September 1988

Recording ENGINEER/PRODUCER

The Applications Magazine for Audio Professionals

$4.00

AN INTERTEC PUBLICATION

Audio for Video and Film

Introducing the DAT recorder a professional can't afford to ignore.

The future of audio lies in the all-digital facility. And Panasonic's new SV-3500 DAT recorder/player answers the need for a fully-professional digital audio tape machine.

The SV-3500 is a full-function studio DAT machine from a company committed to servicing and supporting the demanding needs of audio professionals. It features: □ Dual 18-bit ADCs for encoding, and twin 18-bit DACs per channel for playback. □ Industry standard sampling and replay frequencies. □ Interface for digital-to-digital transfers. □ XLR inputs and outputs. □ Switchable +4/−10dB output level. □ Full-feature wired remote. □ 200-times high speed search mode. □ Comprehensive programming and loop functions. □ High-precision peak-reading meters. □ Erasure and re-recording of Start and Skip IDs.

Drop by a Panasonic dealer and ask to see the SV-3500. Because without a DAT recorder with these features, you never know who might ignore *you*. For the name of your nearest dealer, call 714-895-7278. Or, write to Panasonic AVSG, 6550 Katella Avenue, Cypress, CA 90630.

Panasonic
Professional Audio Systems

Circle (4) on Rapid Facts Card

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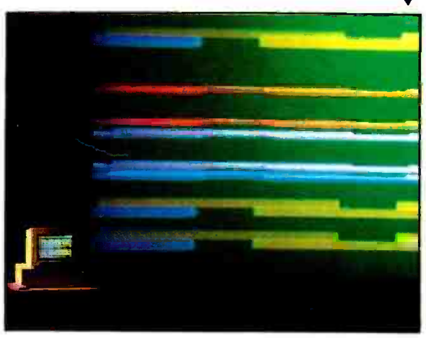

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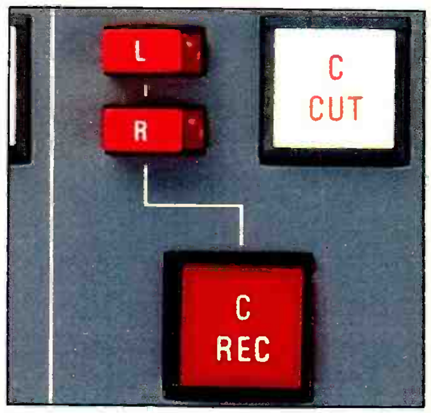

▲On the Cover
The needs of film recording often require tailoring an existing console design to fit this specific application. Photo courtesy of Neve, Bethel, CT.

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NEVE

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Applied imagination. The perfect synthesis between artistry and sound technology. That's the dream that built the Neve range of consoles. From the Classic 80 Series to the V Series and Digital Studio Systems, Neve provides a full range of multitrack consoles.

The Formant Spectrum Equalizer, a technological breakthrough integral to the acclaimed sound of Neve, is now teamed up with the most comprehensive in-console dynamics units ever developed. By combining total control of limiting and compression and comprehensive noise gate with hysteresis control, processing has never been easier on the V Series and Digital Studio Systems. More than a decade ago, Neve introduced moving fader automation. Now in hundreds of installations worldwide, Necam 96 drives the finest production and post-production projects in the world's top studios.

Contact us to make Neve's sound quality, reliability, consistency of performance, speed and ease of operation your tools for translating ideas into recorded reality.

NEVE. . .PUT YOUR CREATIVITY IN FLIGHT.

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GUEST EDITORIAL

By Bob Predovich

The ES Bus: Is It Adequate?

March 1988 heralded the arrival of the long awaited "ATR Dialect" for the ES Bus, the language standard set out by the EBU (E) and SMPTE (S) for communication with audio transports via the previously agreed upon RS-422 serial bus architecture.

With it came a flurry of activity from manufacturers announcing that they had, or would have soon, equipment that would use the new standard, *or a derivation of it.* Considering the number of years that had gone into the development of the *standard,* professional observers might be slightly confused as to why, as soon as it was announced, some manufacturers immediately felt it necessary to modify or augment it for practical industry use.

For years, the industry, especially the broadcast sector, has longed for a common form of communication among equipment in its physical plants. The problems that are inherent in trying to integrate brands A to Z smoothly into an efficient team are horrendous. There are just so many different means to a similar end that trying to achieve harmony among disparate designs, each with its own merit, ends up with the whole being less than the sum of its parts. Enter two of the industry's governing bodies to develop a standard to help solve the problem. High on the wish lists was a common protocol, an electronic "language" that all relevant devices would speak and understand. An easily implemented electronic path was also needed to put devices in touch with each other.

A framework resulted that would embrace the latter by using "serial" data, via the RS-422 38.4Kbaud standard, as the communications link, and new "dialects," or tailored languages, that would be developed to exchange the information. RS-422 was chosen because of the long-distance runs it could handle, the small, 9-wire cable that was needed, the ability to run a single cable to a studio to daisy-chain the equipment (rather than the need for an individual cable from each control device to every controlled device), and the speed of its communication, which is twice as fast as the more-common RS-232.

However, the choice of a serial communications format over parallel was clearly one of ease and convenience, not speed, as the following analogy will demonstrate. Think of rush-hour traffic in a big city. Compare the time it takes for traffic to reach a common destination when moving in a single lane (serial), with the same number of vehicles traveling along 12 lanes or more (parallel).

Because of the bottlenecks and "accidents" that can result in serial communication, "traffic signals" in the form of interrupts and time sharing have to be installed to control access to the "road." Devices wait their turn to get onto the highway (data bus). And although some parallel systems employ these traffic signals too, there are current designs that don't require interrupts or time sharing. Devices get on the bus at will.

> ***The choice of a serial communications format over parallel was clearly one of convenience.***

To get around the RS-422 bottlenecks and time delays, for the most part, designers have distributed the intelligence of serial systems so that less information needs to be sent down the line, keeping the highway less congested. A "coded" transmission triggers an intelligent device at the receiving end to implement a pre-programmed instruction set. Time-sensitive programmed "events" are sent earlier than needed to this intelligent unit, which then executes the event, on time, on its own, at the required moment.

But it is at this point that electronic audio post-production entered an arena that had been primarily dominated by broadcast and video concerns. The needs are quite different. Whereas the broadcast/video industry is dealing with increments of 1/30 of a second (1/25 in Europe), the audio industry insists on 100 times more accuracy (1/3,000 of a second). Unlike videotape equipment that is released to reference its speed to video sync (after it's initially synchronized), analog audiotape recorders must be *continuously* controlled externally, and to a much higher resolution. Further, in the 1990s, electronic time code systems will be asked to replace, and significantly enhance the current capabilities of, decades-old sprocket technology. This is no mean feat. Therefore, is a standard that was designed to be a very elegant "remote control" for the broadcast industry, and that can be quite readily adapted to video post-production, adequate to the needs and sophistication required in audio post-production? The answer seems to be before us, and lies in the *derivations* of the *standard* that are now being implemented.

Paraphrasing one manufacturer's literature: "Although the ES Bus delays are minimal, we felt it important to implement a separate Crash Record Bus." Explanation: The music industry, for example, is used to manually "punching into record" rhythmically. This is not a pre-programmed event, so an instantaneous response speed is required at the remote control.

If the ES Bus standard alone is used, a possible delay will result. Depending on the number of devices daisy-chained on the bus at the time, the delay could be inconsistent (randomly variable, caused by the traffic jams we spoke of earlier). In fact, as more devices are added to the bus, all wishing to be able to communicate on it, this cumulative congestion can cause problems in other time-sensitive areas.

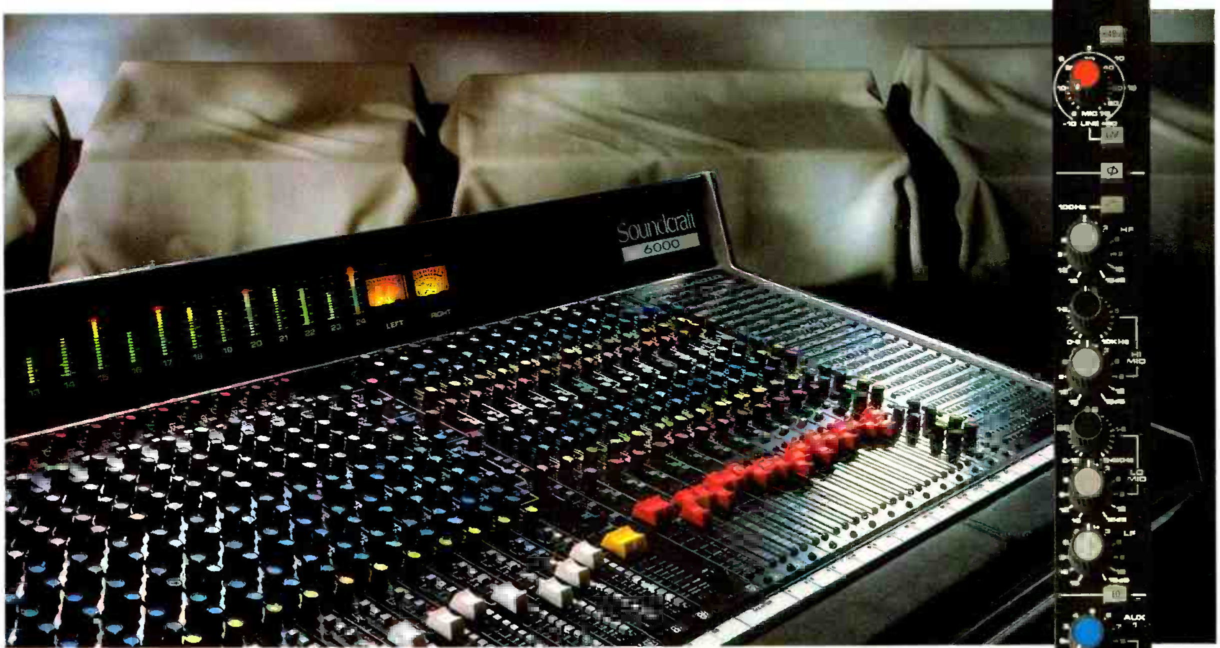
Are there solutions?

Yes. For example, real-time recalculation of reference data into a new timebase is available allowing varispeed synchronization between two transports, while their numeric difference remains constant. To maintain this resolution requires rates of information exchange more than 1,000 times faster than serial communication. This is achieved via parallel lines running at clock rates of 5MHz to 10MHz. Currently, this cannot be achieved relying totally on controller-to-synchronizer and synchronizer-to-synchronizer data communication at RS-422 rates.

It can be clearly demonstrated that features such as those described above will be needed to achieve future demands. Communication among devices such as

Bob Predovich is vice president and general manager of Master's Workshop, the electronic audio post-production division of Selkirk Communications Ltd., Toronto. He is also the co-designer of the Soundmaster Integrated Audio Editing System.

If Only More Expensive Consoles Performed As Well.



For a 16 or 24 track studio owner, the future looks very good.

With MIDI systems and digital outboard gear, you're faced with extremely sophisticated productions. But it's very hard to find a recording console to match the requirements without spending a small fortune.

That's precisely why we've developed the new Series 6000, an evolutionary design that clearly demonstrates the forward thinking of Soundcraft. Behind the classic layout is a revelation in performance and capability.

For one thing, it's equipped with enough busses and routing options to make adventurous productions a pleasure, not a nightmare. The 6000 is a full 16 or 24 buss console with six auxiliary sends per channel. The split format of the 6000 means each of the tape returns will double as extra inputs, with EQ.

We've also provided each input with push-button routing, EQ by-pass, and programmable electronic muting that eliminates the clicks produced by ordinary switches. You even get true solo-in-place, sadly lacking on more expensive consoles.

But it's the 6000's sonic performance that really sets it apart from the competition. Our revolutionary input design gives you 2dB to 70dB gain without a pad and virtually unmeasurable distortion, crosstalk, and noise.

Our new grounding system yields superb hum immunity and a routing isolation of 110dB (1kHz). And our active panpot comes close to theoretical perfection, exceeding our competitor's performance by a full 25dB.

The Series 6000 input module gives you programmable electronic muting under optional MIDI control, solo-in-place to get a clear picture of your progress, and a patented active panpot with isolation of 90 dB (1kHz).

To give you the subtle control it takes to achieve dramatic results, you also get four-band EQ with mid sweeps on each input channel.

When you specify Soundcraft's Series 6000, with options including 16 to 56 channels, stereo input modules, and built-in patchbay, you'll find it an affordable slice of progress. Series 6000, simply the most comprehensive production console in its class.

