_t = *the voice coil resistance at room temperature.*

For aluminum and copper, which are the materials normally used for voice coils, the value of α is approximately 0.004°C.

It is not unusual for voice coils to reach temperatures in the range of 200°C (400°F). A voice coil that has a dc resistance of 6Ω at room temperature will increase to 10.3Ω at such temperatures. High power handling voice coils may reach about 270°C (520°F). Under this condition, the resistance will have at least doubled.

Mark Gander is vice president of marketing at JBL Professional, Northridge, CA.

Shifts in low-frequency balance

The effects of such changes are fairly audible. Music is robbed of much of its "punch," and low-frequency alignments will be shifted, changing the balance of the system.

Figure 14 shows what happens to a natural flat low-frequency alignment when the transducer is operated at an elevated voice coil temperature of 150°C (302°F). System damping decreases *and a response peak at the transducer's resonance frequency appears. Midband sensitivity shows a decrease of about 3.5dB, half resulting from the efficiency decrease and half from the higher impedance, which draws less power from the amplifier.*

The culprit, of course, is excessive temperature rise and the cause is heat generation in the voice coil with insufficient cooling. Most low-frequency transducers used in sound reinforcement applications are far less than 10% efficient and the bulk of the input power is converted directly to heat.

Large diameter voice coils are far less susceptible to dynamic compression than are smaller ones, as shown in Figures 15 and 16.

Figure 15 shows the 1W, 100W, dynamic compression of a 380mm (15-inch) low-frequency transducer with a 100mm (4-inch) diameter voice coil. Note that the compression is about 1.25dB over the entire range.

Figure 16 shows the same measurement made on another speaker with a 63mm (2.5-inch) diameter voice coil. The dynamic compression is about 2.5dB over most of the frequency range.

For the same dc resistance and voice coil axial height, the wire chosen for a 100mm diameter coil will be 100/63, or 1.26 greater in cross-sectional area than the wire used in the 63mm coil. This will result in a proportional reduction in resistance per unit length. Less resistance translates into greater current capacity, and thus less heat. The increased surface area of the coil will also promote better heat transfer.

Ferrofluids, which have been used successfully in small high-frequency transducers for consumer loudspeaker systems, are of little use here. The high temperatures, which are very much a part of our business, simply result in deterioration of the ferrofluids.

A concern for system integrity and accuracy requires that the effects of dynamic compression be anticipated and compensated for at the design stage. Large voice coil transducers are presently the most economical solution to the problem.

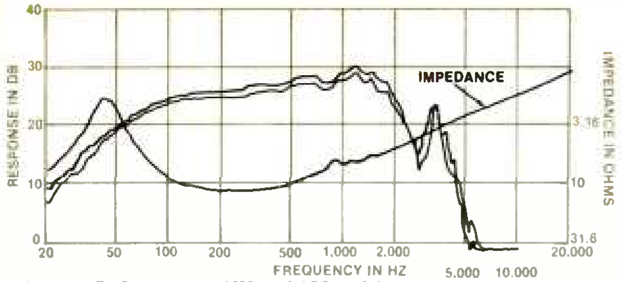


***Figure 13.** Sweeps for 1W and 100W drive levels superimposed and adjusted by 20dB. The bottom curve represents 70dB/SPL and 90dB/SPL for 100W.*

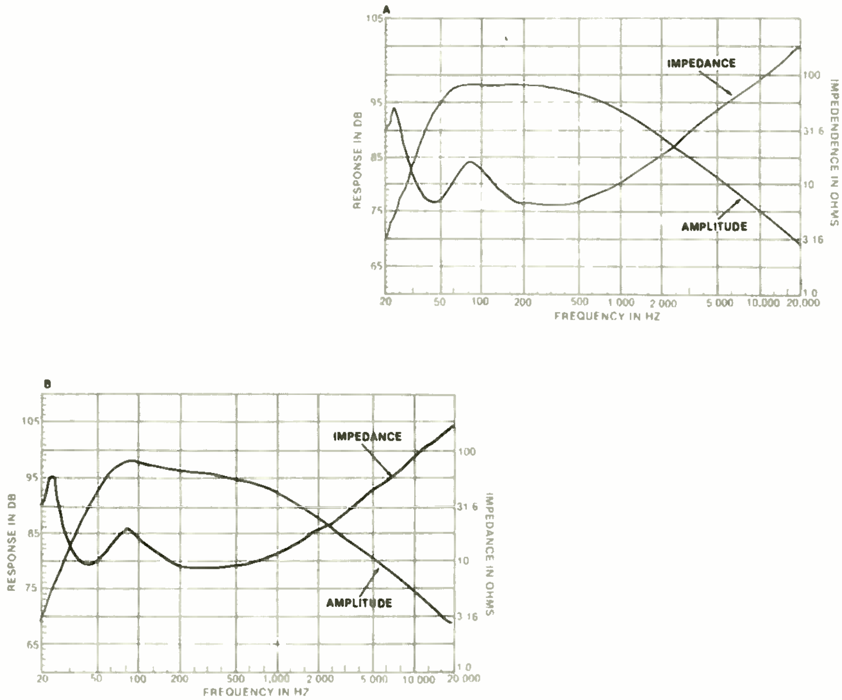


***Figure 14.** Low frequency alignments shifts caused by voice coil temperature rises. A is the alignment optimized for maximum flatness at 27°C (80°F). B is the alignment with a voice coil temperature or 150°C (302°F).*

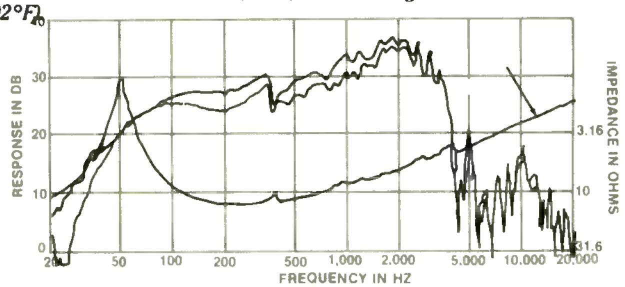


***Figure 15.** The dynamic compression data for a low-frequency transducer with a 100mm (94-inch) diameter voice coil. The bottom curve represents 70dB/SPL for 1W and 90dB/SPL for 100W.*

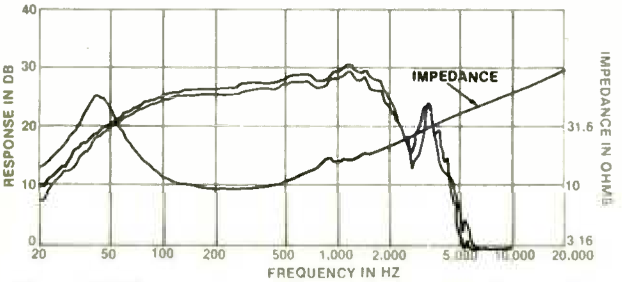


***Figure 16.** The same type of dynamic compression data as shown in Figure 15. However, this data was taken on a different 380mm (15-inch) transducer with a 63mm (2.5-inch voice coil. The bottom curve again represents 70dB/SPL for 1W and 90dB/SPL for 100W.*

of the coil. In addition, a patented Teflon-based coating is applied to the gap face portion of the top plate.

Users will often overpower speakers past their rating, intentionally or not. When this occurs, the coil in a standard design will eventually expand into contact with the steel top plate, causing almost certain disaster. As a last line of defense, this Teflon-based coating does an effective job of saving the speaker by both lubricating and insulating rubber contact caused by overpowering.

Surviving coils taken out of severly overpowered speakers show a burnishing of the coil's outer diameter and no burns or shorts. This kind of thermal engineering is crucial in the decade ahead where higher power and less equipment is a must for concert sound.

Which coil to choose?

Bigger coils have lower inductance for the same dc resistance. Smaller coils usually have more high-end response (500Hz to 3kHz) because of the mechanical action of a smaller drive circle on a given size cone. Speakers with different-sized coils sound different. Drive stresses are higher for a smaller coil, but practical design and construction can prevent this from being a problem.

Small coils (approaching one inch) become problematic; magnet design where the pole flux becomes too high; and a modern high-output design becomes a practical magnetic impossibility. Per a previous discussion, large coils become a leakage nightmare and the customer pays dearly for a cost-eating monster magnet. Large- and small-coil speakers sound different, but which is better?

Here's one suggestion on how to choose. First and foremost, don't use coil diameter as the only criterion. Look around and see what's going on in the field. Ask your friends. Buy some competing speakers or get a dealer to compare them for you. Hook them both up to the same amp and see which one blows up first and which one sounds the best. Use your own ears.

Many of us are afraid to state an opinion. What if you disagree with a "golden-ear" or even a manufacturer? It's your opinion and you shouldn't give it away. If you're not confident, practice and get good; do it a lot and your opinion will be much more solid. Try reading data sheets. Reputable manufacturers survive in the professional market partly on telling customers exactly how a speaker behaves. Do all this and re-evaluate it after you get the equipment and see how it really works. But don't use only coil size to select a speaker.

R·E/P

Part 1

Time Code and Synchronization

Keeping Music Videos and Films in Sync with Audio Playback

By Larry Blake

For the unwary, dealing with time code can result in untold synchronization headaches. This 3-part series of articles will concentrate on practical hands-on information to help alleviate most of the common sync problems.

Sensible techniques developed at the beginning of sound films in the late Twenties placed filmmakers around the world in good stead for the next 45 years. Cameras and sound were locked to the same reference via multiduty or selsyn motors. Recording was on sprocketed film, first using optical and later magnetic sound, while playback on the set often employed an interlocked turntable. The hassle of everyone being tied together with "umbilical cords" paid off in post-production, and such techniques were used in Hollywood into the Seventies. Indeed, the concert scenes for *Woodstock* were shot using ac line-frequency reference.

Around this time two developments were thrown into the works that have forever expanded the potential requirements of proper sync. One was replacing the cabling between tape recorders and film cameras with crystal control. Camera motors are driven at a constant speed, while portable ¼-inch recorders use a crystal oscillator only to record a reference "pilot" track; the tape machine motor is not normally servo controlled while recording. Variations in the speed of the audio transport are negated during transfer to mag film either by locking both playback and recording machines to line frequency, or by driving one from the other.

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

The real variable is the relative accuracy of the crystals in the camera and recorder. This can be stated as: How close to a *precise* 50Hz or 60Hz are the crystal oscillators capable of dividing to? Modern equipment is accurate enough to hold sync to within a frame over an 11-minute camera roll (1,000 feet in 35mm and 400 feet in 16mm). Nevertheless, there is no longer the cabled assurance that picture and sound are obeying the *same* time base.

The other Pandora's Box was the development of SMPTE-EBU time code to interlock film, audio and video machines. A good many of the problems resulting from the misuse of time code have nothing to do with the recording or use of the code itself, but instead relate to a film-to-tape or tape-to-film transfer at some point in post-production.

Such a transfer changes not only the time base but also the speed of the program. (More about speed changes later.) Shooting, editing and releasing in only film or only video is a relatively straightforward pursuit; mixing the two mediums at any point after shooting and before release requires careful attention to a small number of golden rules. By way of illustration, we will first look at the steps involved in a standard music video shot on 24fps film to playback, and edited on videotape.

(During this article, for purpose of simplicity the word "video" is to be read as NTSC color, with a "field rate" of 59.94Hz. Each frame is composed of two fields; the field rate is twice the frame rate. Problems concerning other video and time base standards will be featured later in this series.)

Before the shoot

If he is fortunate enough to be brought on prior to the shoot, the first concern on the part of the sound person in charge of playback is to ensure that the playback tapes contain a time base reference derived from a *sync* audio master. In almost all instances this master will be a copy from a 2-track mix that contains no sync pulse or time code. Absence of sync reference renders a tape *absolutely useless* for music video purposes; once the sync audio master is made, the original mix ceases to exist as far as the filmmakers are concerned. This fact should be stated upfront to the artist or whatever representative you deal with (producer, engineer, manager, etc.).

If they quibble, remind them that this problem could have been avoided by simply mixing onto ½-inch 4-track, with time code or a sinusoidal (i.e., sinewave) sync tone on track #4, or by mixing to a digital 2-track master, which includes a stable time base (otherwise you wouldn't be able to replay the tape). Even better would be the presence of a sync pulse on the *multitrack* master, thus allowing the song to be remixed even after the video is edited. Because the audio master's time base has been derived from the master mix (if it isn't the master mix itself), which in turn copied it from the multitrack tape(s), there will be an unbroken sync chain.

The other issue upon which the artist should be consulted concerns which version of the mix is to be used to make the sync audio master. Very rarely will you have access to the original 2-track mix; more often than not you will receive a 1:1 copy or even the equalized disc mastering copy. (Or a copy of *that* tape: can you say "digital?" This current discussion assumes the more common procedures using standard analog recorders. The unique sync requirements of digital machines will be discussed in a later installment.)

In any event, try to find out what version the artist and the production team consider definitive. It's possible that the contribution made by the mastering engineer was such that they would never want the original 2-track mix to be heard in any medium, regardless of the generation-loss issue.

Making the sync audio master

With the "best" non-sync mix in hand, it is time to make the sync audio master that will be used to stripe audio and time code to the ¾-inch off-line and 1-inch on-line video masters, as shown in Figure 1. In addition, it will be used to make the ¼-inch playback tapes for the shoot.

The format of choice for the audio master is generally considered to be 4-track ½-inch tape, with stereo audio on tracks #1 and #2, and 29.97fps time code on #4. Note that 29.97fps time code, and not 24fps code, is used on the audio master when shooting 24 or 30fps film for eventual video editing. (This concerns playback shoots only; use of 30 and 24fps time code in film production *recording* will be discussed in next month's issue.)

A 59.94Hz sinewave representing the field rate can be recorded on track #3 as a backup in case of damage to the time code. Even in the unlikely event of a total loss of code, the sinewave could be used as a time base reference for a time code generator to stripe the tape with new code. Use of track #3 in this manner is optional, and is by no means necessary. Note, however, that the field rate sinewave must be recorded at the same time as the time code, and derived from the *same* sync generator.

The code at the beginning of the song should be a sensible time such as 01:00:00:00; additional songs on the same shoot can start on the 2-hour mark, etc. If time code is being recorded on track #4 while the song is being transferred to the sync audio master, a synchronizer can be used to make sure that the song starts exactly on the hour. The

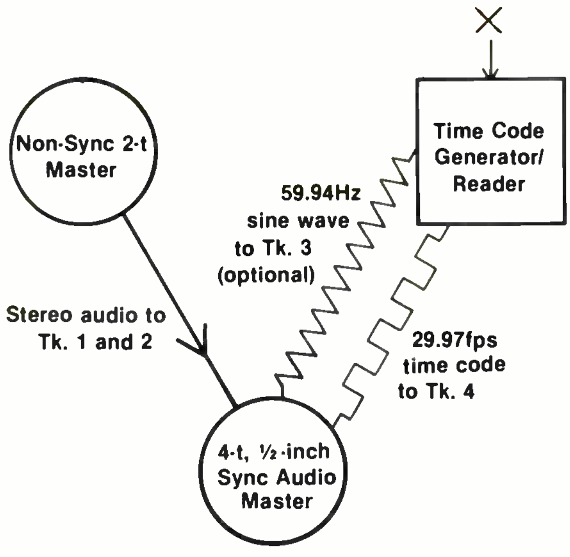
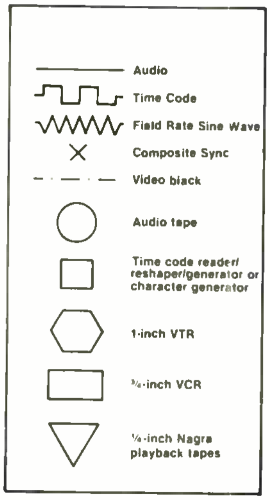


Figure 1. Creating the Synchronous Audio Master.



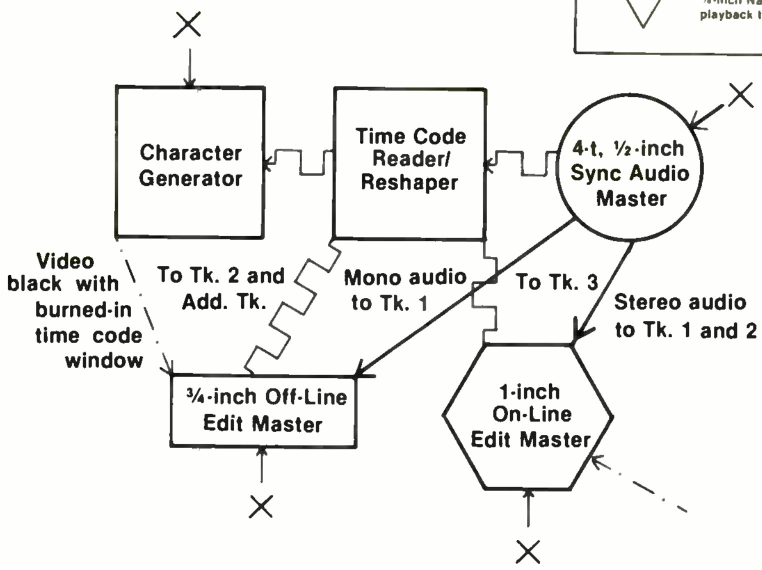


Figure 2. Creating the off- and on-line Video Edit Masters.

first modulation is found and the master non-sync tape is wound back one second. During the transfer, both the time code generator and the 4-track tape start at approximately 00:59:00:00, giving a generous 1-minute pre-roll. Thirty seconds is recommended, and 15 seconds should be considered minimum.) At 00:59:59:00 a relay contact closure is triggered to start the playback deck, and if the song doesn't start exactly on the hour, the start time can be fudged on the synchronizer.