Soundcraft
6000

Soundcraft USA/JBL Professional
8500 Balboa Boulevard
Northridge, CA 91329

GUEST EDITORIAL

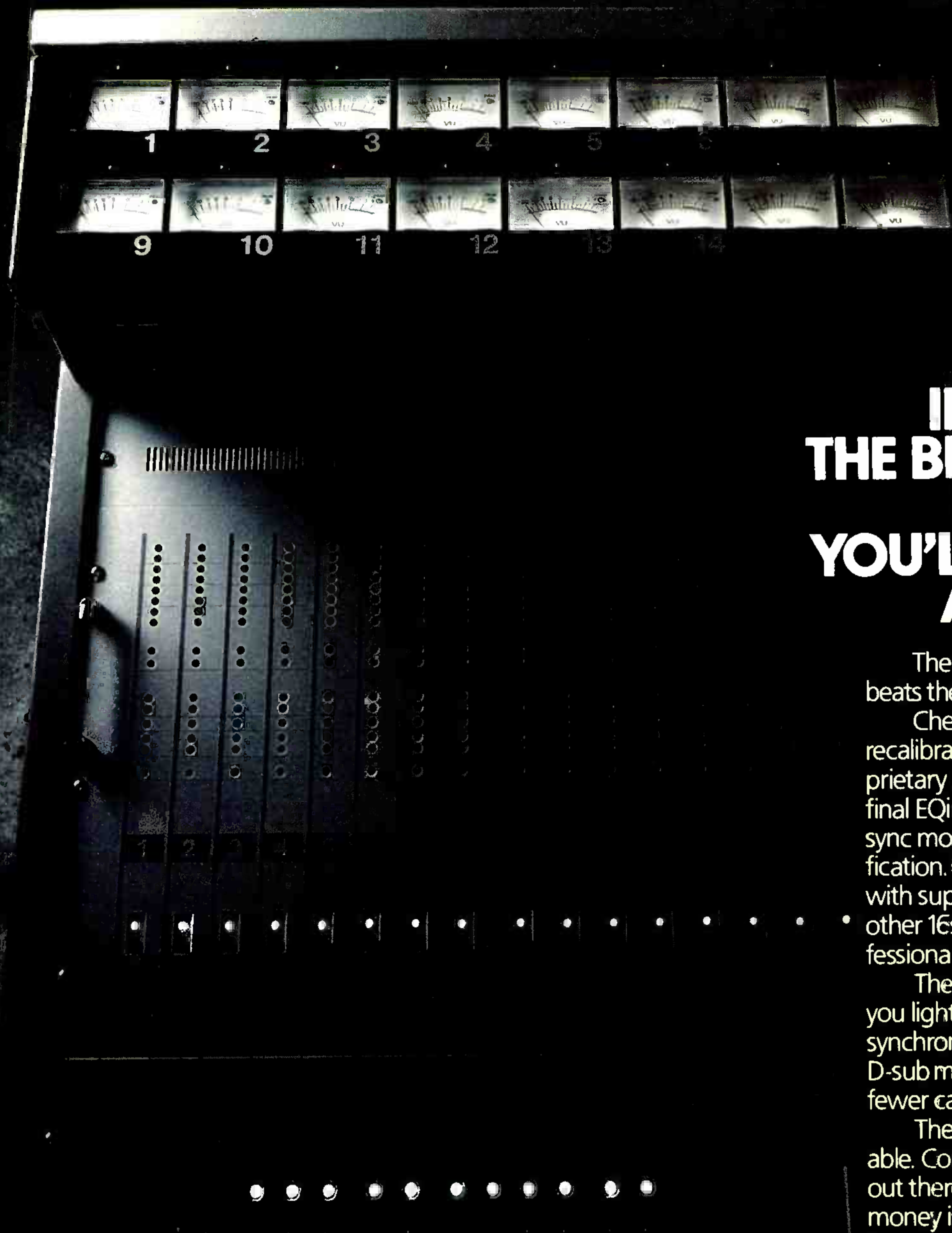
PC-type command units and the hardware processing those commands must be so fast as to produce a single virtual entity, unifying these two physical components to act as one.

Does this mean that the audio post-production industry has to give up the flexibility that serial communication will bring to the broadcast and video sectors? Not at all. It simply implies that we should use serial communication for data links that are not nearly so time sensitive (such as communication between the implementing hardware mentioned above), in this case, a synchronizer and the controlled device, such as a multitrack recorder.

We should use serial communication for data links that are not so time-sensitive.

Always keep in mind the need to augment the serial data with dedicated control lines for real-time requirements. It follows that these serial lines are totally independent and isolated from other serial communications occurring in a multi-machine system, and thus not susceptible to the congestion resulting from a daisy-chain configuration. It also follows that the information interchange among the synchronizers themselves, and between the synchronizers and the command unit, is occurring at far higher data rates. This leaves serial communication to do what it does best: act as a very flexible, long-distance "remote control" for commands issued from the synchronizer, removing the specialized needs of the audio post-production industry to another area that is better equipped to deal with them. The net result: Tasks given to the serial part of the system are more equivalent in difficulty and resolution to those found in the broadcasting and video sectors, and the special needs of the audio industry today, and for the future, are accommodated.

RE/P



IF YOU WANT THE BEST PRODUCTION 16 TRACK, YOU'LL HAVE TO SPEND A LITTLE LESS.

There's no getting around it. No one beats the 60/16 on features. At any price.

Check it out. Two speeds without recalibrating each time you switch. Proprietary head technology so accurate that final EQing decisions can be made right in sync mode without rewind and repro verification. Gapless/seamless punch in/out with superior transparency. And unlike other 16s, the 60/16 has built-in dbx professional Type - I.

The compact, rugged 60/16 also gives you lightning fast lockup for use with synchronizers, incredibly precise spot erase, D-sub multi-connectors for faster setup with fewer cables and, oh yes, brilliant sound.

There simply *is* no finer 16 track available. Compare it with any other machine out there. Then compare the price. If money is an issue, you may have to settle for the best.

TASCAM

Circle (7) on Rapid Facts Card

NEWS

E-V recalls loudspeakers

At its own initiation and in cooperation with the U.S. Consumer Product Safety Commission, Electro-Voice has recalled EVX sound reinforcement loudspeakers built and shipped between July 31, 1987 and Feb. 8, 1988.

The speakers contain uninsulated lead wires and overly long terminal lugs, the commission says, and may ignite the speaker cones during maximum excursion. Model numbers affected are EVX-150, -156, -180, -184, -1500 and -1800. As of late June, approximately 65 speakers were unaccounted for, sold mainly in New York and Northern California. The majority of the speakers are said to have been taken out of the United States.

Speakers should be taken back to the place of purchase. New speakers with a tested modification will be given to holders of EVX-150, -156 and -180 speakers. E-V will issue credit to holders of EVX-1500, -1800 and -184 speakers.

The commission says that all EVX loudspeakers purchased after Feb. 8 have improved performance characteristics and full safety modifications.

PAN debuts new system

The Performing Arts Network, whose System/1 is for musicians and others involved in creating music, has opened PAN System/2, designed for professionals who sell, market and support music. The primary emphasis is on people serving major tours and releases, and access is limited to users involved in some way with a major tour. PAN users can contact the system on-line for more information; for non-subscribers, contact PAN at Box 162, Skippack, PA 19474; 215-584-0300.

NED establishes training school

New England Digital has established an official training school for the Synclavier and the Direct-to-Disk system at the Full Sail Center for the Recording Arts, Orlando, FL.

The school, which has three systems, will provide courses in beginning and advanced Synclavier and Direct-to-Disk operation. The courses will be directed by Richard Head, owner of Song Bird Digital, a Nashville-based independent NED dealer for the Southeast, and an expert Synclavier user.

Classes will run six days a week for two weeks, with 48% of the time devoted to small lab groups for on-line training. The curriculum has been developed with NED and will be updated quarterly. Full Sail also plans to open a second facility in Los Angeles in 1989.

3M awards Lyras to audio teams

3M has presented its seventh annual Lyra awards to five audio teams nominated for Academy Awards in the achievement in sound category.

Receiving the awards were:

For "Empire of the Sun": Tony Dawe, production sound mixer; Robert Knudson, supervisor, re-recording mixer, dialogue; Don Digirolamo, re-recording mixer, music; and John Boyd, re-recording mixer, SFX.

For "The Last Emperor": Bill Rowe, dubbing mixer; and Ivan Sharrock, re-recording mixer.

For "Lethal Weapon": Bill Nelson III, production sound mixer; Les Fresholtz, supervisor, re-recording mixer, dialogue; Verne Poore, re-recording mixer, music; and Richard Alexander Cas, re-recording mixer, SFX.

For "Robo Cop": Robert Walde, production sound mixer; Michael J. Kohut, supervisor, re-recording mixer, dialogue; Aaron Rochin, re-recording mixer, music; and Carlos de Larios, re-recording mixer, SFX.

For "The Witches of Eastwick": Art Rochester, production sound mixer; Wayne Artman, supervisor, re-recording, dialogue; Tom Beckert, re-recording mixer, music; and Tom Dahl, re-recording mixer, SFX.

Original scoring mixers who received Lyra awards were Shawin Murphy, "Empire of the Sun"; Mike Jarratt, "The Last Emperor"; Robert Fernandez, "Lethal Weapon"; Eric Tomlinson, "Robo Cop"; and Armin Steiner, "The Witches of Eastwick."

Nagra to finalize portable digital recorder

Nagra has announced that it is working out the final details of a portable digital recorder, which will use a ¼-inch open-reel, rotary-head digital format, helical scanning heads and 7-inch open reels. According to the company, the open reel design will allow edits to be made easily in the field.

NY MIDI Expo set for December

The second MIDI Expo, a conference and exposition of MIDI music, is scheduled for Dec. 3 and 4 at the Sheraton Centre in New York. The expo will showcase the entire range of MIDI products for all aspects of music production, with 100 companies expected to exhibit. Also planned are demonstration rooms and a seminar program. For more information, contact Tony Scalisi, show manager, Expocon Management Associates, 3695 Post Road, Southport, CT 06490; 203-259-5734.

News notes

Lakeside Associates has moved to 1540 E. First St., Suite 243, Santa Ana, CA 92701; 714-836-6496.

Ultimate Support Systems, Ft. Collins, CO, has celebrated its 10th anniversary in business.

Gentner Electronics has appointed rep firms for the following areas: Audio Resources, Westmont, IL, North and South Dakota, Wisconsin, northern Illinois, Indiana and Kentucky; John B. Anthony Company, Stamford, CT, metropolitan New York and the surrounding area, northern New Jersey and Fairfield County, CT; McFadden Sales and Marketing, Columbus, OH, Ohio, West Virginia and western Pennsylvania; Northshore Marketing, Seattle, Washington, Oregon, northern and western Idaho, and western Montana; Plus Four Marketing, Walnut Creek, CA, Northern California and northern Nevada; and Pro Tech Marketing, Salt Lake City, Utah, Arizona, New Mexico, Colorado, Wyoming, eastern Montana, southern Nevada and El Paso, TX.

Fostex has manufactured a limited edition of its T-20 studio headphones, to be given to 200 industry professionals in performing, recording and music production. Each hand-made set comes with a personal thank you from Yoshiharu Abe, company president.

Innovative Electronic Designs has appointed rep firms for the following regions: Repworks, New England states; Darmstedter Associates, upstate New York; Bi State Marketers, northern New Jersey and metropolitan New York; Sigmet Corporation, the Eastern Seaboard; Secom Systems, the Southeast; C.L. Pugh Company, North Central states; Ludwig Marketing, the Ohio Valley and Illinois; CM Sales, Michigan; BC Electronic Sales, the Midwest; Marketing Concepts, the

90 ways to depart from the straight and narrow.

Prepare to expand your horizons. With the Yamaha DEQ7, the completely digital equalizer. With not one but 30 different EQ and filter configurations, in stereo.

To be specific, it has 11 graphic EQs, four parametric, three tone configurations, five band pass/band reject filters, six dynamic filters and one multi-notch filter configuration, plus up to 738 ms of delay per channel.

But, in the Yamaha tradition of never leaving well enough alone, we did some expanding of our own.

Like memory. The DEQ7 has sixty user-programmable memory locations, so you can store and title your favorite curves, and recall them any time manually or via MIDI. There are digital I/Os for direct connection to other digital components, as well as analog inputs and outputs.

Since we're on the subject of output, it's equally important to mention that, in the opinion of many critical users, the DEQ7 produces an open, natural-sounding EQ. Which means that you can create extreme EQ settings without creating extreme anxiety.

The Yamaha DEQ7 Digital Equalizer.

It'll throw you a few new curves. Catch one at your nearby Yamaha Professional Audio dealer.

Yamaha Music Corporation, Professional Audio Division, P.O. Box 6600, Buena Park, CA 90622. In Canada, Yamaha Canada Music Ltd., 135 Milner Avenue, Scarborough, Ontario M1S 3R1.

YAMAHA®
Engineering Imagination™

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NEWS

Texas Panhandle, Oklahoma, Arkansas and Lousiana; Torbett & Associates, Rocky Mountain states; North Shore Marketing, the Northwest coastal area and Alaska; and Island Instruments, Hawaii.

Audio-Technica has appointed three new sales agents. The Torbet/Keiser Group, Loveland, CO, will handle Montana, Idaho, Utah, Wyoming, Arizona, Colorado, New Mexico and west Texas. Progessive Audio Representatives, Little Canada, MN, will cover Minnesota, North and South Dakota and western Wisconsin. On The Road Marketing, Upper Montclair, NJ, will cover New Jersey, Long Island and upstate New York.

BASF has named Leo Burnett Company of Chicago as the agency of record for the audio/video department, and will assume the strategic advertising role for the company's consumer and professional lines.

Promusic Incorporated is now the exclusive U.S. representative of the Coloursound Music Library.

Aphex Systems Ltd. has signed a licensing agreement with Proton, allowing the company to market the Aphex ESP-7000 surround decoder as the SD-1000. Aphex will no longer market the consumer model but will retain the right to market a professional decoder using the surround technology. The company has also signed two foreign distributors: Gould Marketing, Montreal, for Canada, and LEAB, Stockholm, for Sweden.

Total Audio Concepts Ltd. has received the Export Award for Smaller Businesses, a national competition in the United Kingdom for independent smaller businesses. More than 300 companies entered; five received awards.

Optical Disc Corporation has received an Emmy Award for Outstanding Achievement in Engineering for the company's Recordable Laser Videodisc System. The award was presented Aug. 27 during the non-televised portion of the Emmy Awards.

Allen & Heath is the new name of the console and mixer manufacturer, known previously as Allen & Heath Brenell and AHB. The use of the company's original name is the first of a number of changes designed to increase visibility, according to the company.

Turbosound has moved to a new facility housing sales, marketing, accounting, manufacturing and dispatch in Partridge Green, West Sussex, in response to a 60% increase in sales in 18 months. The company's Capel facility, which formerly housed operations, will be used for raw cabinet quality control, wood finishing and painting. The company's new address is Star Road, Partridge Green, West Sussex, RH13 5EZ, England; 0403 711447.

Soundtracs has announced a joint venture with Samson Technologies Incorporated for the sales and marketing of Soundtracs consoles in the United States, effective June 1. In the United Kingdom, the company has split its product distribution arrangement. Larking Audio now handles consoles for recording, video post-production and other editing/production applications. Shuttlesound handles sound reinforcement, either mobile or installed.

Rainhill Tape Specialists, a Northern England cassette duplication plant, has taken delivery of a Lyrec P-2000 duplication system.

CompuSonics has announced that sales of its DSP 1500 have passed 100.

New England Digital has been awarded the International Teleproduction Society Monitor Award for engineering achievement, one of three engineering awards given this year by the ITS. The award was for the implementation of SMPTE time code in the Synclavier.

Ampex 467 digital audiotape has been chosen as the worldwide primary reference tape for digital open-reel audio recorders by the International Electromechanical Commission. The reference

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NEWS

tape, designated IEC primary reference tape #8000, is available from Ampex in ¼-inch, ½-inch and one-inch widths.

Simon Systems has incorporated in the state of California as Simon Systems Engineering Incorporated. Richard A. Simon, the company's founder, has been elected president and CEO, and Verlyn M. Simon is vice president and secretary. The company has moved to 707 Clear Haven Drive, Agoura Hills, CA 91301; 818-707-9980.

DAX Audio Group has named the following sales reps: Smith & Company, New England, New York and northern New Jersey; Taug Sales, southern New Jersey, eastern Pennsylvania, Delaware, Maryland, Virginia and Washington, DC; Raleigh Perry & Associates, Alabama, Tennessee, Georgia, Mississippi and the Carolinas; Northland and Associates, the Dakotas, Minnesota and western Wisconsin; Rowe Marketing Group, Illinois and western Wisconsin; and Michael Welch Enterprises, Florida.

Pro Media has consolidated its equipment sales, contracting and rental division to a new building at 3563 San Pablo Dam Road, El Sobrante, CA 94803; 415-222-0307.

People

John Howe, president of Telex Communications, resigned June 9, citing health reasons.

Murray Shields has been named director of sales at Auditronics.

Margaret F. Coppenrath has been named regional distribution manager for Agfa-Gevaert's Atlanta distribution office.

Randall Smith has been named Western regional sales manager for FOR-A Corporation of America.

Fuji Photo Film USA has announced three appointments. **Keith Scott** is account representative for professional products in the magnetic products division. **Tom Volpicella** is professional district sales manager in the Northeast and **Scott Petrozzini** is senior account representative in the New York metropolitan area.

David Armon has been named director of international sales for the Mitek Group.

Shure Brothers has announced two appointments. **Lottie Morgan** has been named vice president of sales. **Alan G. Hershner** has been named director of sales for domestic distributor products. The company has also named **Steffey Marketing,** Northbrook, IL, as its sales representative of the year.

Full Sail Center for the Recording Arts has announced two promotions. **Garry Jones** has been named senior vice president. **Dana Roun** has been named director of education.

RE/P

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

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

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

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

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

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

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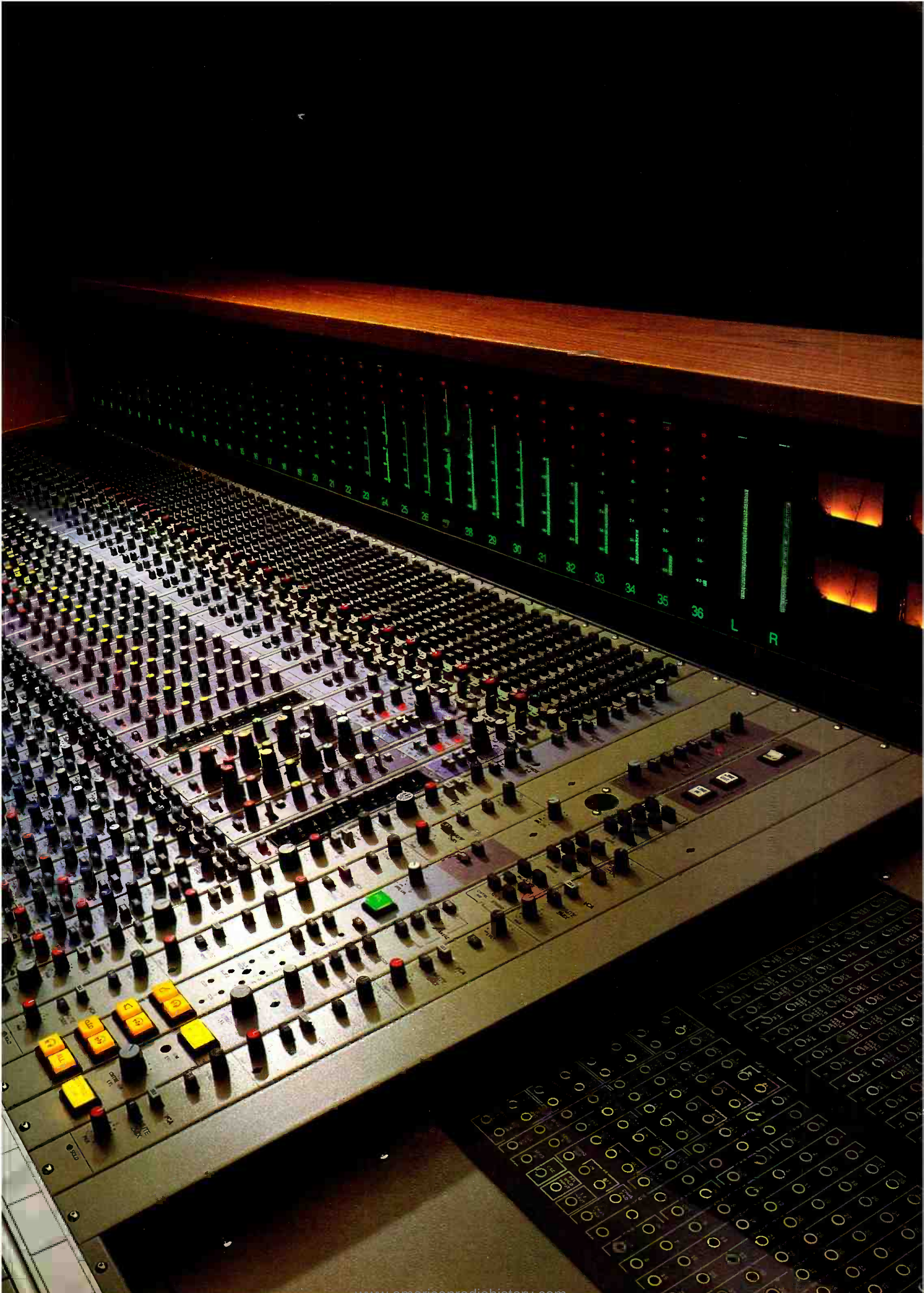
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By Paul D. Lehrman

A MIDI Primer

One of the joys of writing a regular column is that you don't have to please every reader every month. So sometimes I write for the MIDI veteran, who's been struggling with the stuff for years, and sometimes I write for the utter novice, who's just joining the good fight.

This month, the column is for the rank beginner. But even the most grizzled MIDiot shouldn't turn the page—there might be something here for you, too.

I spend a lot of time at classical and traditional music concerts, among other reasons to keep fresh in my mind what *real* musical instruments are supposed to sound like. At one such recent concert, an experienced recording engineer of passing acquaintance who specializes in acoustic music accosted me and asked, "If MIDI only has a bandwidth of 31kHz, how do you get 16 channels of audio onto it?" His question reminded me that, although little kids now talk about sampling and flangers as if they were candy bars and bicycles, there are still many fundamental misconceptions about MIDI among experienced audio professionals.

MIDI is not digital audio. Digital audio is a numerical representation of the waveforms that, when they travel through air, are interpreted by our ears as sound. If you put a MIDI signal through a D/A converter and a speaker, you would hear, if anything, a nasty whine.

MIDI is not music. Music is organized sound, and MIDI has nothing to do with sound.

MIDI is an instruction set, like a computer program, a piece of sheet music or the directions that come with a model airplane kit. By itself, it is nothing. But when it is interpreted by some device that knows how to, all sorts of things can happen.

MIDI instructions, or commands, mimic not musical sounds, but musical *performance.* The most common MIDI commands are "note-on" and "note-off" which, as you might imagine, are produced when you press a key on a keyboard and when you release it. These commands are accompanied by three important pieces of data: which note is being played or released, how fast the key is moving down or up and which channel the data is supposed to be sent on.

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How these commands are interpreted is up to the device receiving them. A note-on command can result in a single drum stroke, or it can initiate a series of events using an arpeggiator or sequencer, or it can trigger a five-minute sample of a full symphony orchestra.

Besides note commands, MIDI has what's known as "continuous controller" commands, which are used to represent more subtle musical changes, like the movement of a volume pedal, the bending of a guitar string, or the wavering in breath that a wind player uses to create vibrato. These commands are not really continuous the way an analog volume control is, but they offer much higher resolution—128 values—than the simple on-off of a keystroke. If that's not enough, often two controller commands can be paired in double-precision mode, to give 16,384 different values.

The MIDI specification allows more than 100 different *types* of controllers on each channel, so the range of musical expressivity that is conveyable over MIDI is very wide.

Another type of MIDI command is "Program Change." Composers writing for traditional orchestras understand this one pretty readily: Tell the third wind player to put down his alto flute and pick up his soprano sax. When a MIDI synthesizer receives a program change, it knows it is supposed to reach into one of its internal memory banks and pull out a new sound.

Program changes allow synthesizers to perform double, triple, etc. duty, changing their identities as a piece goes on. They can be sent down the MIDI line as fast as any other commands, so that each note of a piece could conceivably have its own program change. Some synthesizers don't often respond very politely to program changes, however, so this is usually not considered good practice.

All of these commands have musical names and uses, but the beauty of MIDI is that they don't *have* to be used strictly for musical events—they can be used for other types of audio events, or even for things that have nothing to do with sound. MIDI-controlled mixing consoles use program changes to go from one "scene"—a complete set of fader and other control settings—to another, automating the mixing process. An even greater degree of automation is provided by consoles that let the user assign MIDI continuous controllers to *individual* faders, EQ settings, sends, etc., so that all of the controls on a console can be operated remotely, or from a sequencer's memory.

Processing devices can also recognize MIDI data. Graphic equalizers that can store multiple curves can recall them when sent the proper program change commands, and compressors and limiters can do the same. Much exciting work is going on now with reverbs and effects processors responding to MIDI control in real time, with one type of controller determining RT_{60}, another taking care of flanger speed, and yet another controlling delay feedback.

Now, about all those channels. Having 16 channels on a MIDI line does not mean the bandwidth has to be 16 times greater than if there were just one channel. MIDI bandwidth is always the same: 31,250 bits per second (that's not *bytes,* mind you, so it would make a pretty awful digital audio format on even one channel). A certain amount of room is *not* put aside for each channel: getting data onto a MIDI line is always first-come, first-served, somewhat like a rush-hour subway train. If there's no room at a particular instant for a particular command, the command waits until there is, regardless of what channel it's on.

Receiving MIDI data is akin to getting TV signals off the air: picture 16 different TV sets connected to a single antenna, but each tuned to its own station. Now picture one MIDI line running through a dozen synthesizers, each one tuned to a different channel, and each one playing its own music. Now imagine each of those synthesizers as a section of an orchestra, and a sequencer is not only keeping them together in time like a conductor, it's also feeding them the sheet music as they play.

I think that's as good a way as any to start thinking of MIDI. I hope my acoustic friend can see now that MIDI and music are important allies, but they are not the same thing. RE/P

SPARS ON-LINE

By Tom Kobayashi

Opportunities in Film and Video Post-Production

In the 1940s and 1950s, before the growth of television, the only way anyone could obtain a position in audio post-production for the motion picture industry was to be related to someone already in the business. With its new technology, the TV industry created a large number of new jobs and opened up opportunities for audio technicians. However, as the industry matured, most of the jobs became unionized similar to the motion picture industry. Entrance into these positions also became difficult to obtain.

Have things changed? How does one get the opportunity to design sound effects for "Star Wars," to mix a movie like "Top Gun," or to be involved in post-producing the sound for "Miami Vice"?

Things are changing. With the advent of new technology, jobs are becoming more challenging, and we are seeing many new positions being created. But it is still difficult to enter the work force in places like Hollywood, where the majority of the work is unionized and large numbers of people are available. Knowing the right people, having patience and being in the right place at the right time are still important.

However, the situation is evolving in the post-production field, both in the type of jobs that are being created and in the background needed for these positions. No longer can someone's offspring be brought into a studio and expect to work their way up to becoming a sound editor or mixer without the proper training and a thorough understanding of digital processing. Fortunately, skill and experience are becoming the determining factors in landing a job.

Tom Kobayashi is vice president and general manager of Sprocket Systems, a division of Lucasfilm Ltd., and is a regional vice president of SPARS.

The merging of technologies from the video, music and computer industries have made great inroads into the television post-production arena. Because television was born of the electronic age, it has accepted new technology more readily than the motion picture industry. Digital samplers, synthesizers, the compact disc and hard disk editing systems are examples of the new technology that is accelerating the editing/mixing process and eliminating some of the drudgery of audio post-production. Much of this technology was developed in the music industry. Manufacturers are now adopting these new tools for television and some areas of the film business.

The use of digital processing equipment has opened the doorway for audio engineers in the music industry to cross over into the video and film post-production field.

The use of digital processing equipment has opened the doorway for audio engineers in the music industry to cross over into the video and film post-production field. The knowledge and experience gained from the use of synthesizers and MIDI devices has given them the background necessary to be hired over technicians with only sprocketed film experience.

On the down side, changing the methodology of film post-production is a slow and difficult process. With escalating budgets, no director wants to experiment with new technology on a major motion picture. The risks are too great, and there is usually no time in which to experiment during tight post-production schedules. For the most part, sound for films is being edited and mixed in the same way it was 30 years ago. The exceptions are that consoles have become automated and the recording machines are more versatile and complex.

Change is on the horizon. Digital consoles are being developed and sound effects are being put into random access CD jukebox or R-DAT libraries. With the proliferation of new technologies, the positions for editing, recording, transferring and mixing are beginning to overlap, thus eliminating specialized jobs. The number of educational institutions offering training in computers, digital signal processing and video post-production will continue to grow.

In the past, maintenance engineers for the film industry usually gained their experience through on-the-job training. Today, the maintenance engineer must have a specialized education and must continually study and know sprocketed, sprocketless and digital audio equipment in order to perform the job and keep the company competitive. The need for training through a trade school or college is becoming a necessity.