If the transfer is being made at a video post facility, you can be almost certain that the time code generator and all video machines are referenced to a common NTSC composite sync generator, and thus the time code will be in the desired 59.94Hz time base. (Remember, you will be doing all post work on video.)

However, insist that the transfer person understands that you require NTSC sync, and that he can *assure* you this is what you're getting. Think twice if a glazed look comes over the transfer person's eyes when this issue is broached, and be especially careful if you are at a

Tips for synchronous playback

• Obtain a copy of the song's lyrics and retype them, numbering the verses and choruses. A few seconds before each verse begins, place a numbered EditTab splice on the tape so you can find your place in the song easily and repeatably. The pre-cut EditTab splices are easy to use in the field, and are whiter than normal splicing tape, which improves visibility.

• Have three copies made during the transfer from the 4-track audio master. Keep two on the roll for the playback Nagra, and the third somewhere else as an emergency backup.

• During the transfer from the audio master, use the Nagra's reference oscillator to record three beeps leading into the song. (This is another reason to start the song exactly on the hour, since you automatically know where the song begins.) If you forget during the transfer, and are using a mono Nagra, it's easy to put the beeps on later. Proceed as follows: find the first modulation of the song and then reverse the reels on the deck, with the body of the song now safely on the takeup reel. Record the three beeps and then flip the tape over and edit the last beep so that the three lead into the song in time.

However, if you are using a IV-S TC with center-track time code, you must record the beeps when the song is transferred from the audio master, because the presence of continuous time code precludes the ability to remove an inch here or there. The only other option would be to record the beeps on a studio machine where the time code track could be put into the "safe" mode. Recording these beeps should always *be done because they are needed by the group, not to mention giving the crew notice that "here it comes."*

• Put a piece of gaffer's tape on deck of the Nagra extending down over the right side of the main function knob. This will prevent you from accidently going into record mode.

• When playing back during the shoot, have the "line and phones" tape/direct switch on "direct," giving you control of the output from the Nagra's fader(s). This way you can ensure that no one will hear the song until you can "see" that you have achieved lock. Lock is visually verified in the following manner:

On a mono Nagra 4.2, put the meter switch on "synch" and wait until the needle is stationary, indicating that the motor is stable and phase-lock has been obtained. If the transport is slewing back and forth and cannot lock, apply a little pressure on the feed reel with your left thumb. Watch the meter and "guide" the machine into stability. Then bring up the fader(s).

The procedure is identical with stereo machine (either the normal IV-SL with a FM sync track, or the IV-S TC with center-track time code), except that you use the meter on the external QSLS synchronizer. The other difference between mono and stereo machines is that the function switch on a 4.2 must be in the "Playback with Loudspeaker" position (straight down) to make use of the QSLI internal synchronizer. Either playback position will allow resolving on a stereo machine.

• When shooting on videotape (for example, Betacam, M-II, ¾-inch U-matic, etc.), verify that audio and time code are getting down, and do this prior to the first take. Record a short segment and play it back, making sure to check all *cameras: listen to the audio and read the time code. If no code reader is available, the Nagra can be set to read external time code.*

If you are playing back with a mono Nagra while shooting on videotape, you must remove the Nagra SK jumper plug and feed in a 59.94Hz reference. This can come either from an external crystal or it can be stripped from the composite sync that is fed to the video camera.

It is best to record the playback time code on audio track #2, reserving the time code track for the VTR-generated time-of-day code, which will provide a unique address for each take. The playback code can be burned into a second window while making the off-line editing copies. Be sure to disable any noise reduction or automatic gain controls that might be applied to the audio tracks during the shoot.

• Finally, when using an IV-S TC, double check that you are set to the correct frame rate (29.97 or 30) and whether drop or non-drop time code has been selected. Although the switch is on the inside of the Nagra, the time code can be checked using the "status" key.

recording studio that doesn't handle much sync material. Beware!

Prior to this transfer procedure, you should inquire of the video's director and editor whether they prefer drop-frame or non-drop-frame time code. Probably the single most common misunderstanding regarding time code is the assumption that 59.94Hz/29.97fps equals drop frame and 60Hz/30fps equals non-drop. While it is true that drop-frame time code was developed to match 59.94 NTSC color video to the 60Hz "clock on the wall," it is *incorrect* to regard the two as being synonymous. You can have drop-frame with a 60Hz time base or non-drop code in color video, and the decision on which format (drop or non-drop) to use is a matter of individual preference.

The point to be made here is that the *time base* reference—60 or 59.94—is *the* important factor regarding correct sync. If you understand the fundamental issues involved in selecting the correct field rate/time base, then matching the frame rates of time code with that of film cameras becomes second nature.

The next step will be to make the "blacked and coded" ¾-inch off-line and 1-inch on-line edit masters, as shown in Figure 2. As must be done prior to a video edit session, we will record a continuous control track plus color black on the picture. From the audio master, we will be striping time code and recording the song on the audio tracks for editing reference purposes. Both video recorders *must* be locked to the same NTSC sync generator.

It is usually standard practice to record a mono combine on track #1 of the ¾-inch tape, with time code on both track #2 and the "address" time code track. This is a safe practice, since you are usually unsure of the requirements of the off-line edit system prior to the shoot. The code can also be burned into a "window" on the ¾-inch's image, serving as a reminder to the editor where he is in the song during the course of assembly editing. (As the song is edited, the blacked and burned-in picture is replaced, shot-by-shot, with the picture.) Stereo audio is recorded on tracks #1 and #2 of the 1-inch on-line master, with time code on track #3.

Why can't I play back from a cassette recorder?

Now it's finally time to make the ¼-inch copies to be used for playback on the set. Many a producer will ask: Why can't you just play back from a cassette recorder? The reasons should be self-evident at this point: cassette recorders do not record any kind of sync track that can be copied from the audio master. Even if you did record a sinewave sync tone on track #2 of a high-end portable cassette deck, the machine is unable to self-resolve.

"Resolving" can be loosely defined as the comparing of a sync tone off tape with a stable sync reference, either from an external source or a machine's internal crystal, and servo-controlling the transport to phase lock the two tones.

If you are asked why a cassette deck cannot be used, perhaps the best answer is another question: Why *not* use a Nagra? A standard 4.2 portable mono recorder rents for approximately $30 a day, a price that even the most shoestring video production should be able to afford.

Some thought must also be given to the recording of playback tapes because one simple mistake here can negate all of the prior and subsequent careful attention to sync as shown in Figure 3.

First, you must understand what will happen during the film-to-tape transfer. When a film is shot at the standard speeds of 24fps or 30fps, the motor on the camera is referenced to a 60Hz crystal. However, because NTSC color video has a "field rate" of 59.94Hz, film is actually slowed down during telecine, with 24fps film playing back at approximately 23.97fps. The difference between 60Hz and 59.94Hz amounts to a 0.1% speed reduction.

With this difference in mind, we have to go back a few steps to the recording of the sync audio master to understand how we achieve correct pitch and program length through many generations. We played back the non-sync tape at its correct speed while recording time code referenced to a 59.94Hz NTSC sync generator. (The 4-track machine should be resolved during all transfers.)

Next, that code, plus the song, was transferred to the video edit master tapes, again with the time code and the video edit master tapes, again with the time code and the video decks locked to the same 59.94Hz sync source. The speed of the song, therefore, has never changed: If it lasted four minutes when played back at exactly 15ips on the non-sync mix, it will still be at that length when played back from a properly resolved video or audio master.

However, we have to account for the fact that whatever the artist lip-syncs to during the shoot will be slowed down during the film-to-tape transfer. The answer is simple: we must speed up the song while it is playing back on the set, thus negating the later slow-down. Then, although the song was "too short" during photography, and the musicians were playing slightly fast, they will be in sync with the 4:00 song after the slow-down.

To ensure that the speed changes go up and down together on the set and during telecine, the sound on the ¼-inch playback tape must be recorded with a 59.94Hz sync pulse, but resolved against a 60Hz crystal reference as shown in Figure 4. When the tone on the tape is

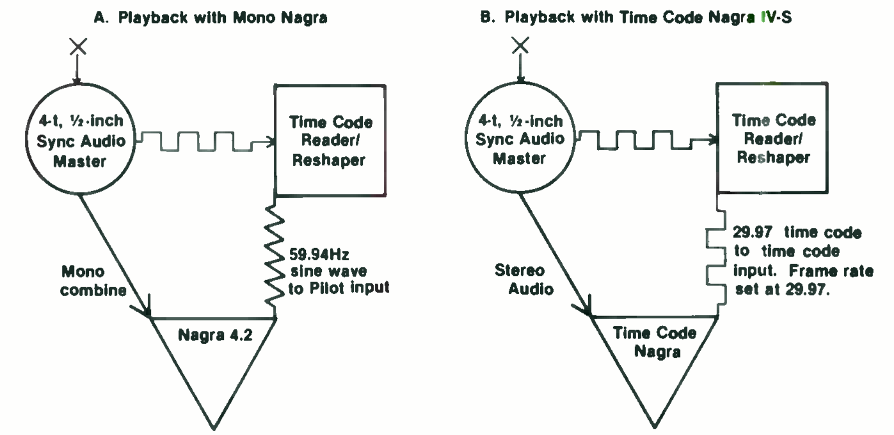


Figure 3. *Creating the ¼-inch Nagra playback tapes.*

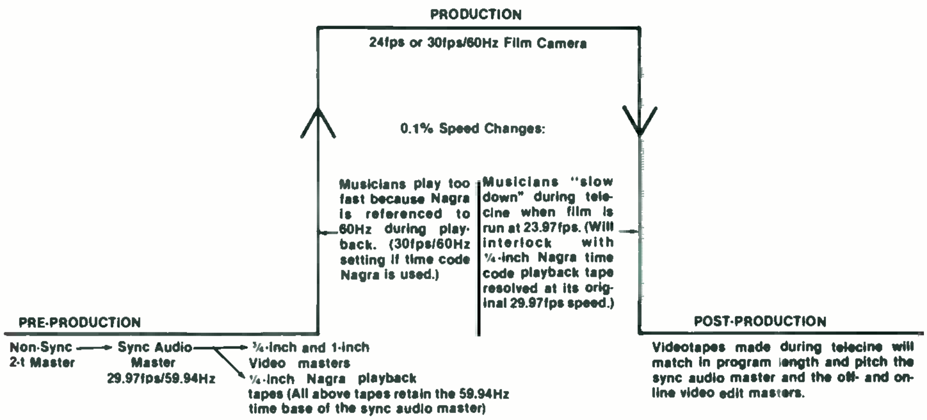


Figure 4. *Speed changes during production of music video shot to playback on 24fps or 30fps film and edited on videotape.*

War stories in the Tri-state area: Case Story no. 1

Film was photographed in 16mm with an Eclair NPR camera at 24fps referenced to 60Hz, while sound was recorded on a mono Nagra 4.2 with its internal crystal printing neopilot 60Hz sync pulse. All material would be transferred to videotape for off- and on-line editing.

In order to interlock the sound with the picture during telecine, it was decided to transfer the original ¼-inch tapes to a second ¼-inch tape, this time printing time code on track #2 while recording the mono program audio on track #1. The initial sync point would be found using the clapstick (rocking the tape over the head to find modulation), and then the synchronizer would use the time code on track #2 to preroll the audio tape and maintain lock during the mostly 11-minute, 400-foot camera runs.

Once sync was found on the first take and the telecine locked to time-coded ¼-inch tape, everything seemed fine. A minute or so into the take, something started feeling funny. Eventually, picture and sound started walking away from each other and all in attendance were sure: something was wrong.

It was discovered that the time code generator used to make the second-generation telecine ¼-inch tape had been referenced to an NTSC sync generator, and thus was referenced to 59.94Hz, and recording 29.97fps time code. Because the Nagra self-resolved using its internal 60Hz crystal, that same speed would be maintained during telecine when all machines, synchronizers and time code generators were referenced to NTSC sync.

So, instead of the sound slowing down to match the picture that was undergoing a time shift, the sound was staying the same, quickly getting ahead of the picture.

The moral of the story so far: You should not change the speed of the time base reference without also changing the speed of the program.

The transfer was tried again at the same facility, this time with the time code generator referenced to its internal 60Hz crystal, thus putting out true 30fps time code.

However, once again during telecine the sync that was verified by the clapstick at the beginning of the take didn't last for long. After examining the telecine system with test material, the crew went back to the original field tapes.

Playing them back on Nagra 4.2 with a QSLI resolver board installed, a 59.94Hz field rate sinewave stripped from the NTSC house sync was fed into the pilot input. Starting the Rank Cintel flying spot scanner and the Nagra at the same time resulted in perfect sync throughout the 11-minute take.

Later on, the accuracy of the time code generator used to make the second transfer was verified to be well within tolerances. Therefore, a true 60Hz/30fps time code had been recorded on track #2.

Therefore there appears no logical reason why the second transfer didn't work. The only variable is that the Nagra was self-resolving; this too, shouldn't have been a problem given the accuracy of the crystals in Nagras.

The only sure answer is that the safe way is the best way. The third (¼-inch neopilot to ¼-inch with time code) transfers that did work were done by feeding a field rate sine wave into the Nagra's pilot input, thereby assuring perfect lock relative to the time code. In essence, this procedure "copied" the neopilot sync pulse, converting it into time code. (It doesn't matter whether the field rate of this time code was 59.94 or 60.)

Moral of the second part: whenever possible, reference all sync sources to the same generator when making transfers.

locked to the crystal, the tape will be playing slightly faster because the resolver blissfully assumes that the sync reference on the tape is 60Hz. The synchronizer does *not* know the time base of the original recording; all it does is phase-lock the two tones. This procedure—purposefully playing a tape back at a time base other than its original reference—has come to be known as "cross-resolving."

In the current example, the ¼-inch playback tape will be recorded while feeding into the "pilot" multi-pin connector of the mono Nagra 4.2 an 59.94Hz sinewave stripped from the time code that is playing back on the 4-track audio master. Note that this means *taking out* the "sync bug" that is usually attached to the pilot plug, and which jumpers the crystal output on pin #3 to the pilot input on pin #4. (Seeing this "SK" jumper plug attached to the side of the Nagra is a security pacifier to production mixers but, in this instance, it is poison.)

If the only connection between the 4-track audio master and the Nagra is the audio track, and sync plug is in, you are recording at a 60Hz reference and will play back at that time base during shooting. No speed-up will occur and the song will last four minutes during shooting. Because of the slowdown during telecine, musicians will be playing "slow" and the song will now be 4:00:07 on the videotapes created during telecine, which will *never* match the off-line edit master that we made earlier.

In short, you will have a serious problem.

Center-track time code

The previous discussion assumes the use of the mono Nagra 4.2, which records the neopilot sync track down the middle of a full-track mono signal. The 4.2 will cross-resolve (during playback) and proper time base will be maintained.

The problem with using only a 60Hz pulse is that it contains no location information. In film musicals, this dilemma is sidestepped by using a second machine to record the playback track, with sync point obtained with the standard clapstick. During editing, each take can easily be matched by ear to a master uncut print of the playback track.

The situation is a bit more difficult in videos because rarely will the crew use a second, recording Nagra. Even if a track is recorded, telecine is an expensive time to try to match clapsticks with "wild" starts and non-interlocked sound.

Thus, on videos employing playback from a lone mono Nagra, the editor receives silent videotapes of drummers drumming and singers singing, with no way of knowing, other than detailed script notes, where in the song this take occurs. Even then, precise, frame-accurate sync—matching the audio on the off-line edit master to the footage shot—requires that the editor eyeball each take. The answer, of course, is to play back the tape on a Nagra IV-S TC with center-track time code.

(The Coherent Communications TC-200A time code module, used with a standard sync Nagra IV-SL, offers virtually all of the features of the Nagra IV-S TC, except that the TC-200A does not provide a time code reader. Also, since it records the time code as part of the normal FM sync track, it must be demodulated before it can be used in time code form.)