Another change that is opening up jobs in the post-production field is the move by major TV and motion picture production toward hiring non-union personnel. Even in union negotiations in Hollywood there has been a request by producers to eliminate union rosters. If the roster is eliminated, jobs would open for those who are qualified to work with the new technology. A qualified person, depending on skills, would not be barred from being hired and accepted into the union.

The result of these changes occurring in the film and television fields is the gradual opening up of the industry to accept the new breed of audio technician who, in the past, had an extremely hard time getting that first job.

Despite the apparent change, there is still a long way to go. There is a great need to re-evaluate the hiring process and increase awareness through comprehensive educational programs and curriculums. SPARS has contributed to the change through its examination program and the very successful internship projects launched at various studios. It would be wise for more companies to set up internship programs with universities that have music, video and cinema schools. This would allow students to develop into future engineers and technicians who are able to effectively deal with the ever-evolving technologies.

If you are seeking an audio position in film or television, or if you are a company wishing to explore new territory, please contact SPARS.

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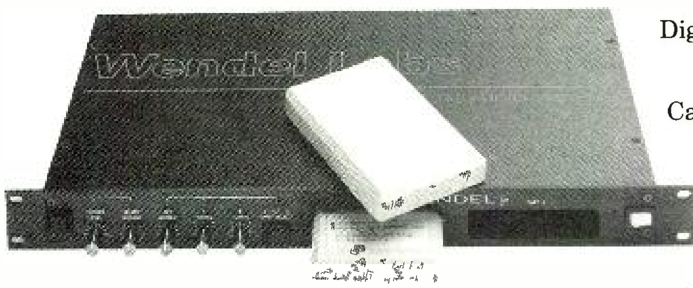

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UNDERSTANDING COMPUTERS

By Jeff Burger

Programming Languages

Last month, we looked at the heart of the computer, the microprocessor, and discussed how it interprets a stream of data as commands in its native tongue. Delving deeper, the microprocessor responds to different bytes, or data words, as instructions to perform functions such as manipulating an intrinsic number of subsequent bytes or words. This stream of data is referred to as machine code or machine language. Because working with a stream of data that looks like EA 01 5F 23 4C B5 10 69 A3 is not the average programmer's idea of a good time, many languages have been developed that are more appropriate for people to deal with.

Programming environments that are more akin to human languages are known as high-level languages, whereas those closer to the machine's native format are referred to as low-level languages.

Programming languages fall into two additional categories—interpreted or compiled. Interpreted languages can be typed into the computer and run more or less instantly. The name refers to the commands being translated on the spot into executable machine code. Compiled languages are typed into an editor, then slowly processed by the language's compiler into an executable file of machine code.

Each has its advantages and disadvantages. Interpreted languages have the gratification of immediate feedback. A program's integrity can be tested immediately, and mistakes are quickly detected and corrected. While interpreting is instantaneous by human standards, it is slow compared to actually running the machine code without interpretation time.

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

The advantage to compiled languages is that the final result is a program file in the microprocessor's native lingo, which runs much faster than its interpreted counterpart. The drawback is that compiling takes time and feedback is far from immediate at the writing and debugging stages. The larger the program, the longer the compilation time. I've seen some programs take the better part of an hour to compile, only to find an error afterwards.

Assembly language is the lowest-level language for any given microprocessor. Each machine code command has a mnemonic associated with it such as LDA (short for Load Accumulator Register) or INX (Increment X-Resister). Those acronyms are the rudiments of making programming more intuitive for us folk of the living, breathing variety. The resulting file is compiled into the actual machine code in a relatively short period of time.

The world of programming languages is as interesting and diverse as the many human languages.

Writing a usable program completely in assembly language offers the advantage of fast access to the complete power of the machine but simultaneously requires the programmer to take full responsibility for every aspect of what the machine is doing, at a fundamental level! Assembly language is more often used to write small, fast, subroutines which can be called from a higher-level language (more on subroutines next month).

The world of higher-level programming languages is as interesting and diverse as the many human languages. Different ones are better suited to certain tasks—sort of the "right tool for the right job" syndrome. The most common introductory language is BASIC, because of its similarity to English. It provides an easy way for hobbyists, students and other first-time experimenters to harness the power of the computer without facing a difficult learning curve. FORTRAN and COBOL are languages more oriented toward scientific applications. Languages like PROLOG are being developed, which are specially tailored to artificial intelligence. Certain computers have languages dedicated to getting the most from that specific machine, such as MacAPP on the Macintosh.

Music applications are most often written in compiled languages because of speed requirements. It is not unusual for time-sensitive portions of a program (such as dealing with a MIDI port) to be coded in assembly language for optimum speed. These efficient, modular routines would then be called by the main program written in a more manageable, higher-level language, such as C.

The C programming language has become very popular and provides an interesting case in point. It is designed around the concept of libraries of small-to-moderate-sized routines that can be reused from program to program. For example, say you have a routine that gets input from the user and tests it against valid entries. Why start from scratch every time a program is written that needs such a routine? Instead, this proven code goes into the archives and can be linked into any future program.

The other reason C is so popular is that it is transportable to any machine that supports a C compiler. In this way a sequencer program, for example, that has been written on an IBM can be ported over to a Mac, Amiga, Atari, C-64, or Apple II, because they support the C language.

Let's say the sequencer program needs to create an audible metronome "beep" through the computer's speaker. The main program might simply include a command that in effect says "beep the speaker." Each machine's compiler contains a library of routines that knows how to handle the common tasks like, you guessed it, "beep the speaker." In this way, languages like C are machine independent, and development time across many brands of computer is substantially reduced.

Again, you may use a computer all day long running various applications without having to even begin to think about the language it was written in, let alone how to program. Before we leave the subject of programming languages, next month we'll actually look at how a simple program might look. RE/P

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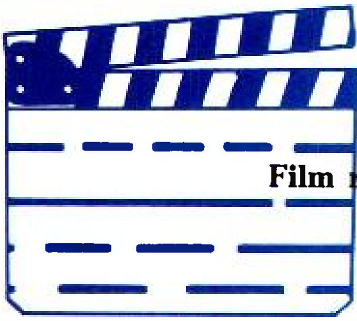
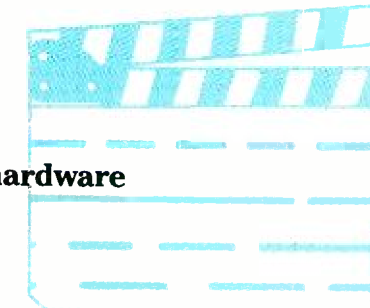
Film Re-Recording



By Dominick R. Tavella



Film re-recording requires specialized hardware and engineering techniques.

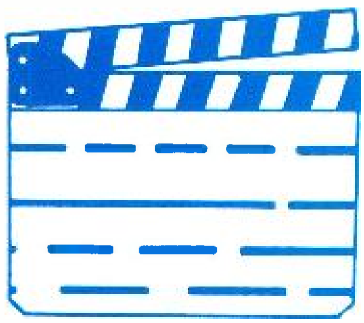
There are many differences between music recording and film recording, both in console design and engineering technique. Let's start by discussing basic film release formats.

There are several common formats for film audio, from mono to 8-track with stereo surround, but the most common are standard mono and 4-track stereo.

A standard, mono film soundtrack is usually known as an academy mono track. The academy in question is the Academy of Motion Picture Arts and Sciences, the Oscar people. Besides the Oscar, the Academy has given us a standard for the theatrical response of a mono film soundtrack, known as the academy curve. This is a rather severe frequency response curve, with a rolloff starting at 1kHz, and basically cutting off all above 8kHz. Severe as it is, the academy curve is a reflection of what happens in the real world: how your film will sound in the local Bijou.

In the late 1970s, Dolby Laboratories invented a system for making multichannel, high-fidelity theatrical sound possible at reasonable cost to both filmmakers and theater owners. This system takes four channels of audio (left-center-right-surround) and encodes them into two channels (called Lt, for left total and Rt, for right total), because photographic optical tracks can only handle two channels of audio. On playback in the theater, the system decodes the 2-channel audio into the four original channels.

The process works like this: During encode, left channel goes to Lt, right goes to Rt, center goes to Lt and Rt evenly and



in phase, and surround goes to Lt and Rt evenly and 180° out of phase. When decoding, everything in Lt only goes to the left speaker, anything in Rt only goes right, anything in both Lt and Rt that is equal and in phase goes to the center speaker, and all Lt/Rt out-of-phase information goes to the surrounds.

Film sound is usually mixed onto 35mm fullcoat, which is just 35mm film with a magnetic emulsion on it. Up to six tracks can be recorded on fullcoat. The machines used to record and playback mag film are called dubbers.

A mono film soundtrack is never actually recorded as a mono, or single-track, mix. Rather, it is usually recorded as a 3-track, or more commonly, a 4-track mix. The tracks are usually broken up into the broadly defined categories of dialogue, music, and sound effects. The fourth channel on a 4-track record is sometimes used for voice-over narration, common in documentaries, or for additional effects, such as ambience tracks or Foleys (dubbed effects, such as footsteps or clothes rustles).

One of the main reasons for this separation is to facilitate dubbing into a foreign language. Because the dialogue is on a separate track, only this track needs to be re-done, and the music and effects (known as the M&E) can remain intact. Similarly, any changes to the music or effects can be done easily without having to re-create the entire mix.

When recording a Dolby stereo film, the four tracks on the dubber are for left, center, right and surround. However, dialogue, music and effects still have to be kept separate. This is done by using multiple 4-track dubbers, one for dialogue, one for music, one for effects. These are called "stems." From these stems, the Lt/Rt 2-channel master is made, and a separate L/C/R/S M&E mix can be derived.

Different skills

The skills and equipment needed are somewhat different for film than for music. Music recording usually occurs in two distinct phases, the tracking phase and the mixdown phase. Although there are studios that specialize in one or the other, most music studios, and certainly most commonly used consoles, are set up to do both.

Because of this, most consoles are equipped with a full set of mic pre-amps for laying down the musicians on tape, plus a large and versatile set of auxiliary or cue outputs, for creating several different headphone mixes. There are two sets of output buses—the 24 (or 48) group outputs, to feed multitrack recorders, and the main stereo outputs, to feed a 2-track final master.

There are also two distinct signal paths—the main path (fed to the output buses), and the secondary, or monitor path (fed to the speakers). Both main and secondary paths can be EQ'd, fed to aux or

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Dominick R. Tavella is re-recording engineer at Photomagnetic Sound Studios, New York.

cue, and panned from left (odd) to right (even) outputs.

Film mixing doesn't require a buildup of elements. Because the tracks are built up by sound editors beforehand, there is small need for a full bank of mic pre-amps. Although there are ADR (automatic dialogue replacement, or dubbing dialogue) and Foleys to be recorded, these are usually done in specialized facilities, before the mix, transferred to 35mm mag, and physically cut into the tracks by the sound editors. (However, it's nice to have a few mic amps on hand, as there may be the need to record a quick Foley or dub that was overlooked in the editing stage.) There is little or no actual recording taking place in a film studio, so there is little need for an elaborate cue/phone system. The actual name for film mixing is "re-recording."

The largest differences between standard consoles and film-specific consoles are in the panning and monitor sections. Because film is true 4-channel sound, provisions must be made for quad panning. Although simple left/right panning will suffice in many situations, there must be a way to do complicated special moves, circular swoops and the like.

Also, true quad panners allow for precise positioning in the L/C/R/S field. Because of the nature of the Dolby matrix and the way it separates channels, this can become important.

The monitor section of a film console is usually where most of the custom work goes. Film monitoring is significantly different from music monitoring. A secondary monitor fader is not used in a film console. Rather, all signals to be monitored, both the sends to the recorders (the console bus outputs) and returns from the recorders, are routed through a central switching matrix. This matrix allows the engineer to toggle between send and return on any particular channel, or combination of channels.

In addition, any channel can be soloed or muted. The individual channels are then routed to up to eight different speaker positions (left, left-center, center, right-center, right and L/C/R surrounds). This monitoring is all done at fixed unity gain because the monitor signal must be an accurate reflection of the mix as printed to fullcoat.

Because all EQ and reverb are done in the original recorded mix, there is no need for these functions in the monitor path. There is no need for dynamic panning, either, as the return from the fullcoat track designated "Right," for instance, must go to the right speaker and nowhere else.

In addition to this matrix, there is usually a custom panel installed in the console for machine transport function and recorder control, as there are no standard remotes or autolocators for dubbers, nothing "out of the box." There are individual record on/off switches, as well as safe/ready switching, for each recorder channel, with masters for each machine.

Film transports allow forward and reverse motion in real time, as well as up to 6x speed. One of the real advantages to film mixing is the ability to print a mix, in real time, while going in reverse. This feature can save a lot of time in certain situations. Another advantage is the ability to easily slip sync any element in the mix, because they're all on different transports—and there could be a lot of transports involved.

Large consoles

It is not unusual for even a modestly budgeted film to have 50 or 60 tracks, and complicated films can end up more than 100 tracks. As a result, film consoles tend to be very large. A traditional Hollywood mix uses three engineers, one each for the dialogue, music and effects, so film consoles also tend to be split and built in sections.

Engineering a film session is a little different, too. For one thing, the soundtrack doesn't exist in a vacuum. It has to work with and support the visuals, as well as add to the thematic structure of the film.

There are also purely technical constraints. Because of the nature of optical soundtracks, there is a limited frequency response and dynamic range available. In order to maintain consistency from studio to theater, there are published frequency response curves to equalize the studio to, and monitor levels are fixed at 85dB/SPL per speaker.

The actual mixing requires some specialized skills. First and foremost are the dialogue tracks. The dialogue has been recorded in a variety of locations, all out of sequence, sometimes with differing equipment from shot to shot.

Thus, the quality of the dialogue is often vastly different from scene to scene, with wildly divergent amounts and types of background noise, traffic, airplanes, you name it. Throw in the odd dubbing line, done in a dry recording booth. These tracks have to be made to sound consistent from shot to shot and scene to scene, and pleasing throughout. As a result, a lot of signal processing (EQ, gates, compression, de-essing, etc.) is done.

The music tracks are usually less of a problem, having been recorded in a controlled environment (usually). A problem can arise if the music was recorded in 2-track (L/R) stereo, and is to be used in a 4-track (L/C/R/S) film. Because of the way the Dolby matrix works, a stereo music mix can, more or less, collapse into the center speaker position, ending up sounding fairly mono.

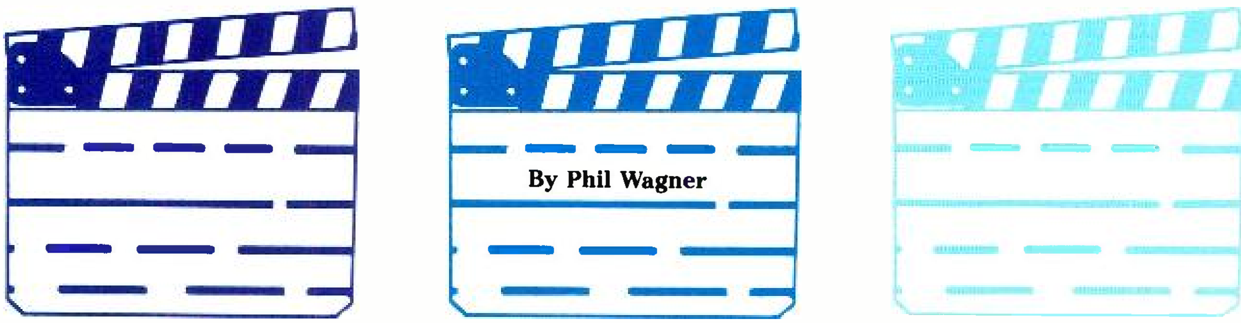
For this reason, any film intended for L/C/R/S film release should be mixed through a Dolby matrix and monitored in a L/C/R/S environment. The final mixing processes should also be done through the Lt/Rt matrix to discover any anomalies before it is too late.

Effects tracks are the most numerous in a fix mix and come from many different sources, such as location recording, SFX libraries or a Foley stage. Once again, they have to be blended within the rest of the track and made to sound consistent with each other: the dialogue and music tracks, and the visuals of the film.

As you can see, a film mixing studio is a somewhat strange beast compared to the average music room. Film rooms almost always require custom consoles and have to spend a lot more for transports. (A single dubber costs about $9,000. Multiply that by 25 or 30, and compare that to the price of a 24-track recorder.) Film mixing engineers need demanding and specialized skills, so good ones are hard to find. There is a limited demand for film mixing services as well.

All this adds up to a high client cost for a film mix, so studios have to provide a high degree of client satisfaction to remain profitable. This brings out the best in everyone, from console manufacturer to re-recording engineer. RE/P

Custom Consoles: Sound One



By Phil Wagner

Recently, Rubert Neve Inc. had the opportunity to design a custom film re-recording console for Sound One Corporation, which was looking for new film-style consoles for two studios under construction.

The project began when Dominick Constanzo, project engineer of Sound One, and I were introduced at the SMPTE show in October 1986. About a month later, Dominick called to discuss plans for their next phase of expansion.

Dominick gave an overview of their film mixing process and how Sound One's engineers like to operate. The setup is as follows: The dubbers (single-channel reproducers) feed the console line inputs. These are processed, mixed and routed to the 24 output buses, which feed up to four 6-track mag recorders. The output of these 6-track recorders feeds 24 monitor inputs of the console (see Figure 1).

However, the manner in which the console output (bus) and the machine output (mag) are monitored are different on film consoles. A secondary monitor fader is not used, as is common in music recording.

Instead, a centrally located, monitor switching matrix is used. This matrix enables the engineer to rapidly change between the bus or mag signals for level matching, while constructing music, dialogue and effects "stem" mixes or the final mix. Mag returns are then routed to one or more of the eight speaker feeds for positioning in the mix. In addition to monitor switching, the matrix provides solo and record "ready" for any of the 24 mag tracks.

All dubber and mag machines are electronically interlocked to the Magnatech 8LB. The 8LB feeds a biphase signal to the Adams-Smith synchronizer that runs the console automation. It also feeds a biphase to the film chain for motion control.

Because film engineers can mix in either forward or reverse, motion controls are used to change direction and speed of the machine transports in the film chain.

Phil Wagner is Eastern regional sales manager for Rupert Neve Inc., Bethel, CT.

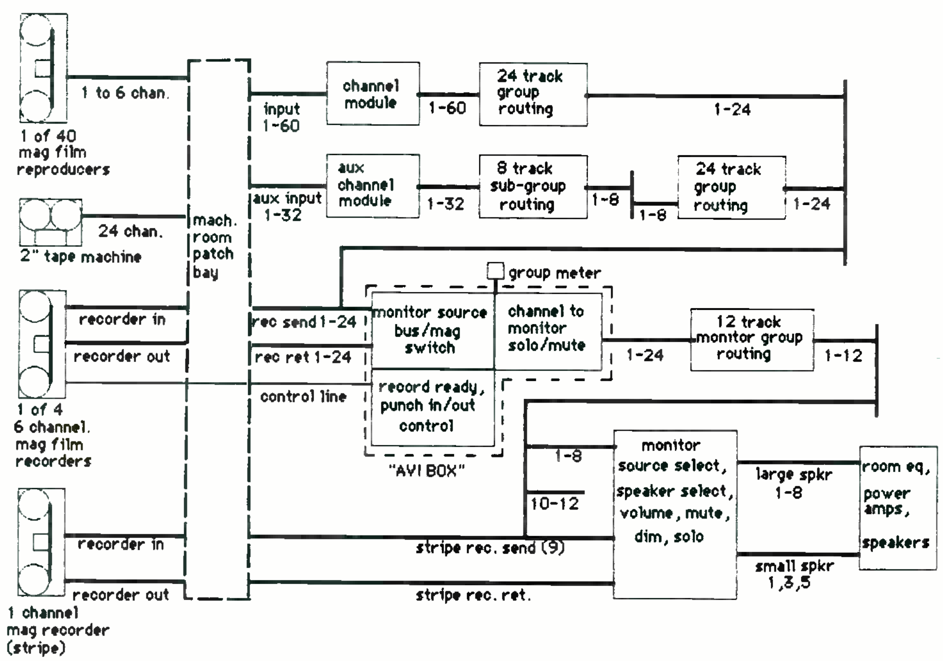


Figure 1. Routing configuration for Sound One custom film re-recording console.

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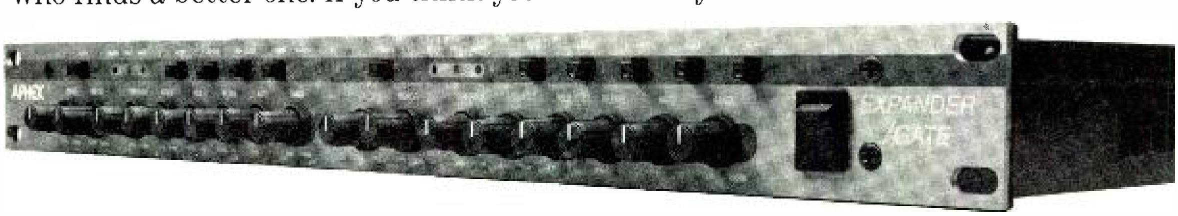

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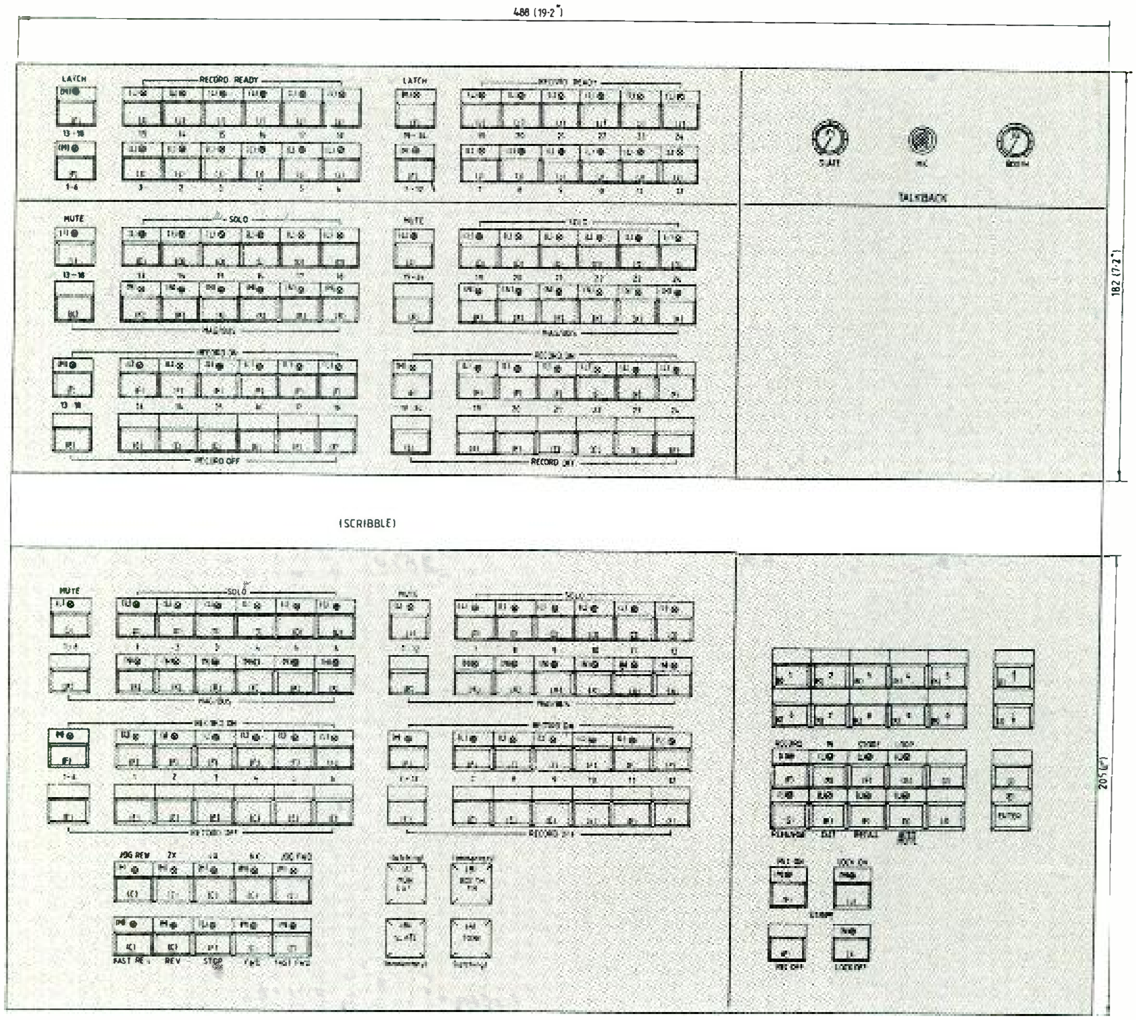


Figure 2. Control panel for bus/mag monitor switching, record ready, motion control and display for automated record in/out function (AVI Box).

Talkback and stripe (single-channel recorder) controls are also provided.

Console configuration

In the early stages of discussion, Dominick and I looked at the possibility of repeating the existing design of the Studio F console which was practical for compatibility between the other rooms. A second idea was to supply a standard V Series modified for their application.

The console configuration was to be as follows:

1. Channel in to bus out: 60x24.
2. Aux input to bus out: 32x8/8x24.
3. Mag return to monitor out: 24x12.
4. Necam 96 moving fader automation.

Mechanically, the console consisted of five buckets (12 inputs each), two master sections, patchbay and a two-bucket-wide producer's desk. The overall console length was about 16 feet.

In February 1987, after a few preliminary designs variations were discussed, design engineer Andrew Bachelor of Neve U.K. met with the Sound One staff to speed up the design quotation process. This was done because of the time constraints under which Sound One was working.

During this meeting, all design possibilities were thoroughly examined. At that point, it was felt the best approach was to provide a custom version of the V Series.

The new consoles were to include an automated monitor switching matrix, which was being designed by Avi Laniado, Bob Troeller and Dominick Constanzo of Sound One. The monitor matrix box was dubbed the AVI Box, with all hardware and software to be designed on a CAD system, and manufactured by the Sound One staff and integrated into the console design architecture by Neve.

Technical provisions for the proper interface and installation of the AVI Box had to be carefully planned. The AVI Box was to be installed by Sound One after the console was delivered. Close attention was placed on the input and output connections, connector types, grounding methods and interface levels. Neve was to provide the metalwork for the AVI Box, which would be electronically and physically compatible to the console.