The only difference during the transfer from the sync audio master is that the time code itself, and not the sinewave representing the field rate stripped from

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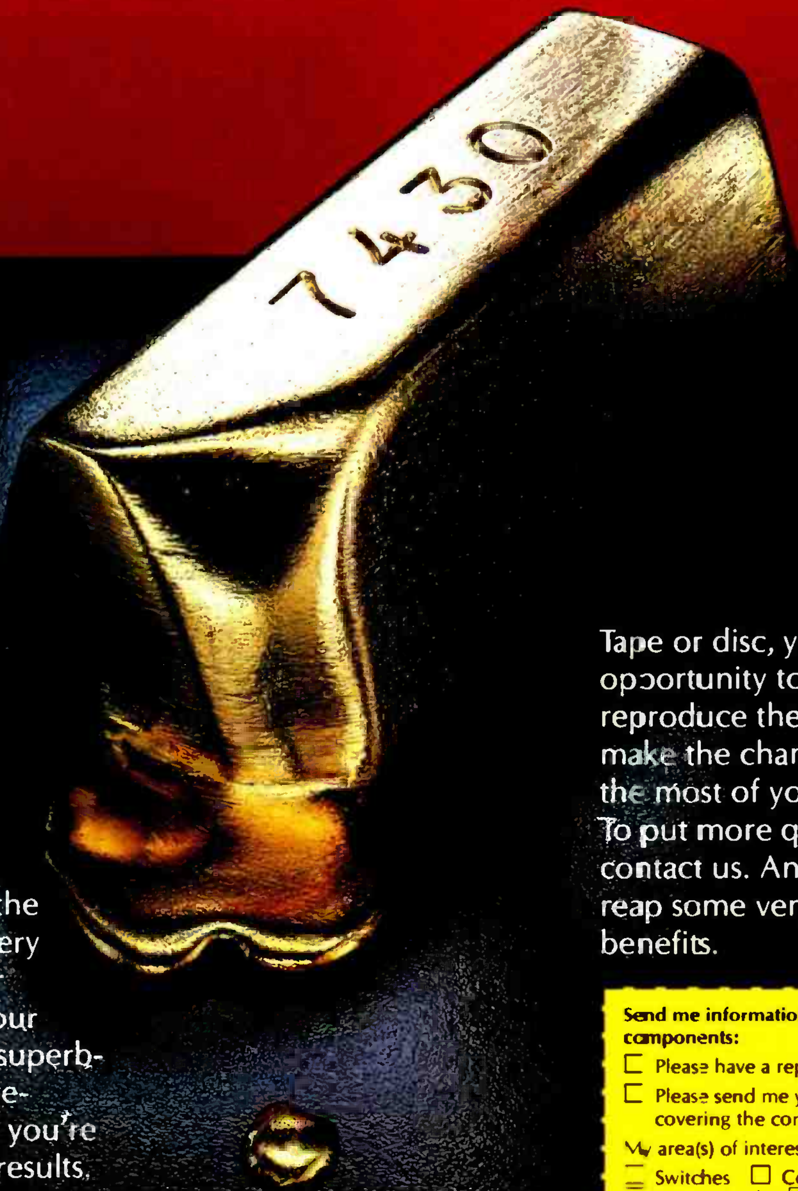

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Circle (32) on Rapid Facts Card

Table 1. Common U.S. frame rates for film and video production.

Recording Medium	Frame/Reference Frequency of Camera	Editing Medium	Editing Frame/Field Rate	Frame/Field Rate of Playback Sync Audio Master	Frame Field Rate During Playback
Film	24/60	Film	24/60	30/60 or 24/60*	30/60 or 24/60
Film	24/60	Tape	29.97/59.94	29.97/59.94	30/60
Tape	29.97/59.94	Tape	29.97/59.94	29.97/59.94	29.97/59.94

* Use 24fps time code only if time code is recorded on the film by the camera; use 30fps time code in 24fps film production only when using time code slates.

The next two installments

The next two installments in this 3-part series will be published in the September and October issues.

- *The second part will focus on the use of time code for film production, including a look at current time code-on-film systems.*
- *In October we will present a basic overview of synchronization and time code, explaining the uses of VITC, jam-syncing and reshaping/regenerating time code.*

Both parts will contain various "War Stories" relating synchronization problems that fellow professionals have run across. Names will be omitted to protect the guilty.

the code, is recorded on the playback tape. In our hypothetical shoot, we will be cross-resolving during playback, so we will have to maintain the 59.94Hz time base of the sync audio master. Therefore, the time code frame rate switch inside our IV-S TC must be set to 29.97fps (with drop or without drop, depending on the editor's choice) when recording the playback tape. On the set, however, the machine must be switched to 30fps (again, with drop or without drop) to resolve to 60Hz, thereby speeding up the tape to anticipate the pull-down in telecine.

Playback with a IV-S TC requires some extra equipment, the total cost of which is around $180 per day, including time code slate and wireless link. In addition to the tape machine, you need an external QSLS synchronizer and ARPC time code resolver box that converts the code into a 60Hz pulse that the QSLS can read. (Cooper Sound Services of Los Angeles, designers of the ARPC, now offer a dedicated time code resolver.) The ARPC also provides a time code output to drive an electronic time code slate that is photographed prior to each take. (When shooting video, the code is sent directly to the cameras.) [Refer to Table 1 for details of frame rates employed at various stages of film and video production.]

Approximately five seconds of good code must be photographed by the camera prior to shooting the scene. Tail slates can be used, although this requires extra handling of the negative, with an increased chance of dirt especially when shooting in 16mm.

It should be noted that not all telecine suites are equipped to interlock the film-to-tape transfer machine, such as the Rank Cintel Flying Spot Scanner, to a ¼-inch deck. The process is straightforward: time code, as photographed on the slate, is read off the monitor and entered into a keypad, thus giving the film a time code address that will be synthesized from tach pulses on the telecine. The ¼-inch deck containing a playback tape searches for this time code location in the song, picture and sound are both given a pre-roll, and sync is achieved.

What version is heard?

Once the off- and on-line editing stages are completed, and there exists picture to match the song, it's time to ship the film to music-video networks and shows. What should we send them?

Assuming that the song will play as recorded on the sync audio master, with no deletions or additions, then make a 1:1 copy of the audio master and ship that with a dub of the 1-inch video master.

All too often it is assumed that the sound will replay from the 1-inch master; this means that your song will never sound better than the audio performance of your average C-format VTR. Additionally, the on-line video master was itself a copy of the audio master, so what would be shipped is already a pair of 1-inch generations away from the sync audio master.

Instead, it makes much more sense (not to mention being considerate of the artist, producer and engineer) to provide 4-track, ½-inch copies to everyone who receives a 1-inch video copy. You not only save one generation, but you see to it that what is *heard* involves only standard audiotape recorders. Despite the claims of high standards that many video post facilities make for their 1-inch machines, as a rule the degree of audio tweaking we expect with tape machines is just not carried out with VTRs.

All of which is by way of reiterating the earlier statement that, in the best of worlds, the audio laid on ¾-inch and 1-inch video masters is for editing reference *only*. However, because there is no way of knowing what is going to happen to the video once it leaves your hands, it is prudent to assume the worst case (i.e., the audio will go through as many 1-inch generations as the picture). The use of noise reduction, therefore, on both the ½-inch audio master and the 1-inch on-line video master, is probably a good idea.

Again, it is acknowledged that digital recording at any point in the chain can be a big help with reducing the generation loss.

The most important ally in your quest for sync is communication. Make sure that you know what the "video" will be shot on (and, if film, at what frame rate); what it will be edited on, and what the contractual requirements are. If you are unclear as to what to do, ask questions. And, most importantly, do not stop asking until you are *sure* you know what you are doing and are also sure that other concerned parties are doing their jobs. [Part two of this series, to be published in the September issue, will cover the use of time code for film production. R·E/P

Bibliography

This series of articles on time code and synchronization will concentrate on the nuts and bolts operating procedures. If what you are looking for isn't covered, the publications noted below should be of some help:

- **Time Code Handbook,** *by Walter Hickman and Milan Merhar, 1982. The single best introduction to time code. Available from: Bang-Campbell Associates, 3 Water St., Box 47, Woods Hole, MA 02543; $8, postpaid.*
- **Using Time Code in the Reel World,** *by Jim Tanenbaum and Manfred N. Klemme, 1987. A must-have pamphlet covering the operation of the Nagra IV-S TC in detail. Available from: Nagra Magnetic Recorders, 6043 Hollywood Blvd., Suite 201, Hollywood, CA 90028.*
- **Manual of Sound Recording,** *by John Aldred (Argus Books, London: 1978). The chapters on Motor Systems and Motion Picture Sound provide the best available overview of film synchronization techniques.*
- **Understanding Synchronization,** *1986. In addition to basics of time code usage, this book also touches on the uses of MIDI. Published by the TEAC Corporation of America, 7733 Telegraph Road, Montebello, CA 90640.*
- **The Time Code Book,** *1984. A quick overview of time code applications. Published by EECO Inc., 1601 East Chestnut Ave., P.O. Box 659, Santa Ana, CA 90272-0659.*
- *"New SMPTE Time Code Applications in Film," by David Howell.* **Millimeter,** *April 1981 issue, pp. 59-68.*

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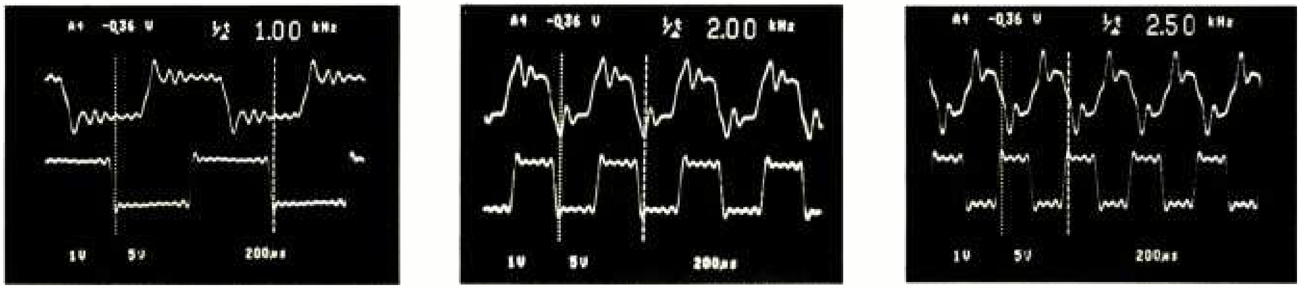
Circle (46) on Rapid Facts Card

Design Parameters of Digital Anti-Aliasing and Anti-Imaging Low-Pass Filters

By Bruce Jackson and Christoph Heidelberger

There has been much debate in the recording industry regarding the audio quality of low-pass filters used in all PCM digital systems. Recent developments suggest that the sonic quality of such systems can be radically improved with replacement linear-phase filters.



These three oscilloscope traces show the 1kHz, 2kHz and 2.5kHz square-wave response of three cascaded channels of a digital multitrack, to simulate the minimum number of A-to-D-to-A conversions of recording, remix and CD mastering. The upper trace for a multitrack equipped with standard anti-aliasing filters should be compared with the response of the same transport with replacement filters (lower trace).

While it's not unusual for a recording engineer to take great care in selecting a favorite microphone, mic pre-amp and special equalizer for a particular session, the resulting audio signal will then be passed through a recording chain over which the engineer has no control.

By taking a look inside current digital tape recorders, we hope to provide the user with a better perspective and understanding of a critical component that is sometimes the weak link in a recording chain: the low-pass filter.

Although such anti-aliasing filters have various forms and characteristics, they are common to all digital systems that involve conversion to and from analog, including digital reverbs, delays, synthesizers, processors, tape and disk-based recorders. For clarity, we will restrict our comments to professional multitrack and 2-track digital recorders, although the information presented applies to digital systems in general.

We are all familiar with the advantages that digital recording offers over analog techniques, including potentially highly accurate data storage, transmission, manipulation and retrieval. Why, then, do we still hear expressions such as "brittle," "harsh," "edgy," "crunchy" and "fatiguing" used to describe this highly accurate medium?

This is by no means a new question. A large volume of excellent work has been done by many researchers in identifying the various distortion mechanisms and aberrations, in the process of converting our reference analog audio voltages into a stream of 16-bit numbers and back again.

Our interest in digital low-pass filters began three years ago when we experimented with IC chip sets designed for CD players. As end users of various digital audio processors, we were frustrated with their sonic shortcomings. Having coupled a Philips digital oversampling filter and digital-to-analog converter to our experimental A-to-D converter, which incorporated an anti-aliasing filter with linear-phase response and a gentler rolloff characteristic, the sound we heard had a realism neither of us had experienced before with digital devices. It was obvious we were on to something.

We then proceeded to test a wide variety of experimental filters, using Mitsubishi and Sony multitracks as test beds. From the widely ranging sonic results, it was obvious that their low-pass filters were affecting the audio band they were designed to pass transparently.

Digital audio involves the sampling of an input signal at a certain rate, a process that can accurately represent a continuous time-varying signal with a bandwidth no higher than half the sample frequency. This bandwidth limit is called the Nyquist frequency, after a pioneer in information theory, Harry Nyquist. For a PCM system with a sampling frequency of 44.1kHz, the Nyquist cutoff frequency is (44.1/2) = 22.05kHz.

When we attempt to sample frequencies beyond the Nyquist limit, they in-

Bruce Jackson is chief engineer and Christoph Heidelberger is director of technical developments of Apogee Electronics, Santa Monica, CA.

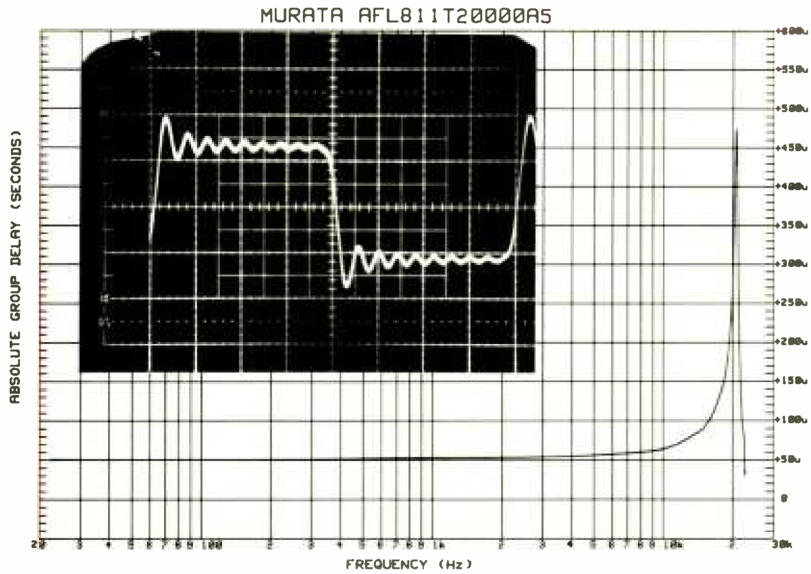


Figure 1. *Absolute group delay of a Murata AFL-811T20000A6 low-pass filter, with a superimposed 1kHz square wave response trace.*

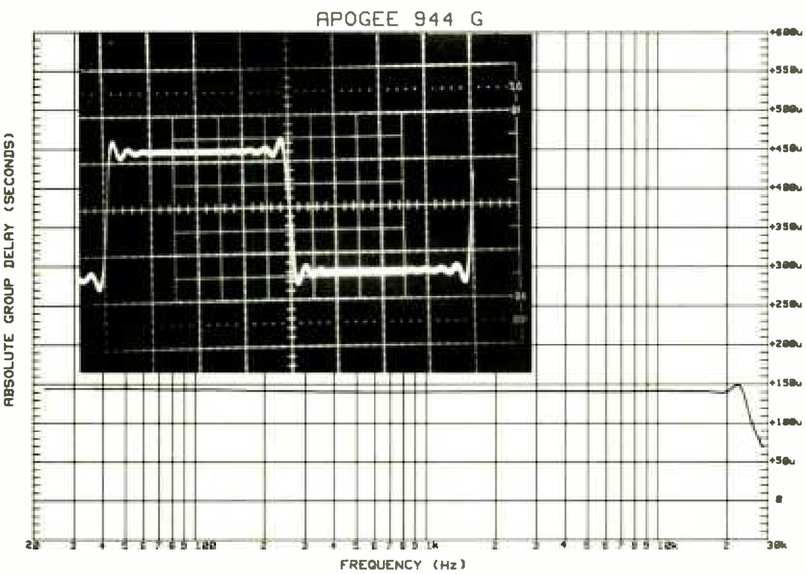


Figure 2. *Absolute group delay of an Apogee 944G low-pass filter, with a superimposed 1kHz square wave response trace.*

teract with the sampling process and fold back down below the Nyquist frequency. In audio-band sampling systems, this effect means that ultrasonic signals beyond the Nyquist frequency can generate unwanted or false audible sounds unrepresentative of the audio signals we are attempting to digitize.