This monitor section was to incorporate a microprocessor enabling automated control of the bus/mag monitor switching, record "ready" status, and record punch-in/out. Facilities were incorporated for the record in/out times of all functions to be entered on the fly or preset manually in foot/frames via a numeric keypad. An LED footage counter was to be provided showing the current position as well as record in/out times (see Figure 2). This

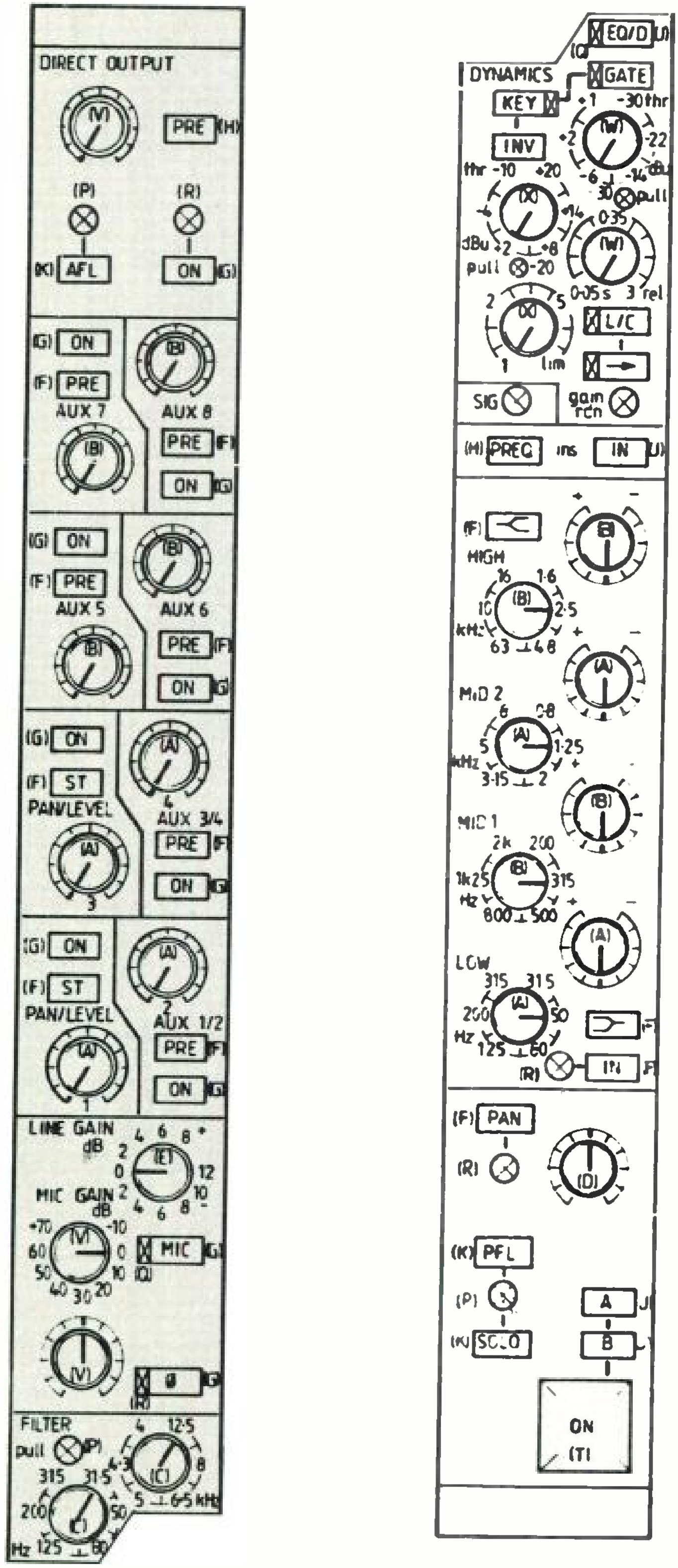


Figure 3. V Series channel strip orientation for Sound One custom film console. EQ and Dynamics/front end are placed close to mixer for ease of operation.

receives its foot/frame location from a Magnetech 9F footage counter.

The implementation and placement of each function on the board were addressed. The Sound One engineers wanted the equalizer and dynamics sections to be close to the operator. These could be located directly above the channel fader, because the in-line monitor on the channel is not necessary for film mixing.

The eight auxiliary sends (four with individual pre/post and on/off, four with pre/post for the pair and stereo operation) were to be placed at the top of the module (see Figure 3), and the routing for the 24 buses was placed in the overbridge section for easy access and identification. All of this was possible due to modular construction techniques.

Sound One asked if we could provide auxiliary line inputs for pre-mixes in projects that exceeded 60 dubber and mag returns. As this discussion took place, Andrew drew a module that was small enough to fit 32 inputs into an 8-module wide (12.8 inches) console monitor section. These 20mm wide inputs have basic facilities such as level, mute, and A and B aux sends, which can be assigned to the eight aux sends. Each aux input has eight sub assign buses that feed the 24-output bus routing.

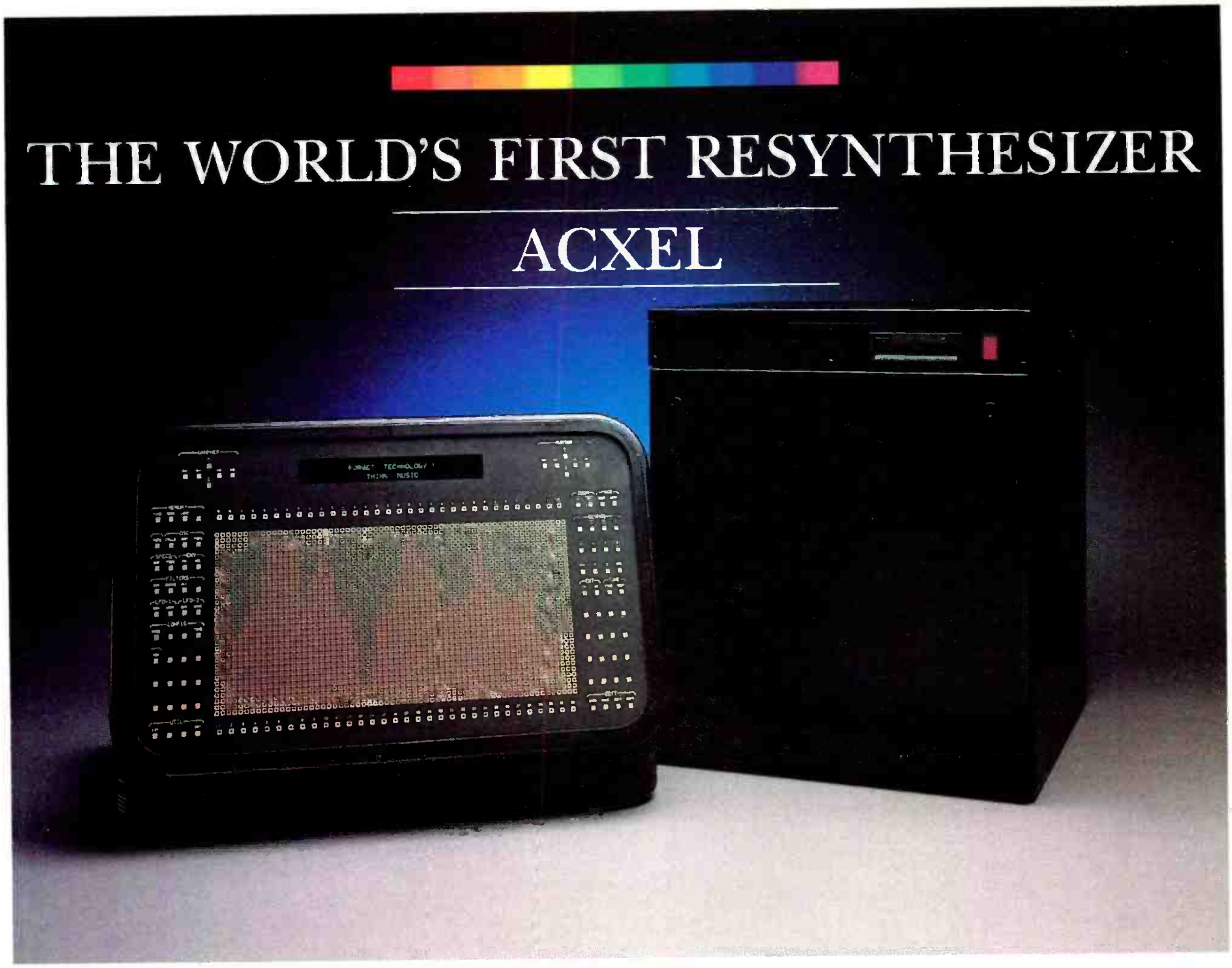
A master facilities section was also designed from scratch to handle output levels of all buses, an 8-way ganged speaker level pot, speaker muting logic for various mixing tasks and other master controls. Simultaneous VU/PPM plasma bargraph metering was to include 24-bus/mag signals, eight sub assign buses (aux inputs), eight aux send buses, eight monitor sends and eight aux monitor sends (stripe, stereo, mono).

Two monitor switching control panels were to be placed strategically in the console. One was placed between inputs 24 and 25, below the 32-channel aux input mixer. The second was placed between inputs 48 and 49, below the master facilities section. They were in parallel control of the AVI box and film chain.

The next two months were spent refining the design, including several issues of drawings, until Sound One was satisfied that every critical detail was reviewed and incorporated in the design.

All of this culminated on June 1, 1987, when Sound One agreed to purchase two purpose built Custom V Series consoles. Sound One took delivery of the first console for the new Studio L on March 1, 1988, and was fully operational in two weeks. The second console was delivered for the redesigned Studio D in August.

RE/P



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Custom Consoles: Skywalker Ranch

By Larry Blake

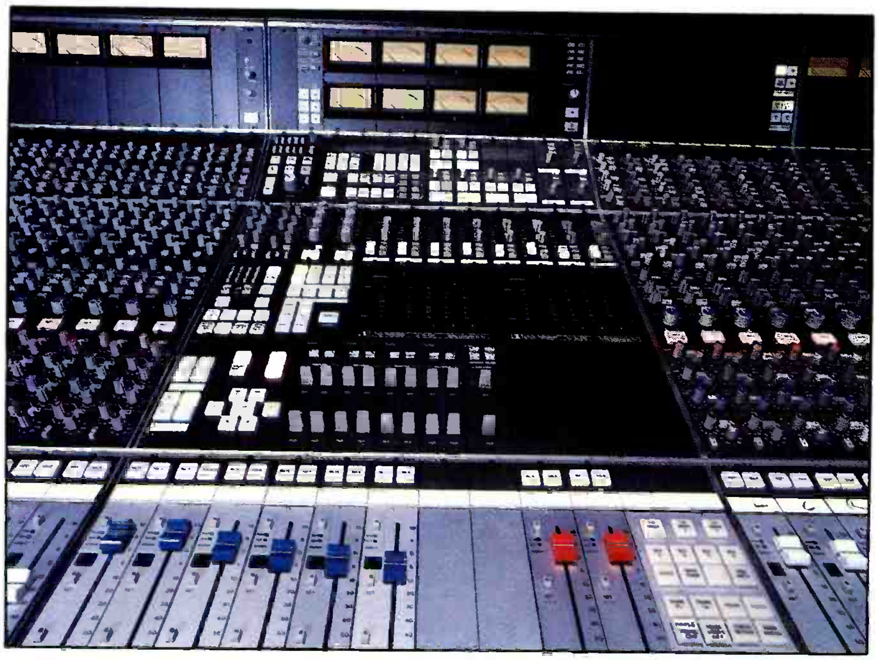


Photo 1. Series SL 5000 M 72-channel, 3-position console at Todd-AO's stages.

Photo 2. Master center section of the stage S, with an SL 5000 M console, at Todd AO's CBS Studio Center.

In the past year, Solid State Logic has begun deliveries of its SL 5000 M Series film production consoles, its first board designed specifically for the film market. The technical building at Lucasfilm's Skywalker Ranch is the home of the first four consoles off the production line, including one Foley console and three re-recording boards [see the August issue].

With the development in 1985 of the SL 5000 M Series consoles, which have a user-specified modular design, the staff at SSL saw the opportunity to build a film board. Initially intended for the broadcasting market, the film console consists of a mainframe of four different depths, each with a fader and from three to six "cassettes" (modules).

The minimum channel configuration consists of an input/output cassette, a pan cassette and a fader. Up to three additional cassettes may be ordered per channel, with the most popular being the equalizer, auxiliary sends, dynamics and channel input meter. The configuration may be swapped. For example, at Todd-AO in Los Angeles, the equalizer was placed immediately above the fader.

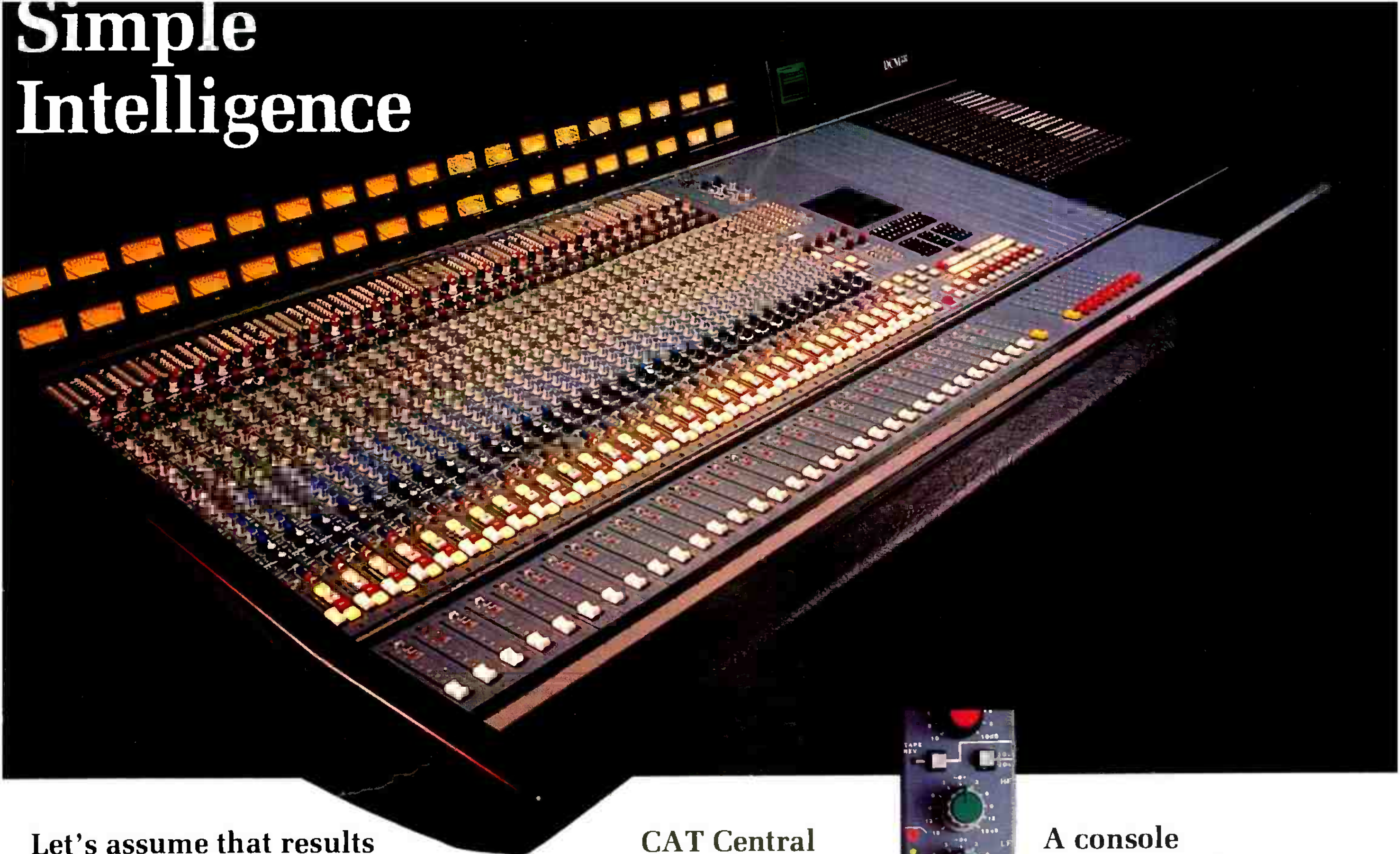
Custom modules

In adapting the M Series broadcast consoles for film work, the design and engineering team had to come up with custom modules for the varied demands of motion picture re-recording. Two of these concern panning, which is normally a simple, single, knob in record or stereo TV consoles.