These unwanted, false signals are called aliases. An anti-aliasing filter is inserted in the audio path to remove, via low-pass filtering, any signals that could cause aliasing. Such a low-pass filter would not be necessary if we knew for sure that our signal of interest contained no frequencies above the magic Nyquist frequency.

The majority of today's anti-aliasing low-pass filters used with 16-bit linear pulse code modulation (PCM) systems were designed to exhibit as flat a frequency response as possible in the audio pass band, and then attenuate as rapidly as possible past their cutoff frequency. Achieving the extreme slope and deep stop-band attenuation of such low-pass filters is no easy design and manufacturing feat. It requires the acceptance of certain compromises and side effects.

Any design requires just the right balance of tradeoffs; when these low-pass filters were designed, it was believed that alias protection was of paramount importance. The general consensus exsisted that the ear was sensitive to potential aliasing products, but insen-

Anti-aliasing/Anti-imaging low-pass filters

For the purposes of comparison, this article includes figure illustrations and data derived from filters used in digital multitracks and 2-track transports. The filters considered here include:

- ***Murata AFL-811T20000A5**—this was the standard filter used in the Sony PCM-3324 DASH-format digital 24-track before the introduction of the Soshin (see below). It is almost identical to the Murata AFL-811T20000A6 used in the Mitsubishi X-800, X-850 and Otari DTR-900 32-track PD-format machines, the main difference being the inclusion of an output buffer in the A6 version.*
- ***Soshin 20kHz**—this unit now supercedes the Murata in the PCM-3324.*
- ***Sony BH-106**—the linear phase filter used in the input stage of the Sony PCM-1630 processor and PCM-3202 DASH-format 2-track.*
- ***Apogee 944S**—a pin-compatible, linear-phase replacement for the Murata 811T20000A5/A6 and the Soshin.*
- ***Apogee 944G**—an alternate pin-compatible, linear-phase replacement for the Murata 811T20000A5/A6 and Soshin, but with a gentler rolloff slope.*
- ***Sony 170-11**—standard filters used in the Sony PCM-F1 (and related EIAS-format processors).*
- ***Sony DTC-1000Es**—an R-DAT recorder.*

Measurement techniques

The filter response measurements were made at Deane Jensen's laboratory using the new Hewlett-Packard HP5183T waveform recorder and tailored pseudorandom noise waveforms programmed into a Wavetek 275 Arbitrary Waveform Generator. The 12-bit dithering 4 megasample/second digitizer with COMTRAN software yields network transfer-function measurements up to 1.66MHz with certified accuracies of ±0.002dB.

An additional HP3325A frequency synthesizer is used as an external clock with a closed-loop triggering configuration to achieve precision for time delay and phase measurements of 1ns, equivalent to 0.01° at 25kHz.

Response errors of the test equipment, fixtures and cables are canceled using deconvolution, defined as FFT to input and output waveforms followed by complex division. The resulting transfer-function data file is used to plot graphs for magnitude, deviation from linear phase, group delay, step response and square wave response.

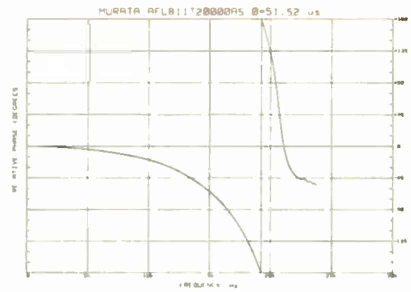


***Figure 3.** Deviation from linear phase response for a Murata AFL-811T20000A6 low-pass filter. (Note that the data presented here contain the same information as that of Figure 1, except that the X-axis is now calibrated in deviation from linear phase, rather than absolute microseconds.)*

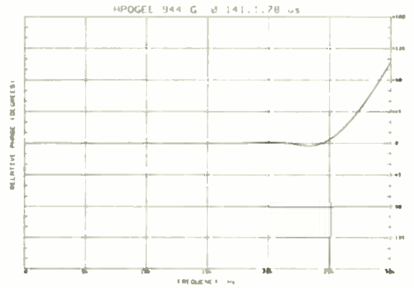


***Figure 4.** Deviation from linear phase response for an Apogee 994G low-pass filter. (Note that the data presented here contain the same information as that of Figure 2, except that the X-axis is now calibrated in deviation from linear phase, rather than absolute microseconds.)*

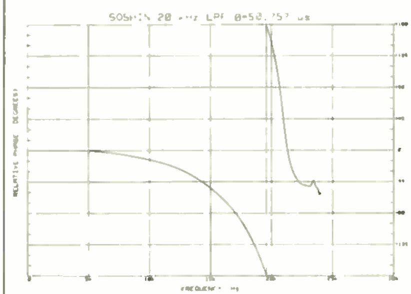


***Figure 5.** Deviation from linear phase response for a Soshin low-pass filter.*

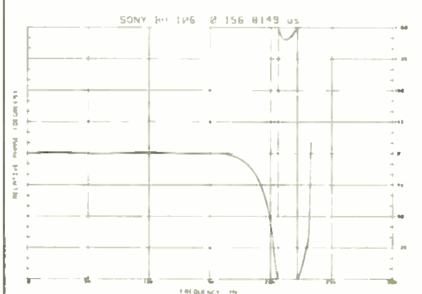


***Figure 6.** Deviation from linear phase response for a Sony BH-106 low-pass filter.*

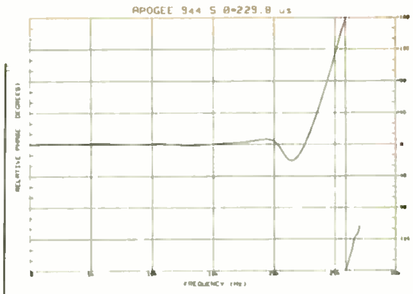


***Figure 7.** Deviation from linear phase response for an Apogee 944S low-pass filter.*

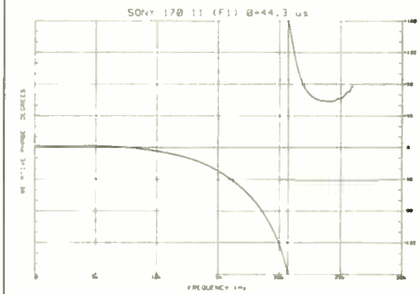


***Figure 8.** Deviation from linear phase response for a Sony 170-11 low-pass filter.*

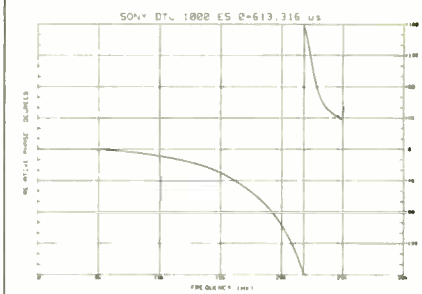


***Figure 9.** Deviation from linear phase response for a Sony DTC-1000 R-DAT recorder. (Note that the response shown here is for the recorder's overall record/replay process at a sampling frequency of 48kHz. However, because of oversampling at the unit's D/A output section, the response shown here is essentially that of the low-pass input filter circuits.)*

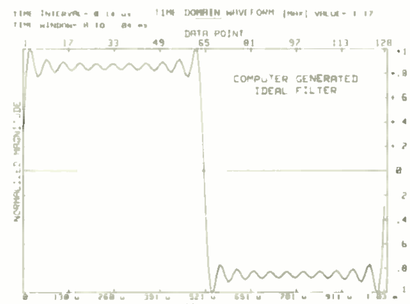


***Figure 10.** Computer-generated 1kHz square-wave response of an "ideal" 20kHz anti-aliasing low-pass filter.*

sitive to the by-products of using filters that had steep rolloff slopes.

Although these design criteria produced excellent-looking technical specifications, in hindsight, they underestimated the ear's true sensitivity to filter by-products.

Harmonic skew or smear

An ideal low-pass filter, passing audio frequencies to the highest frequency of interest, should exhibit—in theory at least—no sound of its own. A major contributor to the sound coloration of anti-aliasing and anti-imaging filters common today is their different time delay at different frequencies—in other words, frequency-dependent delay. A device that exhibits a constant delay with varying frequency is said to exhibit a flat group delay; expressed in degrees of phase shift, the phenomenon is called linear phase response.

Filters used in the current generation of digital multitracks do not exhibit a constant delay vs. frequency. The delay at lower audio frequencies is around 50μs, increasing to around 60μs-70μs at 10kHz, 275μs at 20kHz and a peak of 475μs at 22kHz. The 50μs is pure delay, or a measure of how long you have to wait to hear the signal pass through the filter. The difference between 50μs and 275μs to 225μs is the group delay distortion, or deviation from flat group delay at 20kHz as shown in Figures 1 and 2.

The consequence of this deviation from flat group delay (or deviation from linear phase, measured in degrees) is that higher frequencies take longer to pass through the low-pass filter. Musical instruments express their *personality* and stereo image with a complex set of harmonics, all of which are related to the fundamental in both amplitude and phase.

When these low-pass filters were designed, it was thought that the ear was insensitive to phase information, and that just preserving the amplitude information was sufficient to accurately represent the sound. It is obvious that this approach is a convenient method of sweeping the distorted group delay under the rug, so to speak, because each time complex harmonics passes through a filter, the fundamentals become further separated from the higher harmonics and, at some point, that time difference will be clearly audible.

Independent listening tests[4] have shown that the threshold of audibility of distorted group delay is much lower than previously thought. With the right type of source material, listeners report that listening through one generation of filters (one on the A-to-D side and another on the D-to-A) produces audible results.

Harmonic signature

To visualize the above distortion, imagine a simple recording chain involving just two digital audio generations. If an acoustic guitar is recorded onto a digital multitrack and then mixed via an analog board to a digital 2-track, during playback the signal will have passed through four low-pass filters. If these are the standard type of filters in common use today, the complex higher harmonics that express the guitar's subtle nuances will be separated from the lower fundamental by almost 1mS at very high frequencies. In that period of time, sound travels approximately 1 foot—the effect is like moving the tweeter in your nicely coherent speaker system away from you by 12 inches!

Admittedly, this long separation of time occurs at almost inaudible frequencies. From an analysis of Figures 3 thru 9, however, it is obvious that at even moderately high frequencies the harmonic signature of musical instruments has been altered by filters that deviate from a perfect linear-phase response.

Testing with square waves

There have been quite a few theories offered to explain the poor square-wave response of digital systems. We read where a digital audio designer suggested that square waves can tell us a lot about transformers and amplifiers, but they tell us very little about digital audio. "It is probably best that you don't look at them," the designer concludes[5].

We disagree. Although there is no substitute for actually listening to music, square-wave testing does provide much information in one scope trace.

Although these criteria produced excellent-looking specifications, they underestimated the ear's true sensitivity to filter by-products.

A pure sine wave contains no harmonics other than the fundamental. When we use it to test a filter, we are looking only at one narrow area at any given moment. A square wave, on the other hand, presents a broad view because it is rich in odd harmonics extending far beyond the pitch of the fundamental.

The only time we recover an exact square wave at the output of the device under test is when these harmonics are added to the fundamental sine wave in just the right relationship of amplitude and phase. Viewing a precise square wave is an easily recognized, albeit rough, electronic analogy to undistorted

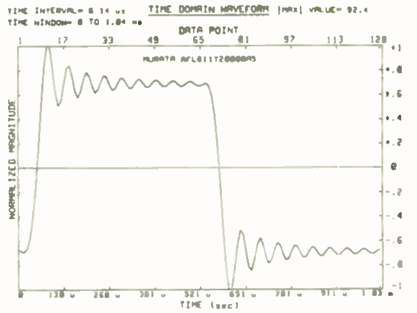


Figure 11. Calculated from transfer-function measurements 1kHz square-wave response of a Murata AFL-811T20000A6 low-pass filter.

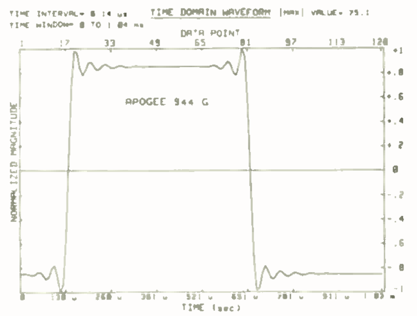


Figure 12. Calculated from transfer-function measurements 1kHz square-wave response of an Apogee 994G low-pass filter.

harmonic signatures of musical instruments.

Why don't we see pure square waves with digital?

Sampling rate standards for digital tape recorders define our audio bandwidth. When we filter an analog signal prior to conversion to digital, we intentionally remove the harmonics that could cause aliasing. If we filter a square wave with an "ideal" low-pass filter as shown in Figure 10, we remove the upper harmonics that pull the component sine waves into a square wave. Without upper harmonic sine waves to build a nice clean edge, we are left with some overshoot and some bumps. This effect is called the Gibbs phenomenon.

When the upper harmonics that do make it through the filter are shifted into a different time frame, they can cause a large increase in overshoot. Increased overshoot reduces the dynamic range, and can be compounded through each successive generation of filters. Dynamic range can be reduced by up to 1dB with typical filters for each stage of filtering.

You'll notice from the accompanying square-wave responses in Figures 11 and 12, that each filter has a characteristic ringing frequency. In general, steeper cutoff filters tend to ring with a greater amplitude and take longer to decay. In most of the present-day filters, the ring-

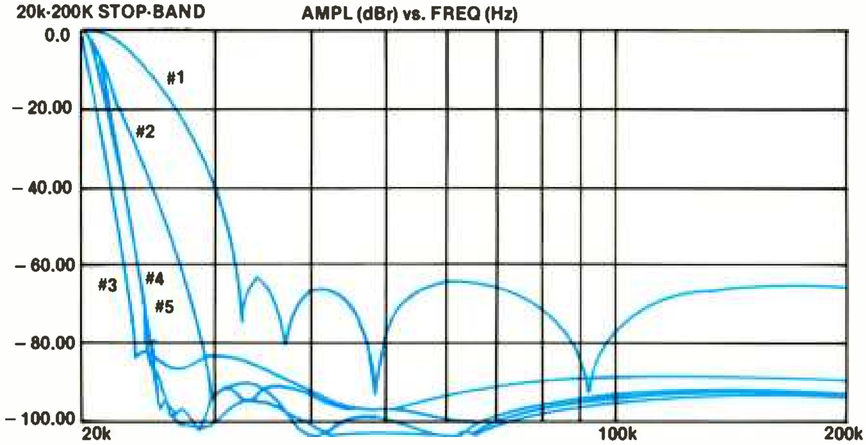


Figure 13. *Rolloff and stopband response of Apogee 944G (*1), Apogee 944S (*2), Sony BH-106 (*3), Murata AFL-811T20000A6 (*4) and Soshin (*5) low-pass filters.*

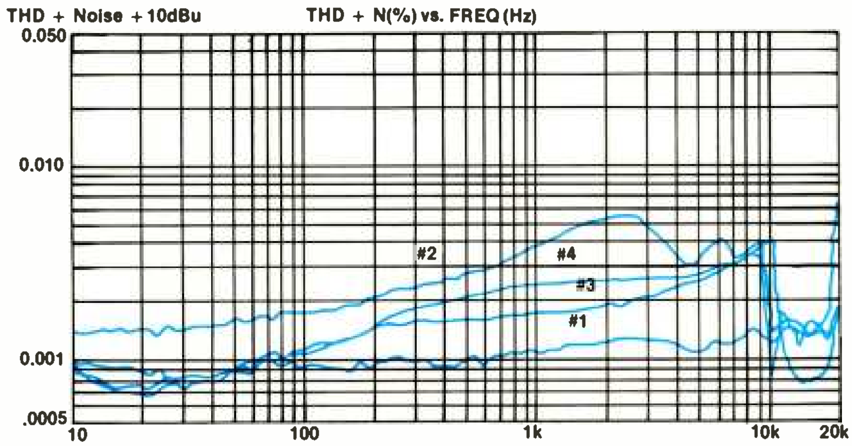


Figure 14. *Noise and total harmonic distortion performance of Apogee 944S (*1), Sony BH-106 (*2), Murata AFL-811T20000A6 (*3) and Soshin (*4) low-pass filters.*

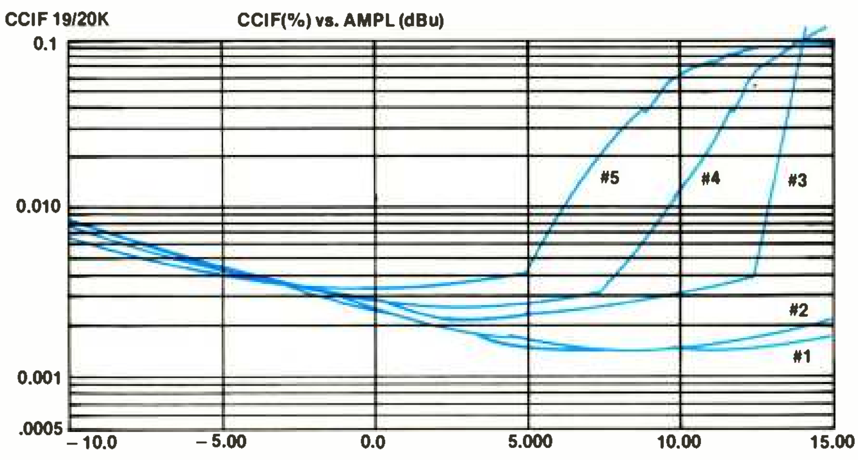


Figure 15. *CCIF intermodulation distortion performance of Apogee 944G (*1), Apogee 944S (*2), Sony BH-106 (*3), Murata AFL-811T20000A6 (*4) and Soshin (*5) low-pass filters.*

ing frequency occurs within the audible range. The Murata AFL/A6 has a measured initial ringing frequency of 16.5kHz, the Soshin—16.7kHz, the Sony BH-106—19.1kHz, the Apogee 944S—21.75kHz and the Apogee 944G—23.75kHz.

Our experiments with a variety of flat group delay, gentle and sharp rolloff filters suggested a definite audible advantage was to be gained from using a gentle slope. Researchers Roger Lagadec and Thomas Stockham have pointed out the audible effects of using steep rolloff filter shapes. Existing filters work hard at protecting our ears from aliasing below the threshold of hearing; ultra-steep rolloff and 90dB of attenuation assumes that we will be recording audio signals containing full amplitude beyond 20kHz!

In the real world, however, it is rare to find high amplitude signals at frequencies much higher than 10kHz and, beyond that, levels tend to steadily decrease with frequency. Even high-frequency bias leakage from analog tape machines is substantially down in level at frequencies that can cause aliasing.

Input filter requirements can be relaxed, therefore, with the knowledge that aliasing frequencies are already naturally attenuated and masked. These filters are referred to as "gentle" although, as can be seen from Figure 13, this is only a relative term.

A growing number of digital multitracks have been retrofitted with gentle rolloff, linear phase anti-aliasing filters, and users universally praise the sonic improvement they provide.

Anti-imaging filter

Output filtering following D/A conversion is common to all PCM systems, and is designed to remove the high-speed switching transients that cause ultrasonic repetitions of the audio signal. These images of the audio signal are beyond human hearing, but can cause distortion in inputs designed for much lower frequency audio. This anti-imaging or reconstruction filter cannot remove any aliasing. In many digital audio systems, the anti-imaging filter is *identical* to the anti-aliasing filter and, as a result, suffers from many of the same shortcomings.

Noise and distortion

Traditional noise and THD measurements of filters can be misleading above a frequency of 10kHz, because the filter can mask premature clipping and op-amp distortion by removing the telltale harmonics. Intermodulation tests (such as the CCIF 19/20kHz technique) can reveal hidden distortion, as shown in Figures 14 and 15.

A common by-product of achieving a

steep filter rolloff curve is a series of bumps and dips in the audio band that we wish to preserve. With careful design and manufacture, these ripples can be minimized and, in some cases, eliminated.

Component choice

Analog anti-aliasing and anti-imaging filters are commonly mass produced as thick-film hybrid circuits, with conductors and resistors silk-screened and fired in a kiln. Such resistors can start their life laser-trimmed to a high accuracy, and then drift with time and temperature change. With incorrect mechanical termination, resistors can become slightly non-linear and produce distortion. Noise performance, especially after laser trimming, is inferior to that of metal-film resistors normally used in premium electronic circuitry.

With the advent of surface-mount technology, it is now possible to design miniature audio circuitry using premium components. High-performance audio op-amps are also available in miniature packages, and their use can improve filter intermodulation distortion by an order of magnitude (at higher input levels) over some of today's designs.

Using a different set of criteria for anti-aliasing filters that were designed a number of years ago, it is possible to design components that are better tuned to providing optimum sonic performance. Our company has developed a family of filters designed to retrofit to existing digital multitrack and 2-track recorders.

In closing, perhaps one of the best summaries of the situation came from acoustic consultant and designer Deane Jensen. After measuring our filters to yield the accompanying figure illustrations, he considers that "the Apogee linear-phase low-pass filters yield the best sonic improvement for digital audio since analog. These low-pass filters solve 'the unnatural top-end,' which has been one of the most common complaints of today's digital sound recording." R·E/P

Editor's note: The mention of specific products in this article is not to be taken as an endorsement by RE/P or Intertec Publishing Corp. The article has been written for the purpose of satisfying reader interest and educational needs.

Our thanks to Sony Corporation for its support and technical information; Village Recorders and George Massenburg for the use of their digital recorders; Deane Jensen for donating several weeks of his valuable time; and Eric Benton of Jensen Transformers for his tireless number crunching.

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Selecting and Using a Production Music Library

By Merelyn Davis

Commercial production music libraries provide a viable alternative for scoring video and film projects.

The author during a music spotting session at The Music Design Group, Hollywood.

Does the following sound like a familiar scenario? About an hour before the mix, your client calls to tell you that she may be needing music for her commercial spot/A-V production/video sales presentation/etc., and would you please pull a few stock music albums with some appropriate cues. A bit of digging under the console for the albums, a bit of creative fading and *voila*—you have a music track.

Unfortunately, such misuse of music libraries is all too common, and is partially responsible for the view held by many mixers and producers that library music is a second-best choice. In this article, I hope to help correct that view, and provide a few pointers on how to effectively score your next production using music libraries.

There are three important elements that must be present if you are going to carry out the procedures described in this article. One is that you and your client both realize that a good music score is integral to a good production, and that sufficient time and money must be budgeted for producing this score. Second, someone who has the knowledge and talent for producing such a score needs to be involved in the project from the start. Third, you need access to a good music library.

Merelyn Davis has been a music designer/editor with The Music Design Group, Hollywood, since its formation six years ago. Her latest projects include Martin Mull's *The History of White People in America*, *Playboy Video Magazine;* network shows *Valerie* and *Nothing in Common*, *The Third Annual Soap Opera Awards* and PBS' *Harold Clurman: A Life of Theater.*

Know your library

Before beginning any production, a thorough knowledge of your own (or intended) library is imperative—and I don't mean simply memorizing the descriptions for each cue found on the back of the album cover! You need to be familiar with the general musical contents of each album, and with your own personal feelings about each cue. For many, this familiarity leads to the development of a system of annotation and cross-referencing, often using a computer database.

On the other hand, it is important not to let a particular cue become associated with only one application. Some cues, by their nature, are limited in their usage. For example, a big orchestral ending is a big orchestral ending; but you could consider, under certain circumstances a Fifties rock-and-roll cue for, let's say, an assembly line segment.

In addition to noting the primary mood suggested by the cue, there are several other things you should be listening for, the obvious ones being style/musical period, tempo, instrumentation and performance quality.

A few of the not-so-obvious aspects include:

- The nature of the melodic line;
- The texture of the piece, which has to do with the number of instruments, their timbre and the way they are combined;
- The manner in which the music is mixed, including the nature of the ambience and reverb, and the prominence or distance of the lead instrument(s), bass line and percussion;
- The modality, key and meter, details of which will be covered later.

All of the above parameters are important, particularly if you are planning to

Selected music cues can be transferred to ¼-inch tape from vinyl albums, Compact Disc or reel-to-reel masters prior to fine tuning timings to match picture. The final edited master would then be transferred to a time code-striped ½-inch 4-track, or bounced up directly to the 16- or 24-track multitrack master prior to the sweetening and remix session.

The author among an extensive music library available at The Music Design Group.

combine individual cues to score a long production, or intercut different cues to form a single cue.

There are a variety of library formats that are important to note as you listen, because they can be helpful in certain situations. For example, although it doesn't occur often, some libraries offer music suites. These usually take up one side of an album and contain a *main theme*, along with several different cues based on that theme. As you can imagine, such music suites can come in handy when you need to maintain a sense of unity in a lengthy presentation. (In fact, this format is more akin to what a composer would actually do when scoring a production.)

Other libraries offer a main cue with melody, followed by the same cue without the melody—usually a matched-level mix that can be intercut with the main cue—followed by a pre-cut 60-second and 30-second version. You may find, as I do, that you would rather edit your own jingle-length cues. This technique allows you to include portions of the cue that apply to your particular spot, and to *catch* internal cue points. Also, I have found that the 60-second and 30-second pre-cut cues are not always accurate in their timings.

Aside from a few libraries which are available only on Compact Disc, most major libraries are now issuing only selected releases on CD. The majority of music library albums are still only available on LP and ¼-inch tape. Unless you live in New York or Los Angeles, where you can go directly to the library publishers and borrow tapes, ideally you should purchase both the LP and the ¼-inch—the LP for day-to-day auditioning, and the tape for transfer.

If you can't afford to purchase both formats, make yourself a tape copy of the LP when it is new. If the LP you have been issued is faulty, return it and request a new one. Always transfer at 15ips, stereo, using dbx or Dolby A-type noise reduction if possible, and leader and slate each cue. You will find it makes cuing easier if you use a 7-4 low-torque reel rather than a 10½-inch reel for your transfers. Of course, as more albums become available on Compact Disc, the above exercise will—fortunately—become unnecessary.

From time to time you will come across cues that are absolutely fantastic, although usually limited in their usage; a cue that sounds exactly like the Beach Boys, the James Bond theme or an impressive electronic glissando that you know you can use somewhere. If, like me, you think that you'll remember where the cue is when the time comes to use it, it's a safe bet that you won't. Keep a list of these cues as you come across them.

Also, if you find you need a cue that your library doesn't contain, call the library representative. If they have it, they will send it to you. In addition, a few libraries are now offering a scoring service and will have cues composed for your specific need, or have an already existing library cue rearranged to your specifications.

Designing the music track

The first thing to do when designing a music track for your presentation is to "spot" the show for those places where music will be needed. Ideally, you will be able to do this with the people who will be making the final decisions on music selection. Although the spotting stage must often be done from a script—and a rough draft at that—you should still try to view at least a rough-cut of the picture, a storyboard or a run-through of the slides.

The more you can be involved in the pre-production stages, the more successful and integrated the music track will be. If you can manage to talk to the person who conceived the show or who is writing the script, and provide them with some samples of the kinds of music that will be available when the time comes, the production can even develop with the music in mind.

For example, a really top-notch opening cue can inspire a graphic artist preparing the slides to new heights. I have even had a producer approach me before shooting specific scenes so that he could take away some selected cues on audiocassette for playback in the field—to aid him in "choreographing" the shoot.

Although I'd be the first to admit that the instances of this kind of cooperation are rare, sometimes all it takes is a suggestion to the right person.

It's a strange thing about producers and pre-production: there never is enough time or money to involve the music designer/editor ahead of production. But, when the show is finished, and music is the only thing that can save what turned out to be a poor production, the rush is on. Music *cannot* save a bad show; it can, however, help a mediocre one and, when properly used, definitely makes a good show even better.

the Music Design Group

MUSIC CUE SHEET

"A Day In The Country" DATE

Cue	Timing	Length	Reel	Description	Music	Instructions
1	00:10:29	:02	A	Hollywood sign	"California Girls"	cut in and out
	00:13:00					
2	01:18:29	:17	A	:04 :06.5 :10 circle graphic appears	KPM 1257-B2	cross fade tail with
	01:35:08					cue 2A
2A	01:35;08	:07	B	cut to photo	KPM 1257-B2	
	01:42:16					
3	03:29:21	:47	A	paintings on wall	BRI 6-A6	
	04:16:04					
4	06:28:17	:04	A	whirling graphic	PM 10-A8	segué end with cue 5
	06:32:02					
5	06:32:02	:33	B	battle graphic	MF 104-2	VARI 14.5 (96.7%)
	07:05:14					

An extract from the author's music cue sheet for the opening section of A Day In The Country, *a recent PBS special. The sheet lists the various cues, their in and out time code locations, lengths, accompanying visuals and music source, plus information that might be useful to the mix engineer during the sweetening session.*

During the spotting stage, keep in mind that there should always be a reason for using music in a particular place. For example, the music might help to tie several short, diverse segments together; to separate segments; to punctuate important statements or action; to move the action along; to set a period or to enhance a mood.

Regarding the length of cues, my own rule of thumb is that a single piece of music should last no longer than two minutes. If a cue needs to be longer, either find a piece of music with enough variety to sustain interest, or combine several shorter cues. On the flip side, you have to guard against using too many short cues that fall one after the other, a format that leads to a choppy feeling (unless, of course, they are being used in "newsreel" montage style).

Another prevailing rule of spotting—especially for documentary style—is that music should not be used while people are talking on-camera. This rule is often difficult to obey when the picture is switching back and forth between talking heads and action footage. In these cases, keep the music going but edit it in such a way that the least obtrusive portion of the cue falls under the on-camera dialogue.

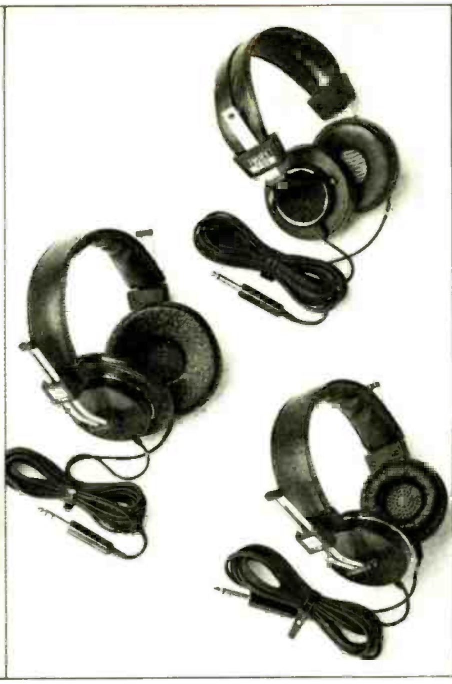
While you are viewing the script or visuals for the first time, jot down your initial feelings about the kind of music you *hear* for each segment. This fresh objectivity can be valuable, especially when everyone else there may have been working on the project for months, and may be short on fresh approaches. Make your spotting notes as detailed as possible in terms of mood, pacing and style, especially if you are not going to be seeing the show again for a while.

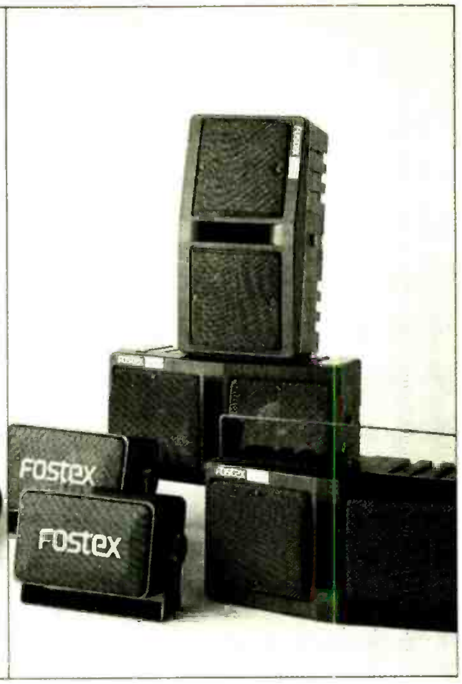
When it comes time to do the final spotting—for example, the one from which you will be taking final timings for the music cues—it's best to work with a videocassette copy of the presentation that has continuous time code burned into the picture. By using a VCR remote control with a search controller, you will be able to notate the first and last video frame of each segment needing music, calculate the total time and keep track of running times on all of the internal points that you want to catch musically.

It is also valuable to have ready access to a videotape while you are choosing the music, since your memory of what *should* work, and what actually *will* work, are often different. It is important that you have accurate timings, for you are going to be editing music to "fit" the picture rather than the "fade-it-in-and-out-and-hope-something-catches" method.

Selecting cues

Choosing cues is the most crucial and the most difficult part of the process. It is at this point in the proceedings that





Microphones

In the studio, over the air or up on stage, there's a Fostex RP mic specifically designed for the job at hand. RP stands for regulated phase, a transducer technology which has been awarded over 20 international patents to date. These mics have the warmth of condensers, the ruggedness of dynamics and a sound as transparent as it gets.

Headphones

These are more outstanding examples of RP Technology. Model T-20 has become almost legendary among studio musicians, producers and engineers. Its flat response at any listening level and its comfortable design help you listen longer without fatigue. And the sound is so clear and well-defined, critical listening is enjoyable.

Speaker Systems

You're up & going with Fostex PA systems. Modular designs let you control the sound according to the needs of the event. Stack them, gang them. From a simple portable PA to an entire rig, look to Fostex speaker systems to help you solve your sound problems.

Powered Mixers

Model MP-800 has 8 inputs and delivers 180 W per channel and Model MP-1200 has 12 inputs and delivers 250 W per channel. These rugged, road worthy stereo mixers have quiet running fans, digital echo, normalled connections at all critical patch points, stereo graphics on the mains and super monitoring flexibility. The best at any price.