One of the mandatory cassettes in each fader channel is pan cassette, which can be either the two-channel model (SL507) inherited from the broadcast version, or

Larry Blake is a sound editor and mixer at Weddington Productions, North Hollywood, CA.

Simple Intelligence



Let's assume that results are what's important, not which tools you use.

Audio production is at least as much art as science; there will always be those who ascribe a magical aura to certain pieces of equipment. But if your client list is built on quality and consistency rather than techno-voodoo, the DCM 232 in-line console with CAT automation can give you more of both.

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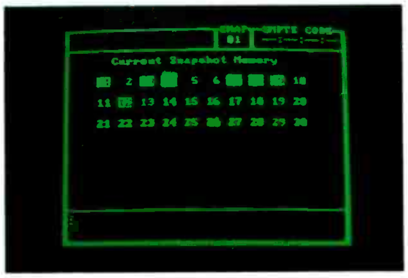
The DCM 232's Central Automation Terminal controls one of the most ingenious automation systems ever to shorten a mixing session. Along with the precise fader and muting control you'd expect, the CAT system includes advanced functions like *Channel Copy* that lets you duplicate a channel's signal flow as many times as you need to. The computer will recall a "snapshot" of most console switch settings manually or via SMPTE code.



With all of its convenience functions, this CAT won't leave footprints all over your tracks. The DCM 232 maintains an overall dynamic range of 100 dB with at least 22 dB headroom at each stage, thanks to exacting calculation of every circuit component.

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The advantages of the DCM 232 in-line console with CAT automation are explained more fully in our brochure. To get a copy, write on your letterhead to the appropriate address below.

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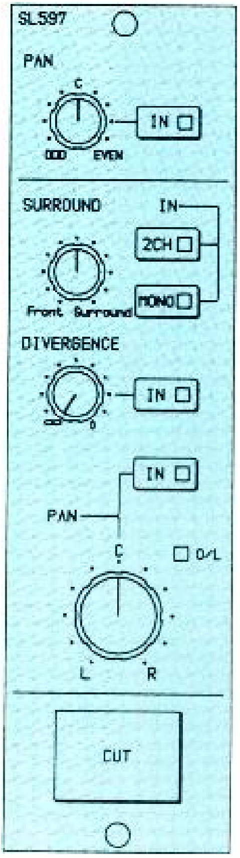


Figure 1. The tree-output section film pan cassette model SL597.

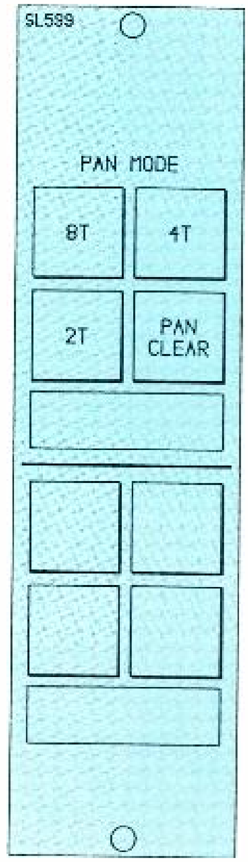


Figure 2. Two-, 3- or 8-channel pan mode master cassette model SL599.

the SL597 Film Pan cassette (see Figure 1). The SL597 can be adapted either for standard 2- or 3-channel "behind-the-screen" panning, or in the 8-track mode (full surround panning between any points in a mono or stereo surround format). Mode switching for all channels in the console, or each section, is controlled by an S1599 Pan Mode Master cassette (see Figure 2).

For more elaborate programmable panning, the SL598 Programmable Joystick Pan can be used. A small microprocessor in the console converts movement and controls selection to MIDI to communicate out to the machine room to another microprocessor that drives VCAs. The joystick gives X-Y coordinates of its location and the processor converts the coordinates into VCA levels.

In addition to all standard output configurations, the joystick panners allow for a user-definable mode where parameters can be placed into a PROM slot. Each panner can have a maximum of eight outputs, which are normally sent to one of the four groups directly.

Output bus configurations

The console design allows for a minimum of four output buses and a maximum of 32, depending upon how many 4-channel SL594 bus master cassettes are installed. In addition, each channel contains a direct output, which is selectable as pre- or post-fader.

The Lucasfilm consoles take this design a bit further, with a total of 128 main mix buses. Each section—dialogue, music and sound effects—has 32 separate mix buses.

Brian Kelly of Lucasfilm, who played a large role in the development of the SSL film consoles, says that this specification comes from having "done so many bizarre movies—on "Captain Io," we used two 32-track digital machines, a 24-track, two 6-track film machines and all our playback dubbers. Times like that make you say 'We're not getting stuck here again.'

"It's not that much more expensive to make it more flexible and if you only use it two or three times it pays itself back in a matter of hours—instead of the downtime to reconfigure everything."

Because of the infrequent need for more than 32 buses simultaneously, the standard dubbing configuration at Lucasfilm has the A/D groups summed to the recorders so that, operationally, the mixers are sharing the same 32 outputs.

Console automation

The SSL Total Recall and Studio Computer systems may be installed, although the data are not interchangeable between 4000 or 6000 Series consoles. A new implementation is Instant Reset, which allows the position of almost all switches to be reset in less than a film frame.

The exact time depends not on the width—the number of channels—in a console, but on the depth (the number of cassettes). A 5300 Series (fader plus three cassettes) can be reset in 21ms, with each additional cassette level "costing" 7ms in response time.

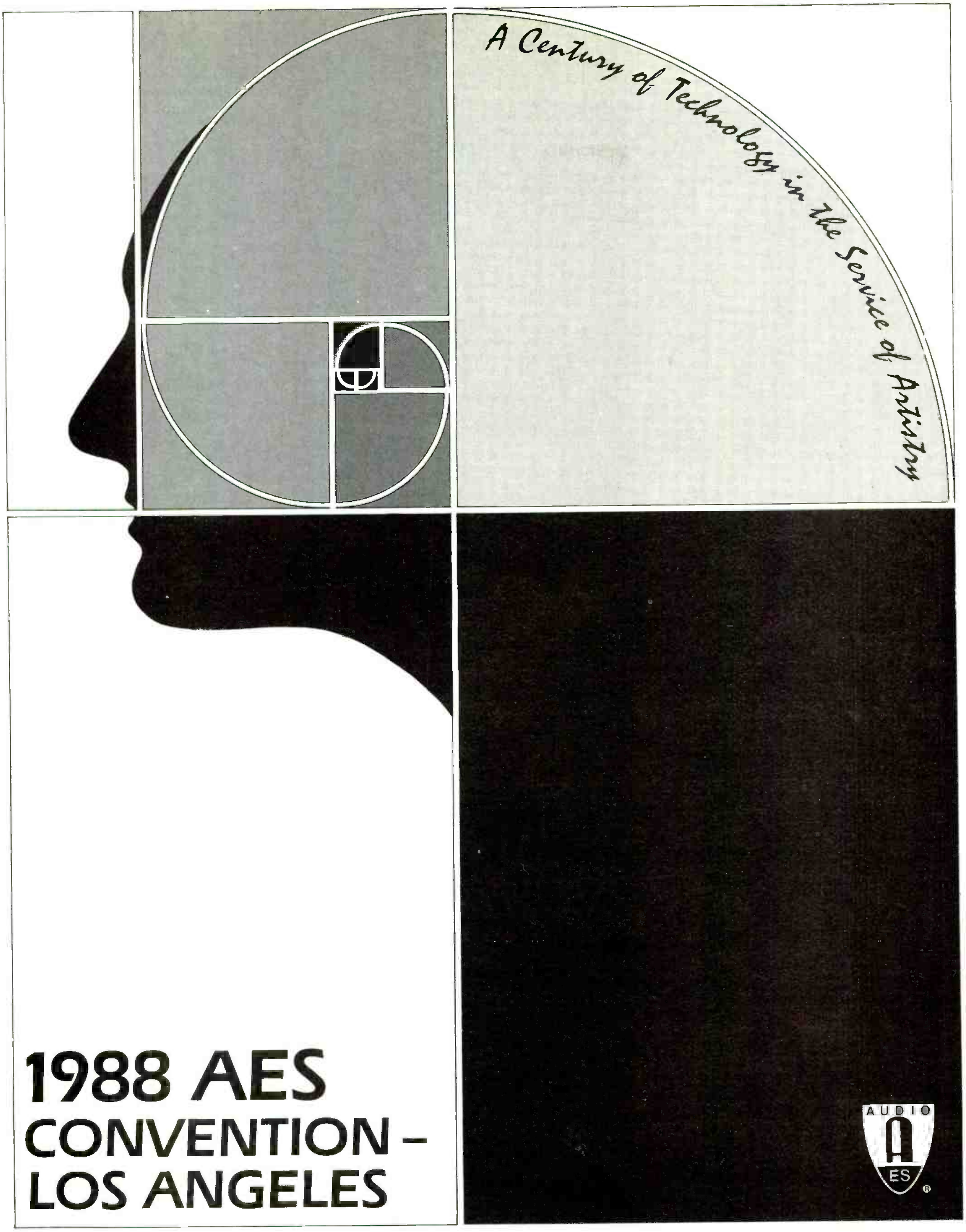
One of the most obvious ways for mixers to use Instant Reset in stereo film dubbing would be to facilitate moving quickly between pre-mix, final mix and print mastering modes. Among the switches controlled are equalization and aux send in/out, output bus assignment and monitoring mode. Total Recall and the Studio Computer allow for storing the position of rotary knobs and matching dynamic fader and joystick panner movement.

At the moment, the Instant Reset system cannot be referenced to time code addresses, but studios that choose to could buy a SMPTE-to-MIDI reader. Because "Instant Reset" communicates with MIDI and has a MIDI In, changing any of the standard preset numbers, on an external device, can change the console setup.

Monitoring

Probably the biggest single difference between the SL 5000 M Series and its sisters, the SL 4000 and SL 6000 recording and post-production consoles, is the monitoring section. In the process of recording the dialogue, music and sound effects stems, across as many as 32 tracks (as opposed to the eight in most stereo TV mixes), provision has to be made for off-tape/input switching by each mixer. In addition, monitor and recording insert points have to be provided for the stereo film encoding formats. The matrix inside the monitor section of the film console is 112x56x8. The first two numbers accommodate off-tape/input switching, with the last conforming to the maximum number of speaker channels in a theatrical mix format.

Although all crosspoints are accessible for custom setups, presets are used for common recorder/monitor configurations. This presumably reduces the chance of error that is present when setting up a complex configuration. RE/P



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Audio For Video Post Production: A Basic System

By Wilber W. Caldwell, Bill Quinn and Joe Neil

Understanding the basic hardware requirements will help any studio that is considering entering the audio for video post production market.

We got your attention, didn't we? You're probably thinking, "These guys are going to presume to define for me the elements of a 'basic system?'"

Well, actually we are, but in a different way than you think. First, we are going to make the definition completely nonspecific when it comes to equipment brand names. Second, we are going to try to work backwards, as it were; that is to say that the definition of the system itself is purely a function of what the system is going to be required to do. Now this may seem obvious to you, but the point here is that you can accomplish some pretty sophisticated tasks with a pretty simple system, if you know what you are doing.

Defining the basic system

"What do I need to get into audio-for-video post-production?" Studio owners ask this question over and over. The constant recurrence of this inquiry has triggered some soul-searching on our part. On the surface the question appears naive. But try to answer it and you will quickly see that it is a sophisticated question indeed. Each part of the answer is multilayered.

Where do we begin? By understanding that the audio-for-video post-production area, like the rest of the audio industry, is poised on the threshold of the digital age. Digital videotape is very much a reality, and the audio performance and capabilities of the D-2 format appear exceptional.

Disk-based audio recording and all of the new digital editors and workstations are naturals for A/V post. Indeed, they offer almost every advantage, including improved sound quality, faster operation in many cases, and improved audio manipulation via SMPTE code. Nonetheless, they have their disadvantages as well.

All of these devices require some sort of downloading to archive or to transport component material from studio to studio. The last-minute, emergency remix in New York presents a problem for the studio in Los Angeles, where all of the component tracks are on a hard disk. The likely solution for now is to transfer it all, track by track, to multitrack tape and call Federal Express. No doubt, satellite data links and compatible, transportable, intersystem, digital storage media will provide solutions to these and other digital problems in the near future.

All of this said, today's analog systems are numerous, affordable and, at last, reliable and functional.

Digital aside (but not for long), what do you need? Well, let us first ask another question: "What do you want the system to do?" We don't mean, "Do you want to sync two audio decks to a ¾-inch videocassette deck?" We mean, "What tasks do you want to accomplish?"

Describing the basic system

The sidebar "Basic System Configurations" describes a system for an independent "for hire" recording studio. Although it may seem like a tall order at first, remember that this is a basic system.

Wilber W Caldwell is the president of Doppler Studios. Bill Quinn is the chief engineer at Doppler Studios. Joe Neil is the senior engineer and the chief technician at Doppler Studios and owner of Sam's Tape Truck, a remote recording service.

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First you need something that will playback picture on videotape. The minimum acceptable format is probably a ¾-inch video tape. It is not absolutely necessary to have a machine that can access address track time code. The address track is a third, longitudinal analog track. Code can be recorded on one of the audio tracks, but remember that some off-line video editors work with address track code, and a significant portion of the pictures that you receive will come from an off-line suite. You will probably save yourself a heartache if you have a machine that can, at least, read address track code.