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(213) 462-2404

MUSIC CLEARANCE LOG

TITLE: "A Day In The Country" (PBS)

Cue	Length	Use	Composer	Publisher	Society
"California Girls"	:02	BGV	B. Wilson	IRVING MUSIC, INC.	BMI
KPM 1257-B2	:17	BGI	R. Harvey	KEITH PROWSE MUSIC	ASCAP
KPM 1257-B2	:07	BGI	R. Harvey	KEITH PROWSE MUSIC	ASCAP
BRJ 6-A6	:47	BGI	B. Bennett	BRUTON MUSIC	ASCAP
PM 10-A8	:04	BGI	P. Kass	PARRY MUSIC	ASCAP
ME 104-2	:33	BGI	Traditional	EMIL ASCHER	ASCAP

An extract from the music clearance log corresponding to the same section of cues detailed on the attached music cue sheet. Listed on this log is the information necessary to ensure that copyright clearance is secured (either blanket or per use) for each identified music track.

familiarity with your entire library will save you the most time.

By now, you should have a good (but not necessarily set) idea of the overall musical style you want to use for the show. Knowing what your library provides in that style (or related styles) means that you can proceed along those lines, and not worry about having to change directions because you couldn't find enough material.

Sometimes the opposite occurs; knowing what types of interesting and appropriate music you have in your library might provide you with an idea for the overall musical style. Also, I often find that once I have started choosing cues, I will come across a dynamite piece of music, and the entire score ends up being built around that particular style and texture of instrumentation. This technique helps to lend a large degree of musical unity to the score—an important feature to strive for if you want your music track to sound as though it were scored.

Musical unity can be obtained as simply as, let's say, just using cues performed by synthesizers throughout the production. Other potentially unifying factors include the ensemble's size; the type of instruments and the predominance of one or more of them; or the style (for example, Seventies industrial, film-style orchestral, etc.).

Additional production factors that will influence your choice of music include:
- the demographics of the audience for whom the production is intended;
- the playback mode and location;
- the purpose of the production;
- the type of sound effects to be used and where;
- the pacing of the picture editing;
- the voice-over timbre.

Add to these the client's tastes and wishes...and you get a total picture.

Speaking of the client, one of the more common requests I receive—usually in relation to "saving" a show—can be summed up as follows: "Use the unexpected...play against the picture...we want our product to be *different*." This approach is valid and has its place. My own general philosophy, however, is to use music which *works* and, which more often than not, is what the audience is *used* to hearing.

The trick is to use the music more splendidly than they may have ever expected it to be. It doesn't matter whether the project you are working on demands dated or current styles; if a piece of music is right for the picture, and the audio quality all the way down the line is superb, it *will* be effective.

Music editing

After deciding which cues to use, it's now time to transfer the music from the master tapes or CDs to the editing medium. I'll restrict my comments to editing on audiotape, because that's the medium I use. The methods described here, however, are equally applicable to cutting on film or electronic editing.

First, preserve the master quality by using noise reduction, or at least by dubbing machine-to-machine without extra electronics in the console. Depending on your final product, by the final mix the music may be as many as seven generations away from the master. Transfer the cues (in the order in which they appear in the show) at 15ips, stereo. Even if a show is going to be presented in mono, there is always the possibility for a future revision in stereo.

There are many different ways to make a music cue fit a specific visual sequence, and generally the easiest but least effective method is to fade it in and out. Unless you luck out, this does not result in a scored sound, which is what this article is about. However, there are indeed occasions when a fade is preferable, such as when transitioning from visuals into talking heads, or when you want to "sneak" the music in just

Major production music libraries

The following list of the major production music libraries is not comprehensive, nor does it review or recommend any particular library. However, we have provided addresses, phone numbers and Rapid Facts card numbers to make it easy for you to contact them. Most libraries will be happy to send you a demo of their music as well as a price list.

Air Craft Music Library
77 N. Washington St.
Boston, MA 02114
800-343-2514
or
25 E. 21st St.
New York, NY 10010
Circle (99) on Rapid Facts Card

Associated Production Music
888 Seventh Ave., 12th Floor
New York, NY 10106
212-977-5680
Circle (100) on Rapid Facts Card
or
6255 Sunset Blvd., *724
Hollywood, CA 90028
213-461-3211
Circle (101) on Rapid Facts Card

Capitol Production Music
1750 N. Vine St.
Hollywood, CA 90028
213-461-2701
Circle (102) on Rapid Facts Card

Emil Ascher, Inc.
630 Fifth Ave., Suite 660
New York, NY 10111-0071
212-581-4504
Circle (103) on Rapid Facts Card

Regent Recorded Music
7060 Hollywood Blvd. Suite 800
Hollywood, CA 90028
213-461-9926
Circle (104) on Rapid Facts Card

DeWolfe Music Library
25 W. 45th St.
New York, NY 10036
212-382-0220
Circle (105) on Rapid Facts Card

First Com
13747 Montfort Drive, Suite 220
Dallas, TX 75240
214-934-2222
Circle (106) on Rapid Facts Card

Network Production Music
4429 Morena Blvd.
San Diego, CA 92117
714-272-2011
Circle (107) on Rapid Facts Card

Omnimusic
52 Main St.
Port Washington, NY 11050
516-883-0121
Circle (108) on Rapid Facts Card

Soper Sound Library
Box 498
Palo Alto, CA 94301
Circle (109) on Rapid Facts Card

Thomas J. Valentino Inc.
151 W. 46th St.
New York, NY 10036
800-223-6278
Circle (110) on Rapid Facts Card

before the end of a section. The remainder of this article, however, will concentrate on editing techniques that provide that "tailored" sound.

Scored music cues have definite beginnings and endings, and follow the action and pacing. In order to achieve the same effect with library music, you need to first audition the cue thoroughly and choose those portions that will best fit the visuals. While you are auditioning the cues, try to get some rough timings of those portions, compare those with the total time you need and listen to whether or not the cue modulates (changes keys) within the piece. You need to take note of modulation because either the actual ending needs to be in the same key as the beginning—or you'll need to edit the cue so that it contains the full turnaround from one key to another.

If you don't have perfect pitch, you can save yourself some time by using a pitch pipe. Your reference point should be tonic or "do" (first note of scale) for that particular key. Very often, this is the bottom note of the chord found at the very end of the cue. A cue can also begin this way but, quite often, it does not get to tonic for several measures. The good news is that composers who write for music libraries know about editing, and their music is usually written to stay in the same key, mode, time signature and tempo throughout. If anything does diverge, it's usually within the cue, and you'll find that it will return to the original before the ending.

Once you've decided which portions of the cue to use, now comes the time to decide how to best edit it. If you have no internal picture points to catch, or specific musical portions you want to use, it may just be a matter of finding an opening piece of music and then chopping on the ending. Because many composers make the introductory (read: "repetitive") portion of their piece longer than need be, you will probably want to start by trimming the intro. Then, using either your tape machine's timer or a stop watch, begin timing from the point at which you want the cue to begin, and continue until you reach the total required time (Point A).

Take note of where you are musically at this point, then fast forward to the end of the cue and look for a musical point somewhere around the ending that would allow you to join it to Point A. Never mind if this will make the piece longer than the time you need. The object is to discard all the material in between the two points, and then trim out bits and pieces to achieve an exact time for the segment.

This method may or may not work, however, depending on the point you have reached in the music when it comes time to tack on the ending. If you are in the middle of the bridge or B section, or a modulation, proceed to the next editing method.

This method is useful when you wish to include several specific parts of a cue, as follows: Go through the cue and cut all of those parts together, making sure that they work musically and that the total time is longer than the timing you want to end up with. Then work your way through this new version and delete pieces to achieve the correct length. As a general rule, when it gets down to the stage of having to eliminate seconds or fractions of seconds, those pieces can often be found either at the end of the piece, or within linking sections.

If there are several internal cue points to catch musically, you will need to do a little pre-planning regarding what parts of the music you want to hit at certain parts of picture. Then you may literally have to rearrange the cue, transplanting a part of the cue from back to front, or vice-versa. You could also tackle this problem with a technique known as assemble editing, which is used most often by editors cutting music on film.

To assemble edit, you simply run the cue up to the first timing where you want something to happen musically. You then fast forward the cue until you reach a musical event that would be appropriate for this punctuation, find a musical way to join the two points and then splice them together. In order to achieve the correct timing for the "hit" to occur, you may have to shave a bit from the part of the cue preceding the edit. You may also have to transfer the cue twice to provide enough material to finish the cue.

Editing tips

When you first begin trying the editing methods described here, you will probably run across as many problems as there are types of music; your solutions must *always* be determined by your ears. Being overly critical is always preferable to hoping, for example, that someone in the audience accidentally coughs at precisely the same point as your bad edit.

Here are a few general tips that I can pass on to you:

• It will soon become apparent that some textures are far more difficult to edit than others: for example, long, held chords that occur behind moving melodic lines; blocks of opaque synthesizer chords; and large orchestral textures with lots of reverb. The best way to learn how to cut your way out of these textures is to experiment. You'll gradually find out what does (and doesn't) work, both from the aspects of musical form and physically cutting the tape.

Whittle away at edits, saving the pieces of course, until you achieve the desired result. If you cut the tape precisely and don't wrinkle or stretch it, you can reinsert even very small pieces and fix your edit. If all else fails, you can always retransfer the piece.

• If you must transfer your music from vinyl albums, remember that you can remove clicks and pops via editing, depending on where they occur in the music and how severe they are. Just make sure your razor blade and grease pencil are always sharp, and remove miniscule sections of tape one at a time.

• When combining two or more cues—either one after the other, or intercut to form a single cue—always be aware of their key relationships. There are certain keys that are compatible with one another, and keys that clash when heard adjacent to one another. Even when you choose cues in the same or compatible keys, because of minute speed differences during the recording or mastering process, the keys may be off just enough to cause a clash. When this happens, use the varispeed control on your turntable or tape recorder. Just remember to make the speed corrections when you *transfer* the cue; if you use varispeed while editing, the real-time counter on your tape machine will not be reflecting the true total elapsed time.

• Varispeed can be used in a number of other helpful ways. Sometimes you come across a cue that just "won't edit" and, try as you may, you are still ½-second long or short. If there is no adjacent key relationship to worry about, you can use varispeed to correct the length, always listening to make sure the instrumental or vocal timbres are not being destroyed. (A flute should sound like a flute, and the Pointer Sisters should not sound like the Chipmunks!) If there are key relationships to consider, you can start in varispeed mode at real time, and then gradually alter the speed.

• If you have access to a multitrack, you can sometimes solve certain editing problems by cross-fading between tracks from one portion of the cue to another. (This technique is particularly useful when working with difficult sound textures mentioned above.)

And, finally, here are a few tips on the mechanical process of editing: use a splicing block that will give you the most oblique angle for your cut; always make sure your block and razor blade are demagnetized; and use only enough splicing tape to cover your splice with a bit to spare.

Although I do not have sufficient space here to cover every aspect of music selection, editing and theory, I hope that some of the pointers offered will be of some help. I especially hope that the article has given a bit more credence to designing with library music, and that you will try it with your next project. R·E/P

All photography by Elizabeth J. Annas.

Public Relations and Advertising Principles for Recording Studios

By Bobbi Marcus

A well-developed public relations and advertising plan can establish a recording facilities image and help maintain the necessary edge over its competition.

The pro-audio market is a highly respected one, and it often can be difficult to establish and maintain a presence within the industry. No matter what the size of your facility, a basic knowledge of public relations and advertising can be beneficial.

Although public relations and advertising are related, you shouldn't consider them to be the same method. There are some important distinctions.

Advertising uses paid media space, and you determine exactly how, when and where your message appears. It's designed to communicate product or service values to your customers in terms of need, desire, quality or service.

Space can be purchased in the form of display and classified ads in the print media and air time in the broadcast media.

Because you pay for advertising, you can determine how successful it is by comparing how many customers or products you sell after running an ad or series of ads.

Public relations, on the other hand, communicates information, impressions and ideas about your studio. It attempts to give your customers and potential customers a favorable response when they think of your studio. This can be achieved through the media (newspapers, trade journals, television and radio). Public relations involves the public's perceptions, so results take longer to achieve and are harder to measure.

If your operation is large enough, you can hire a person or agency to handle your public relations. However, you may not be able to afford one or can't justify paying someone to do it on your behalf.

The alternative is to produce your own campaign, which you can do in your downtime or off-hours. If you work on your own, there are a number of vital steps to take in preparing a public relations campaign.

1. Target whom you want to reach.

The first thing to do is identify who you want to reach with your message. Who is going to be interested in your studio? Who is likely to use the equipment and services you provide?

2. Research the appropriate magazines.

Obtain copies of every trade and consumer publication that applies to your operation. This includes technical trade publications as well as musician-oriented titles. Don't forget publications that serve peripheral industries that you are in. If you offer video post-production services, for example, include the video trades.

3. Analyze the content.

Read the magazines and determine their editorial formats. Do they list cur-

Bobbi Marcus is president of Bobbi Marcus Public Relations, a Los Angeles-based company whose clientele includes Thomas Dolby, Herbie Hancock, Jan Hammer, Universal Recording and Fairlight Instruments.

rent sessions at studios across the country? Do they run items on new personnel? Are there in-depth articles about a single studio? Are these written by staff members or free-lancers? The type of material a magazine runs and in what form will determine what you submit.

4. Obtain media kits.

Available from most magazines, media kits usually include editorial calendars and advertising schedules, as well as information on the audience a magazine services. Although kits usually are geared to advertisers, the editorial and audience information will be useful.

5. Obtain any additional information.

Some magazines have written guidelines that explain how you should submit material. Obtaining these can help you a great deal. Also check and see which staff member should receive the material. Magazines receive a great deal of mail, and getting your information to the right person will help the chances of it appearing.

Once you've finished your research and determined whom you should send material to, the next step is to develop material and send it to the appropriate magazine.

As mentioned, advertising has a separate role from public relations. Because you have control over advertising, you can present your message the way you want it. Your goal is a creative and exciting ad. To achieve this, hire a tremendous art director and/or copywriter to design an ad.

Basically, I don't think that it is necessary for a studio to hire an ad agency. You probably won't be advertising on radio, television or billboards. Instead, you'll be concentrating on a few publications. With the assistance of a media kit, you can place ads yourself.

Self-promotion for production facilities: What constitutes news?

By Dave Stringer

You've just added a MIDI room or hired a new engineer, and you want to put the word out. How do you prepare a news release?

Publications are always looking for news. Many things about your studio or production facility may be newsworthy. Before you write anything, first think about what you do.

Is your studio the only one around offering a certain service? Is it the biggest, the smallest, the busiest, the most diverse? Does it have some newfangled equipment?

Who are your clients? Have any major bands or acts recorded a recent hit at your facility? Have any commercials produced in your studio won any awards? Do you have any unusual clients?

When you determine that you have a story with a good hook, write a short and simple press release on your studio's letterhead. Include the who, what, where, when, why and how of the topic. Keep it short: two pages maximum, double-spaced. One page is better.

Assuming you've researched the appropriate publications that you want to send the release to (see main story), and have prepared the release, mail it in after checking to see that you're sending it to the right person. Don't address it to "the editor."

Sometimes, a magazine will want to use photographs. Depending on your topic, send in a photograph along with the release. However, publications have different standards about what they will use, and it's best to check.

Some publications may have a policy against running a "mug" shot of the engineer you've hired, for example. If you do provide photos, submit black-and-white prints or color slides. Some magazines may not accept color prints, although some might. Do not send instant photos. Generally, they

Remember to provide captions detailing equipment and indentifying all of the people in the photo. Type these on white self-adhesive labels and attach them to the photo, or write them on a separate piece of paper and fold it around the photo. Whatever you do, do not write on the back of the photo.

Depending on the type of release, you can call and see if the editor received it. For more in-depth releases, calling would be fine; for equipment and personnel announcements, it may not be. It's better to wait about three months after you send in the release. If it hasn't yet appeared, you can check on it or send in another one.

When you do receive coverage in newspapers and technical magazines, obtain additional copies or have reprints made to include in your sales literature.

The key to creating a good release is to be constantly aware of the news potential of your business. Look at your local TV news, newspapers and trade publications more critically. Analyze the content of these stories to see how the products or services that you provide can fit in.

And when you have a legitimate story concerning your facility, get the word out.

Dave Stringer is a Kansas City, KS-based public relations consultant whose clients include computer and software companies, the Marlboro Country Music Tour and The Sound Factory studio.

Specialized marketing involves an appreciation of the concept of "positioning"

By Howard Geltzer

An audio production studio is a special business requiring a specialized expertise. To customers of this business, however, a studio is simply a supplier of service. To stand out in the customer's mind among all the other recording service suppliers, a studio has to communicate its ***individual*** *niche.*

Each studio has its own set of characteristics that is different from the characteristics of any other studio. Properly handled, these qualities can be parlayed into a perceived advantage. The most important consideration is not necessarily whether you have digital multitrack capabilities, or the largest suite in town. What matters is how your studio is perceived by your prospects. You have to establish your position in the marketplace. The strategy of positioning entails pinpointing that distinctive niche, and making sure prospects know what it is.

To initiate this strategy, a studio must find out what is in the minds of potential clients, what are their preconceptions about its facilities and its competition. What is its current position? Should it be changed or should it be reinforced?

A studio must find a position in the collective client's mind that is not already occupied and that fits the studio's capabilities. Basic research of a cross section of your prime prospects is essential. Don't forget key magazine and newspaper editors in your area. They can be a great source of information and they will play a vital role in your marketing campaign once it is underway. And don't hesitate to ask about the competition. Their weakness may provide the basis for the position that creates business opportunities.

After the position is defined, a studio has a clear focus and a precise point of view that enables it to attain more successful publicity and more effective public relations. It is crucial that public relations be coordinated with all other forms of marketing. The press releases and the interviews arranged with the technical press should complement brochures and advertising, and all elements must be directed at furthering the position.

In the final analysis, what a studio needs is not more publicity, but better, ***directed*** *publicity, the kind that relates to clearly defined marketing objectives and furthers the facility's unique position in prospect's minds.*

Howard Geltzer is president of Geltzer & Company, a New York-based public relations agency that represents Sony Corporation of America.

Below is a fictitious example of how to construct a press release.

FOR IMMEDIATE RELEASE:
September 5, 1987

CONTACT:

Michael Sullivan
Southward & Associates
9209 West Chestnut St.
Burbank, CA 99506
818-843-1111

Greenwood Studios expands facility

Greenwood Studios, Hollywood, CA, has reopened its facility with expanded control room capabilities and enlarged studio space.

The control room features electronic keyboard interface and reverbs including Lexicon 224, Yamaha REV-7 and SPX-90 units.

Also included is a Studer A-820 2-track deck.

If you decide to place a display ad, be careful of where you place it. An ad in one publication may be appropriate because of the audience it serves, while it won't be appropriate (and will be a waste of time and money) in another. Be sure that you're getting the best value for your money.

Advertising that's tied to editorial coverage rarely, if ever, creates a good impression. I generally don't recommend placing an ad in the same issue in which there will be a story about your studio. In the long run, you'll have more impact if ads are staggered.

Other opportunities

There are other avenues that you should explore. For example, don't forget industry trade shows, where studio personnel often participate in panel discussions and lectures.

Regional or local events are also important. Your studio can host a regional SPARS meeting, host a record company listening party or unveil a new piece of equipment. It's a good way to meet potential new clients and keep in touch with current ones.

Don't overlook the consumer media. Magazine-style TV shows and local news programs and newspapers are interested in entertainment topics. Although your focus will be with trade media, consumer coverage can add some nice icing to your cake.

Call local TV stations and donate free use of your studio or facility for them to tape special music-type interviews during your downtime. Make sure that your logo is visible in the background, and that the station or production company gives you a suitable on-screen credit line.

Some advertising agencies donate their services to non-profit groups to develop campaigns. Call around to local agencies, also offering to donate your services for their campaigns. But be selective and keep these to a minimum.

Make your expertise available to newspaper editors or feature writers on audio-related topics. For industry issues that can cross over into consumer media coverage, such as R-DAT, you can make yourself available. Editors often try to localize a national topic, and you can be a valuable resource if editors know you're available.

Exploring options

Remember to always think of every possible option when thinking about promoting your studio. There are vast numbers of resources available, and with careful planning, you can go a long way in establishing your studio's reputation.

R·E/P

You're in good company when you advertise in Recording ENGINEER/PRODUCER

(Here are just a few of the companies that advertise in Recording Engineer/Producer):

A&R RECORD & TAPE MFG. CO.
A/T SHARFF RENTALS
ACOUSTIC SCIENCES
ACOUSTIC SPACES
ADVANCED MUSIC SYSTEMS
AEG AKTIENGESELL SCHAFT
AEG CORP.
AGFA-GEVAERT
AKG ACOUSTIC, INC.
AKIA ELECTRONICS
ALESIS CORP.
ALLEN & HEATH BERNELL
ALPHA AUDIO AUTOMATION SYSTEMS
ALSHIRE INTERNATIONAL, INC.
AMEK CONSOLES INC.
AMPEX CORP.
AMTEL SYSTEMS
ANCHOR AUDIO, INC.
ANT TELECOMMUNICATIONS
APHEX SYSTEMS LTD.
API AUDIO PRODUCTS, INC.
APOGEE ELECTRONICS CORP.
APPLIED RESEARCH & TECHOLOGY
APRS
ASHLY AUDIO INC.
ATLAS/SOUNDOLIER
AUDIO ACCESSORIES
AUDIO AFFECTS
AUDIO DIGITAL INC.
AUDIO ENGINEERING ASSOCIATES
AUDIO ENGINEERING SOCIETY
AUDIO KINETICS
AUDIO MEDIA RESEARCH
AUDIO PRECISION
AUDIO-TECHNICA, INC.
AUDIORENT
AUDITRONICS, INC.
AURATONE CORP.
AXE ARTISTS X-PONENT ENGINEERING
B&B SYSTEMS
BARCUS BERRY ELECTRONICS, INC.
BASF
BENCHMARK MEDIA SYSTEMS
BERKLEE COLLEGE OF MUSIC
BERTECH ORGANIZATION
BEYER DYNAMIC, INC.
BI-TRONICS, INC.
BLACK AUDIO DEVICES
BRUEL & KJAER INSTRUMENTS INC.
BRYSTON/VERMONT
BURLINGTON AUDIO/VIDEO TAPE INC.
CAE/LITTLITE
CANARE CABLE INC.
CAPITAL MAGNETICS
CARVER
CASSETTE TECHNOLOGIES
CENTRO CORP.
CETEC GAUSS
CHIPS DAVIS, LEDE DESIGNS
CIPHER DIGITAL INC.
CIRCUIT RESEARCH LABS INC.
CMS DIGITAL
CMX CORP.
COMMUNITY LIGHT & SOUND
CONNECTRONICS
COUNTRYMAN ASSOCIATES
CREST AUDIO
CROWN INTERNATIONAL
CSE AUDIO
DAN ALEXANDER AUDIO
DBX
DENECKE INC.
DIGITAL DISPATCH
DIGITAL DUPLICATORS
DIGITAL ENTERTAINMENT CORP.
DIMENSION MUSIC LIBRARY
DOD ELECTRONICS
DOLBY
DORROUGH ELECTRONICS, INC.
E-MU SYSTEMS
EASTERN ACOUSTIC WORKS
EASTERN STANDARD PRODUCTIONS
EDITEL
EDITRON
EDUCATIONAL ELECTRONICS CORP.
ELECTRIC VALVE COMMUNICATION CORP.
ELECTRO-VOICE, INC.
EMBASSY CASSETTES
ENSONIQ CORP.
EUGENE ARTS FOUNDATION
EUROPADISK, LTD.
EVA-TONE INC.
EVENTIDE, INC.
EVERTZ MICROSYSTEMS
EVERYTHING AUDIO
FAIRLIGHT INSTRUMENTS
FENDER MUSICAL INSTRUMENTS
FM-TUBE CRAFT SUPPORT SYSTEMS
FOSTEX CORP. OF AMERICA
FOUR DESIGNS CO.
FRANKFORD/WAYNE MASTERING LABS
FULL COMPASS SYSTEMS
FURMAN SOUND
FUTURE DISC SYSTEMS
GARFIELD ELECTRONICS
GENTNER
GIBSON ACCESSORIES
GMI DISTRIBUTORS
GOLDLINE
GOTHAM AUDIO CORP.
GRD
HARDY CO.
HARRIS AUDIO SYSTEMS, INC.
HARRISON SYSTEMS, INC.
HEID PRODUCTIONS
HILL AUDIO INC.
HY JAMES
IAN COMMUNICATIONS GROUP, INC.
IBANEZ
INTEGRATED INNOVATIONS
INTERSONICS
INTERNATIONAL TAPETRONICS CORP.
JBL PROFESSIONAL
JENSEN TRANSFORMERS
J.L. COOPER ELECTRONICS
JORDAX CALIFORNIA, INC.
JOSEPH F. RASCOFF & CO.
JRF MAGNETIC SCIENCES, INC.
JUICE GOOSE-WHITENTON INDUSTRIES
JVC COMPANY OF AMERICA
K DISC MASTERING
KAHLER DIV. OF APM
KARLBERG ENTERPRISES
KCC AUDIO/VIDEO
KENNETH A. BACON ASSOCIATES
KLARK-TEKNIK ELECTRONICS INC.
KORG USA
KURZWEIL MUSIC SYSTEMS INC.
LA SALLE MUSIC
LAKE SYSTEMS
LARRY QUINN
LD SYSTEMS, INC.
LEITCH VIDEO OF AMERICA INC.
LENCO
LEXICON, INC.
LINEAR & DIGITAL SYSTEMS
LT SOUND
MAD HATTER RECORDING STUDIOS
MAGNEFAX INTERNATIONAL, INC.
MAGNETIC REFERENCE LABORATORY, INC.
MANNY'S MUSIC
MARSHALL ELECTRONICS INC.
MARTIN AUDIO VIDEO CORP.
MASTERING LAB
MEMPHIS SOUND PRODUCTIONS
MEYER SOUND LABS
MICRO AUDIO
MICRO MUSIC
MILAB
MONSTER CABLE
MUSIC WORKS
MUSICALLY INTELLIGENT DEVICES INC.
NADY SYSTEMS
NAGRA MAGNETIC RECORDERS, INC.
NAKAMICHI USA CORP.
NEOTEK CORP.
NEPTUNE ELECTRONICS, INC.
NEVE INC.
NEW ENGLAND DIGITAL
NIKKO AUDIO
OCEAN AUDIO INC.
OMNI-CRAFT INC.
OPAMP LABS
ORBAN ASSOCIATES INC.
OTARI CORP.
PANASONIC
PHYLCO AUDIO
PIERCE ARROW
POLYLINE CORP.
PRISTINE SYSTEMS, INC.
PRO SOUND
PROFESSIONAL AUDIO SERVICES
QSC AUDIO PRODUCTS
QUAD RECORDING STUDIOS
RANE CORP.
RCA TEST TAPES
REMOTE RECORDING SERVICES, INC.
RENKUS-HEINZ, INC.
ROCKTRON CORP.
ROLAND CORP.
ROYAL RECORDERS
RPG DIFFUSER SYSTEMS
RTS SYSTEMS
SAKI MAGNETICS, INC.
SAMSON
SANKEN MICROPHONE CO.
SCHOEPS GMBH
SCV AUDIO
SENNHEISER ELECTRONIC CORP.
SHURE BROTHERS INC.
SIMON SYSTEMS
SKOTEL CORP.
SKYELABS, INC.
SOLID STATE LOGIC
SONY BROADCAST PRODUCTS CO.
SONY CORP. OF AMERICA
SONY PROFESSIONAL AUDIO PRODUCTS
SOUND & VISION
SOUND PRODUCTIONS INC.
SOUND WORKSHOP PROFESSIONAL AUDIO PRODUCTS, INC.
SOUNDCASTLE STUDIO CENTER
SOUNDCRAFT USA
SOUNDER ELECTRONICS
SOUNDMASTER INTERNATIONAL
SOUNDTRACS, INC.
SPECTRA SONICS
SPECTRUM MAGNETICS
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Recording ENGINEER/PRODUCER

The technical journal for audio professionals

P.O. Box 12901, Overland Park, KS 66212, 913-888-4664

News

Continued from page 8

Audio LA formed by Ike Benoun

The new Los Angeles-based pro-audio equipment supplier will specialize in systems design, installations, service and sales. Benoun has a background of 27 years in the audio industry, having formerly managed Audio Industries and Walt Davis Enterprises (professional video sales).

His affiliation with a local dealer, West L.A. Music, will provide a complete selection of keyboards, computer-related products and musical instruments, he says, with extensive expertise in the design and operation of MIDI systems.

Audio L.A. is the newly appointed sales and service organization in the western U.S. for the Tascam ATR-80 series analog 24-track. For more information, call Benoun at: 213-477-1516.

Stolen equipment

On June 23, while parked at the Philadelphia Academy of Music, the following equipment was stolen from the Effanel Recording truck:

- Two B&K model 4007 microphones (serial numbers 973232 and 1040002) in KE0215 wooden boxes with clips.
- A Beyer M500 ribbon microphone (21465) in box with clip.
- Two Sennheiser MD-421 microphones (19817 and 19825) in one box, initials JA inscribed on bottom; one exterior repaired.
- Two Sony ECM-50 microphones (22458 and 22459) in individual boxes with complete clips (initials JA on preamps).
- A Neuman KMR-82 shotgun mic; gray with black leather sheathe and windscreen with notch cutout for Rycote windshield mounting.
- Four Radio Shack PZMs, two with XLRs on cable end; two new in boxes.
- Six AKG and Shure A27M stereo bars.

All equipment was packed in a medium suitcase-size case with silver metal exterior and light blue felt interior.

A report was filed with the 9th District of the Philadelphia Police (215-686-3090) on June 24.

A reward of $500 for information leading to the return of the equipment is being offered. Call either Effanel Recording 212-807-1100, or James Anderson Audio at 718-643-1675.

Valley People change name to Valley International

According to company president Norman Baker, "We've come a long way since 1977, when Valley People was originally founded. In 1980, the company merged with the highly successful Allison Research, so our heritage goes back to 1969.

"During the last 18 months, our company has upgraded all existing products electronically and mechanically. We also undertook an exhaustive research and development effort which lead to the introduction of nine new products.

News

"In enhancing our existing products and introducing the many new ones, we also desired to incorporate a readily identifiable company logo as a part of our front-panel graphics treatment. The name, Valley International, was chosen as a proper reflection of the worldwide acceptance which our products enjoy."

Otari introduces new product line

The new Otaritech line of products developed independently of Otari will be targeted to the recording and broadcast markets.

The first product in line, available in August, is the TC-50 center channel time code/FM processor. The manufacturer claims that the TC-50 provides an inexpensive method for center-track time code to be added to a tape machine. It will retrofit Otari MX-5050 2-tracks or any other 4-head position tape transports.

Time compensation is provided for three tape speeds, and front-panel LEDs indicate time code level at the input and output.

For further information, contact Otari Corporation, 2 Davis Drive, Belmont, CA 94002; 415-592-8311.

Beatles CD releases mastered exclusively on Ampex digital tape

According to Steve Smith, marketing manager of audio tape products at Ampex Magnetic Tape Division, all re-recordings on both open-reel and U-matic formats were made entirely on Series 467 mastering tape.

The original analog 4-track masters, where available, were dubbed across to a Mitsubishi X-850 32-track on 467 1-inch tape. The multitrack recordings were then mixed to a Sony PCM-1610 and recorded on 60-minute 467 U-matics for CD manufacturing. Remix took place at Air Studio, London, England.

People

At **Tascam,** three executives have been added to the sales force: **Chuck Prada** takes over as the new eastern regional manager; **Bill Stevens** assumes the post of southern regional manager; and **Mick Walker** has been promoted to the professional sales position for Southern California.

Neotek has named **Oliver Masciarotte** as its new production manager. Previously, he served as a staff engineer at Criteria Recording Studios, Miami, and as a lecturer at the University of Miami.

Stan Stitgen has been named corporate vice president of **Yamaha Corporation of America,** where his responsibilities will center on the development of institutional educational programs.

James A. Tipton has been named vice president for sales at **dbx**.

Send news items to Dan Torchia, staff editor, **Recording Engineer/Producer**, Box 12901, Overland Park, KS 66212.

Studio Update

Northeast

WPSX (University Park, PA), Pennsylvania State University station and PBS broadcast affiliate, has take delivery of a **Neve 5114/24 console. The console features full EQ and dynamics on 24 inputs.** *Penn Street University, 27 Wagner Annex, University Park, PA 16802; 814-865-1993.*

Sigma Sound Studios (Philadelphia) has upgraded Studio 1 with a 52-input **Neve 8078** console with six additional effect sends and 84-input mixing capability. The console has also been fitted with a new **George Massenburg Labs** automation system and is capable of decoding mixes from **NECAM 96** and SSL floppies.

Also featured in Studio 1 is a **Mitsubishi X-850** 32-track recorder. *212 N. 12th St., Philadelphia, PA 19107; 215-561-3660.*

National Video Center/Recording Studios (New York) has announced the introduction of **Super Sync,** a film-to-tape service allowing filmmakers to perform synchronous transfers of ¼-inch soundtracks with film formats using center track time code. *460 W. 42nd St., New York, NY 10036; 212-279-2000.*

Evergreen Recording (New York) announces the opening of a new computer/MIDI programming room. The facility features a **Macintosh Plus, IBM** AT/hard disk, **Roland MSQ 700** and **MSQ 100.**

Other hardware includes a **Soundcraft 600 series** console, **Crown DC150A** power amp, **Yamaha NS10Ms, Otari 5050 mkIII** 4-track and a **Yamaha SPX-90.** *215 W. 91st St., New York, NY 10024; 212-362-7840.*

New England Mobile Recording (Stow, MA) has added a **Sony PCM-F1** digital encoder/decoder, **Yamaha SPX-90, Aphex Type "B" Aural Exciter, Yamaha REV-7** and a **Crown DC-300A-II** power amp.

The studio is designed for mobile and in-house recording, and will continue to provide live recording. *P.O. Box 409, Stow, MA 01775; 617-562-2111.*

Presence Studios (East Haven, CT) have installed an **SSL 4000E** 56-input console with **Total Recall, Studer A-820-2TC** ½-inch mastering machine, **Drawmer 1960** tube compressor, **Drawmer gates** and a **Lexicon PCM-60, PCM-70** and a **Yamaha SPX-90.** Keyboard additions include modifications to the **Fairlight CMI Series III** and a **Forte MIDI system** for the Yamaha grand piano.

The company also announces that **Milt Sutton** has joined the staff as staff engineer. *461R Main St., East Haven, CT 06512; 203-467-9038.*

Barry Diament Audio (Riverdale, NY) announces the opening of its new facility for mastering Compact Discs. *2728 Henry Hudson Pkwy., Riverdale, NY 10463; 212-543-2079.*

Master Sound Astoria (Astoria, NY) has added the **Digital Creations** Moving Fader Automation system and has developed an interformat patch bay that will facilitate the transfer of material entirely in the digital domain. The move broadens the studio's capabilities in analog/digital recording and in audio post-production.

Also, David Browning has joined the studio and will head up the studio's post-production division. He was formerly with Regent Sound. *34-12 36th St., Astoria, NY 11106; 718-392-5600.*

Victory (Gladwyne, PA) is a new studio for advertising, corporate and music clients in the Philadelphia area. The studio is an extension of Kajem Recording, which specializes in album production. Victory will offer 24-track music recording, voice-overs, original jingle production and audio-for-video. *1400 Mill Creek Road, Gladwyne, PA 19035; 215-649-3277.*

Audio Works (Boston) has relocated to 284 Mount Auburn St., located in the Watertown district of downtown Boston, due to company growth and increasing demand. An office has also been opened at 252 Newbury St. to keep the studio in touch with Boston ad agencies. *Box 856, Astor Station, Boston, MA 02123; 617-236-0300.*

Southeast

The Bennett House Studios (Franklin, TN) announces that **Gene Eichelberger** has joined the company as general manager and chief engineer. **Liz Jones** has also joined the staff as studio manager. *134 4th Ave. North, Franklin, TN 37064; 615-790-8696.*

API Audio Products (Springfield, VA) has sold its first motorized fader system to Tommy Boy Records in New York. The system features 32 API model 940M faders and Disk Mix. *7951 Twist Lane, Springfield, VA 22153; 703-455-8188.*

Strawberry Jamm and Higher Skys Studios (West Columbia, SC) have merged to become **Strawberry Skys,** a complete 24-track automated and computer assisted facility. The 24-track studio is MIDI-compatible and will be equipped to handle all basic video capability.

Owner **Bob Curlee** will serve as chief engineer and **Gary Bolton** will serve as co-manager of the studio. **Ron Hollins,** graduate of the University of Miami, will serve as staff engineer. *1706 Platt Springs Road, West Columbia, SC; 803-794-9300.*

Ealing Mobile Recording (Chicago) has added a **Neve 5442** 8-input, 2-output console. *709 W. Roscoe, Chicago, IL 60657; 312-871-7793.*

Sonic Arts (Lake Villa, IL) has updated the facility with a **Neotek Elite** console with 36 inputs and a **Klark-Teknik DN 780** digital reverb. *23783 W. Petite Lake Road, Lake Villa, IL 60046; 312-356-8992.*

Star Trax (Orland Park, IL) has taken delivery of a **Neotek Elite** with 48 inputs, **UREI 813C** and **Hafler P500A.** *15602 S. 70th Court, Orland Park, IL 60462; 312-429-2760.*

Mark Five/Sandcastle (Greenville, SC) is the result of a merger between Mark Five and Sandcastle studios. The new studio is owned and operated by **Rich Sandidg** and **Chric Cassels,** Sandcastle, and **Eddie Howard**, Mark Five manager. *10 Michael Drive, Greenville, SC 29610; 803-269-3961.*

Charles Brown Music (Cincinnati) has added the **NED Direct-to-Disk** option to its Synclavier, making it the city's first tapeless studio. The system has been used to complete audio post-scores for advertising work, and a music theme and promo package for a local news show. *1349 E. McMillan Ave., Cincinnati, OH 45206; 513-281-5212.*

Southern California

South Coast Recording Studio (Santa Ana, CA) has recently upgraded its 8-track facilty to 16-track by installing the **Fostex E-16** with the **Ramsa WR-T820** recording console.

An **Alesis XTC** digital reverb was also added to the outboard gear. *1818½ N. Main St., Santa Ana, CA 92706; 714-541-2397.*

Westlake Studios (West Hollywood, CA) has taken delivery of a **Neve V** series for its A room. *8447 Beverly Drive, West Hollywood, CA 90048; 213-654-2155.*

Mad Dog Studio (Venice, CA) has recently renovated and added a **Neve 8108** console and a **Studer A800** 24-track recorder. *1715 Lincoln Blvd., Venice, CA 90291; 213-306-0950.*

Pacific Cassette Laboratories (Torrance, CA) has relocated its offices and studios to 20655 South Western Ave., #116.

The facility duplicates in real time on modified **Nakamichi AX-9** cassette decks using specially selected **TDK** metal tape and **TDK** reference standard housings. The facility can also produce **TDK** metal tape in custom lengths. *P.O. Box 6248, Torrance, CA 90504; 213-618-9267.*

Post Logic (Hollywood) a post-production sweetening facility featuring a **SSL 6000E** fully automated console with Total Recall has recently added a **DBX 160x**, **Drawmer** gates, **Drawmer** tube limiter, **Roland SDE 3000**, **Eventide SP2016**, **Akai S900** and **UREI** filter set.

The facility has also updated the monitor system with **Autsberger** cabinets with **TAD** component speakers. *6363 Sunset Blvd., Suite 830, Hollywood, CA 90028; 213-461-7887.*

Post Group (Hollywood, CA), a post-production facility, has expanded to include the addition of an on-line edit bay, four off-line edit bays, a second film-to-tape transfer suite, a third audio sweetening room and a new motion-control shooting stage.

Audio Sweetening Bay C, the facility's newest sound room, features a **Neotek Elite** 24-track console and a **Betacam** player. *6335 Homewood Ave., Los Angeles, CA 90028; 213-462-2300.*

Studio Update

A & M Recording Studios (Hollywood, CA) has taken delivery of the new **Neve Focusrite** module, which incorporates miniaturization of high-performance transformers. The low source impedance of the output and the floating winding achieve independence of load and grounding.

"This new module allows you to equalize and tailor the sound to any taste, with absolutely musical results," says Paul Sloman, general manager. *1416 N. La Brea, Hollywood, CA 90028; 213-469-2411.*

Paul Sloman, general manager of A&M Recording Studios, Hollywood, CA points out the Focusrite modules delivered by Audio Intervisual Design, the West Coast representatives for Focusrite.

Kren Studios (Hollywood, CA) has updated with a **Mitsubishi/Westar** 44x24x88 with Compumix and an **IBM PC** hard drive floppy disk automation system.

Recorders include a **Mitsubishi X850** 32-track digital recorder, **MCI/JH 16** 24-track, **Ampex ATR 102, Studer A-80** and a **Sony PCM F-1** 2-track.

Monitors feature **Yamaha NS10m, JBL 4311, Auratone** and **Advent.** Mics available include **Neumann tube U-47, U-48, U-67, Sennheiser 421** and **441,** and **Shure SM57** and **545.** *6553 Sunset Blvd., Hollywood, CA 90028; 213-461-5781.*

Take One Recording Studios (Burbank, CA) has taken delivery of a **Mitsubishi Westar** 48-input console with **Compumix PC automation and a Mitsubishi X-850** 32-track tape machine.

The facility has also updated its new room with a **New England Digital Synclavier.** *619B S. Glenwood Place, Burbank, CA 91506; 818-841-8697.*

Northern California

Sensa Studio (Sunnyvale, CA) has been purchased by Doug Hopping and Scott Smith, who will manage the facility under the new name, **The Recording Studio Inc.** New equipment includes an **Amek** matchless board and a **Kawai** grand piano. *1016 Morse Ave., Suite 17, Sunnyvale, CA 94089; 408-734-2438.*

Syncro International Studio (San Anselmo, CA), has recently completed construction on a new 24-track MIDI recording studio. The facility will function as a production facility for producer Satoshi Suzuki's film, industrial, album and video projects. R·E/P

Send studio news to Studio Update, **Recording Engineer/Producer**, Box 12901, Overland Park, KS 66212.

New Products

Crown introduces hand-held mics

The mic line includes the CM-100 P2M omnidirectional, the CM-200 cardioid and the CM-300 Differoid. All three are electret-condenser mics and feature built-in pop filtering for suppressing breath noises. Each mic can be phantom-powered from the console or other remote power source.

Circle (150) on Rapid Facts Card

Akai S9V2.0 system disk

The software addition for the S900 digital sampler features crossfade looping to smooth out amplitude variations across the loop, the company claims.

Also featured is the pre-trigger recording feature, which automatically starts the sampling process 2,000 points before the recording threshold is reached.

Circle (161) on Rapid Facts Card

Gentner Digital Hybrid telephone interface system

The system allows the user to interface a telephone line to audio equipment with the capability of simultaneous 2-way conversation.

By isolating the send and receive sides of a telephone call, the system allows the caller's audio to be heard on overhead speakers while mics are active in the room. In addition, the unit can be programmed to automatically answer and disconnect the line.

Circle (158) on Rapid Facts Card

Roland SS-PC software

Session Saver is a software program for the IBM-PC and compatibles, which organizes and stores system exclusive patch data for all synths, sound modules and rhythm composers.

The software creates session databases that can store data from up to 16 different instruments in to a single disk file. Function keys and menus are used throughout and on-line screens also are included.

Circle (159) on Rapid Facts Card

Soundtracs FME modular mixer

The FME offers modularity of inputs, outputs and groups, in 22 and 20 module mainframe sizes. The mixer may be configured to 4- or 8-track recording and video post-production.

Module types available include mono input, mono input with remote start switch, stereo input including RIAA and line in with remote start. Also featured is monitor input with eight monitor sends, group output with upper and lower monitor sections, monitor outputs and stereo master module.

Metering is provided via 12 LED bargraphs, which provide solo warning and power indicators.

Circle (153) on Rapid Facts Card

New Products

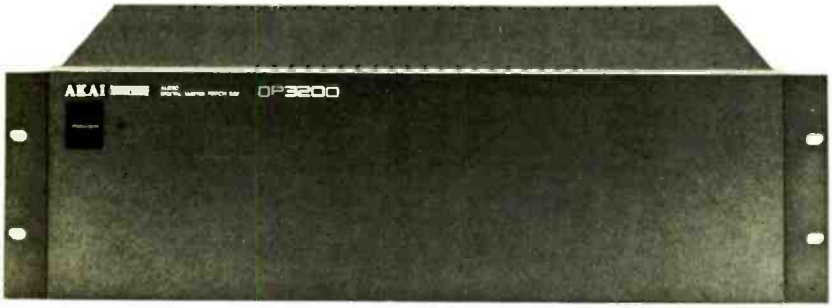
Akai DP Series digital patchbays

The series features automated control with SMPTE lockup to patching systems. Each system is controlled by a computer that automatically routes all signal paths.

The DP3200 and DP2000 feature the PG1000 patchbay programmer and MZ1000 color monitor display. The DP3200 is a 32-input, 32-output system with 64 built-in memory banks for storage of patch patterns.

The DP2000 is a combination audio and video matrix patchbay. Accepting up to 16 balanced audio and 16 video signals, the unit also features 64 memory banks and pattern switching with up to 640 patterns.

Circle (162) on Rapid Facts Card



Audio Precision Bur-Gen signal generation option

Designed for the System One audio test system, the option generates tone bursts, squarewaves, pink noise, white noise and narrowband noise.

The sinewave burst signal is quoted 20Hz to 100kHz and can be selected with any number of cycles, time or duty factor "on," the company claims.

The squarewave frequency range extends from 20Hz to 20kHz. Pink noise is constant amplitude per octave, and narrowband noise is pink noise-filtered by one-third octave band-pass filter turnable and sweepable from 20Hz to 100kHz.

Circle (154) on Rapid Facts Card

AKG C562 BL mic condenser

The C562 boundary layer mic is a prepolarized condenser surface-mount mic for recording situations where unobtrusive mics are needed. The omnidirectional mic features a quoted frequency response of 20Hz to 10,000Hz and impedance at 1,000Hz at 600Ω.

Circle (152) on Rapid Facts Card

New Products

Electro-Voice BK-1632 console

The 16-input, 3-send stereo output console features a pre-fader, a post-fader and an auxiliary-switchable pre-post fade. There is also a provision for sending the reverb to the monitor.

Circle (163) on Rapid Facts Card

Fluke 9100 series digital testers

Designed for microprocessor-based digital circuit boards, the automated series includes two models, including the 9100A for developing test software and the 9100 series emulative board testers.

Built-in functional tests verify the operation of the microprocessor bus, RAM and ROM. A single point probe is also used for isolating faults both on and off the bus, while a system of I/O modules allows simultaneous testing of up to 160 pins at a time, the company claims.

Circle (151) on Rapid Facts Card

Denon DN950F CD cart player

The DN950F plays Compact Discs mounted in special plastic cartridges to protect discs from scuffing, dust and dirt.

The only contact between the discs and the cartridge is at the disc's center hole. A shutter on the signal side of the cart automatically opens upon insertion into the player and closes upon eject. The label side of the CD cart is translucent to read the disc's contents.

Circle (157) on Rapid Facts Card

Tascam Porta 05 mini studio

The new MIDI-compatible 4-track cassette recorder features the company's proprietary head technology. The unit features a 4-channel mixer with full EQ effects send and pan functions and built-in dbx noise reduction.

The mixer is a dual function 24-channel mixer, designed for those who need a single board for both recording and sound reinforcement.

Circle (155) on Rapid Facts Card

Soundtracs CMS3 MIDI/SMPTE interface generator

The CMS3 features a MIDI clock generation in sync with either internal or external time code sources and the generating of SMPTE/EBU time code from a video signal.

The interface enables the update of CM4400/CMS2 and CP6800 console automation to include the control of external MIDI equipment.

Circle (156) on Rapid Facts Card R·E/P

Classified

Advertising rates in Classified Section are:

$1.00 per word per insertion. Initials and abbreviations count as full words. "Blind" ads $25.00 additional. Minimum classified charge $20.00. Classified is also available at $112 per inch. Order must be accompanied by payment to ensure publication.

Classified columns are not open to advertising of any products regularly produced by manufacturers unless used and no longer owned by the manufacturer or distributor.

Classified Advertising should be sent to RE/P, Advertising Department, 9221 Quivira Rd., Shawnee Mission, KS 66215.

HELP WANTED

MAINTENANCE ENGINEER WANTED: For leading pro. recording studio in Los Angeles. Working knowledge of video and tape synchronizers. Address resume to Chief Engineer, 4403 West Magnolia Blvd., Burbank, Ca. 91505—ATTENTION RICK OR MARC OR TEL. 818-841-6807, 213-849-7399. 8-87-1t

WANTED: COMPONENT LEVEL TECH for state of the art analog/digital complex. Send resume to Personnel, 4620 West Magnolia Blvd., Burbank, CA 91505. 8-87-1t

FOR SALE

FOR SALE: 2-Amek 2500 w/Mastermix automation, $70,000; Yamaha GSI keyboard, $5,000, Q-locks fully loaded, $3,000; DX7 II (new), $1,850; Aura Systems (studio monitors), make offer; MKB 2300 Roland keyboard controller, $600; MTR 90 MK II 24 TR w/remote, $32,000; Synclavier Systems full blown, make offer. 818/980-4840. 8-87-1t

FOR SALE: Trident 75, Tascam MS-16, Adams-Smith 2600, Urei 809's, JBL 6132A's, 2 Sharp XCA1U cameras, 2 Sony BVU110's, JVC/Sony. VE-93 Edit., Mics, Tripods, Lights, etc. Everything like new. 212-475-2697. 8-87-1t

MISCELLANEOUS

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FREE CATALOG: Institute of Audio-Video Engineering, 1831 Hyperion, (RP), Hollywood, CA 90027. Recording School. (800) 551-8877 or (213) 666-2380. Approved for International Students. Employers: Call for Employees/Interns. 3-87-6t

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