If ¾-inch video playback is the minimum, then 1-inch, type C is the present optimum. This is a very real option today. With the advent of better and slicker 1-inch, type C machines, and with D-2 around the corner, the used machine market is quite soft. Although you can get by without 1-inch type C, it is difficult, especially in larger markets. This is the on-line standard.

A lot of audio comes in the recording studio door already synced up and laid out on the 1-inch, and a lot of clients prefer to walk out of that same door with the mixed audio neatly "laid back" to the same 1-inch. In addition, in smaller markets, not every video facility can play ¼-inch tapes with code in sync to picture. This is especially true of TV stations where, in most cases, if the audio is not on videotape, there is no synchronous playback.

Now that you can play back the picture, the next thing on your shopping list is a machine upon which to record: an analog multitrack with a servo capstan and a similarly driven ¼-inch 2-track recorder. These machines are on hand in most studios. There are sophistications of course (center-channel time code, 4-track, sprocketed components in sync via SMPTE synthesizers, neopilot tone equipment and so on.)

Next, you need synchronization equipment: a synchronizer for the multitrack slave and one for the 2-track slave, and a synchronizer or a synchronizer controller for the video master, depending on the system you choose. It is advisable that each potential slave have its own synchronizer, but there are ways around this.

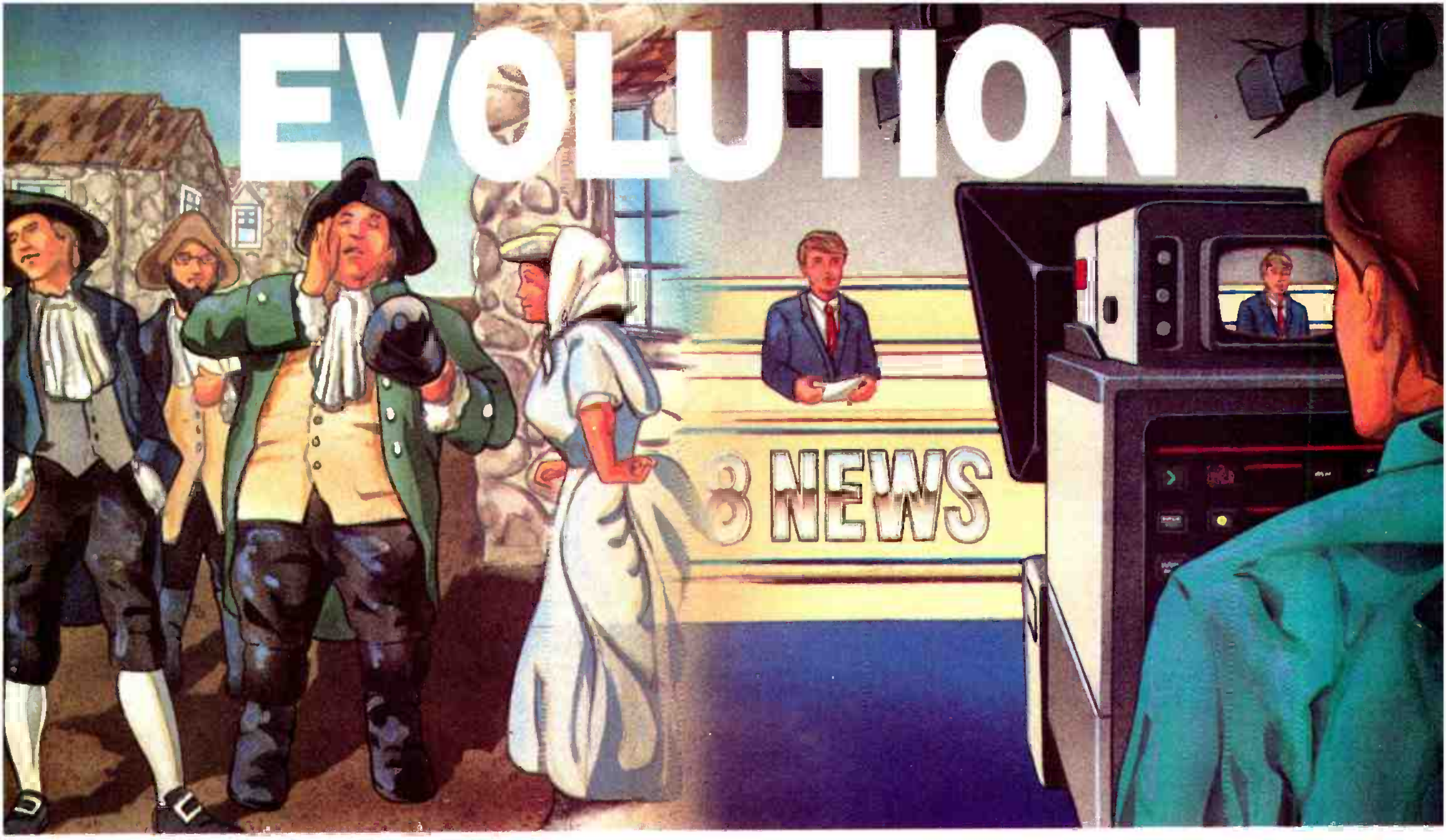
For example, it is easy to run any number of cables to a synchronizer, and then plug in the appropriate interface before the session, depending on the audio recorders selected. In this case, preproduction and adequate setup time are essential. You should be sure to choose synchronizers that can accept an external time-base and thereby act as a speed reference as well as a sync reference for the audio machine that it controls. The choice of a synchronizer controller is perhaps the most difficult and the most critical decision of all.

We will not go into the features of the various competitors in this field except to say that it is advisable to choose a controller that has, at least, the following features:

1. The ability to communicate with synchronizers, change status, and adjust internal registers.

2. The ability to preview edits in order to determine edit points based on both the location, the length and the internal "hits" of the audio source material.

3. Some type of programmable, soft keys for stacking complex commands, in order to create and save personalized, repeatable operator technique sequences in a single key stroke. Synchronizer controllers that feature elaborate cue sheet screens and interfaces with video edit



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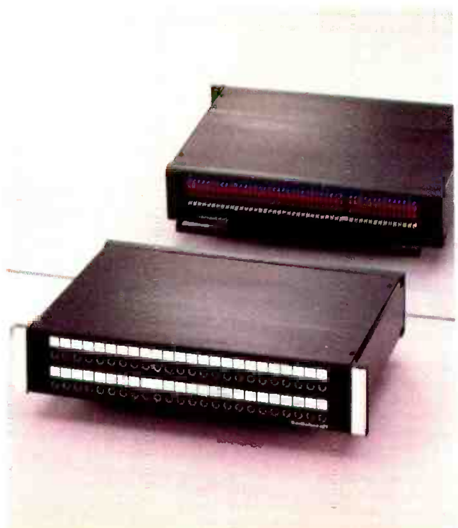

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decision lists are particularly useful when dealing with long program material.

The last items on the basic system list are perhaps the least expensive yet most important. First is house sync: a first-rate sync reference generator that conforms to the NTSC RS-170A standard and will thus put out reliable black burst and vertical drive. Don't try anything without it. Next, a high-quality time code reader/generator that can lock to house sync, jam sync and insert SMPTE code into picture. You will also need at least two large video monitors. Don't skimp on the video monitors, the money spent will pay off in the long run.

Using the basic system

Now, with the basic system in place, we can go back to our list of functions.

1. Recording music to picture.

It is easy to see that the system can sync a multitrack recorder to videotape and, thereby, record music to picture. A very useful tool in recording music to picture is the event programmer, a device that reads time code and closes a relay at a selected time code number. Applications include starting the metronome when laying out click tracks for scoring and starting a sequencer for electronic scores.

2. Recording voice to picture.

With both a multitrack recorder and a ¼-inch two track synced to the videotape, the basic system can handle recording voice to picture in several different ways depending on the nature of the project. On narrations for broadcast television or industrial programs, an announcer might be recorded directly onto a single track of the multitrack machine using backup and punch-in techniques, just like in music recording. Recording announcers for TV commercials usually requires the synchronized 1/8-inch deck to record all takes, while only the "selects" are saved on the multitrack. In this way, the system can accommodate the inevitable substitution of alternate takes and other subsequent incremental adjustments after the announcer has gone home.

3. ADR and looping.

Any number of dialogue replacement schemes can be devised within this basic set up using the multitrack for "selects" and the ¼-inch as a running backup for all takes. Possibly the most sophisticated of these schemes is an attempt to simulate "film style" looping. Film people are pretty set in their ways. In the case of looping, this is not without good reason. The efficiency and quality of the classical, sprocketed looping session is tough to compete with on any analog SMPTE-based system. (However, this is an area where the new digital disk-based systems have an opportunity to soar.) Here is a method that we have worked out using the basic system to accommodate feature film looping. It takes a little homework, and it is usually a 2-man job in session: one to engineer and one to act as sort of a production assistant/clerk.

We begin with all the film reels transferred to ¾-inch videotape with burned in time code and with the production sound for reference on audio channel 1 and SMPTE code on channel 2. At the picture start mark of each reel we assign the time code number to be the reel number in the hours column and zero to the minutes, seconds, and frames.

Then we take all of the film footage information for the loops from the cue sheets and convert it to time code in order to do pre-program edits. (We have a computer program that is designed to do this.) Our synchronizer controller supplies "beep" cues for the actors. During the recording process, select takes are saved on the multitrack. The engineer changes the offset as is appropriate. All takes are recorded on the ¼-inch. (Most feature film producers will prefer this ¼-inch safety to be a neopilot tone tape referenced to the house sync.)

Note that 35mm mag dubbers are not part of the "basic" system, but at the end of the session, the selects can also be transferred, in sync, to individual 35mm mag stripe reels, which are always locked to the picture.

4. Laying out synchronous and non-synchronous material.

If the material supplied is time-coded, then it is a routine matter to lay it out on the multitrack using the editor/controller. The programmable "soft" key feature on the controller is partially useful in this scenario, for most engineers develop their own pet tricks for finding offsets, adjusting to various "select" takes and making subtle, incremental "sync trims."

If it is non-synchronous material, it could be transferred to pre-striped ¼-inch, or leadered up and laid out using the event programmer or even laid in loosely on the fly. In some cases, leadering is still the fastest way, regardless of the lack of sophistication. This is usually true if using a SFX or music library that is already head-leadered, or if the music and EFX don't require absolute sync.

5. Foley work.

Most Foley work can be handled using the same procedures described in the "Recording to Picture" sections above.

6. Automated mixdown.

Console automation is a must simply because it does not take very long to construct a multitrack tape that is nearly impossible to mix. Remember that we are mixing to picture, and, accordingly, it is necessary to watch the picture, not the console faders. Automation allows some freedom in this area. At least an engineer can review and then perfect a mix without looking at the console marks. With automation, identical mixes can be laid back to multiple 1-inch masters, to dub masters and to various safety masters. "Music only" or "effects only" mixes are easy to run off, as are alternate versions, not to mention the inevitable remixes and revisions. In addition, many engineers develop automation techniques that allow them to use a kind of "mix as you go" idea for the layout/tracking session, joining the mixes section by section as they go. Thereby they are all but finished with the mix at the end of the layout.

7. Correcting speed errors.

The video world is not a perfect world. It is not unusual to find that somewhere along the line in the complicated video transferring and editing process someone has taken a nonsynchronous step, and thus original sound components will no longer match the reference audio on the edited video.

Here the basic system is capable of some

Basic system capabilities

1. Record music to picture.

2. Record voice to picture; keeping and editing "select" takes on the multitrack while simultaneously keeping a synchronous safety master of all takes.

3. Do some form of precise ADR or film-style looping with a complete, simultaneous backup of all takes.

4. Layout synchronous and non-synchronous voice, music or effects from a variety of formats, with previewable edits using an editor/synchronizer controller and or a precise event programmer.

5. Accomplish Foley work.

6. Accomplish repeatable automated mixdowns.

7. Calculate and correct speed errors to repair component synchronization errors.

8. Provide a house sync speed reference.

9. Make the time code system compatible with various MIDI setups.

10. Repair or otherwise get around faulty or damaged SMPTE code.

11. Deliver a product with a time code reference that is compatible with the end user's technical requirements.

pretty neat tricks. One approach is to time a section of the original program and time the same section of the reference audio on the video tape and then calculate the speed error in percentages. This done, the original program is varispeeded to match and then striped with SMPTE code. Synchronize the two tapes and check the result. It may take a few tries to get it close. Small timing errors can then be calculated by lining up the very beginning of both tapes and noting the offset and then lining up the very end of both tapes and again noting the offset. The difference in these offsets is the amount that the slave must be changed during the transfer.

If, for example, the program is four minutes long, and the difference in these offsets is eight frames, then the offset must slowly be changed at a rate of 25 subframes every 7.5 seconds. While transferring, use your synchronizer in the slow lock mode (capstan override is limited to a rate undetectable as wow) and trim the offset 25 subframes every 7.5 seconds. Timing errors like the one described are often encountered when working on multitrack recording which began in "personal use" studios employing SMPTE to MIDI converters. This type of hardware often uses the SMPTE to start a sequence but not to synchronize continuously, or, just as often, the hardware is not told to do so in the software.

8. Provide "house sync": NTSC 170A (black burst and vertical drive [59.94Hz]).

9. Make time code compatible with various MIDI setups.

MIDI systems are often a bit squirrelly when it comes to the time code that they will accept. When feeding time code from the basic system to a MIDI system, it is essential to have the ability to flip the phase, make it balanced or unbalanced and to control the level. In addition, some SMPTE-to-MIDI converters do not like to see backwards or offspeed time code. Accordingly, it is sometimes necessary to send code only in the locked, play mode and not in shuttle, rewind or startup modes.

10. Repair or jam faulty time code.

The basic system can accomplish this only if there is a control track on the video or a 59.94 reference tone on the audiotape. If there is a glitch in the code on the videotape, the reader/generator can read and reshape the good code and then be switched into the jam sync mode, wherein it will continue to generate new code and reference to house sync. The repaired code is then audio inserted onto the videotape before the glitch, and the new code is restriped to the end of the program. A similar scenario can be realized with a multitrack with bad code, if there is a 59.94 reference track that can be locked to house sync. Tapes with faulty code that have no control track are nonsynchronous material to the basic system and must be handled accordingly (see item 7 above).

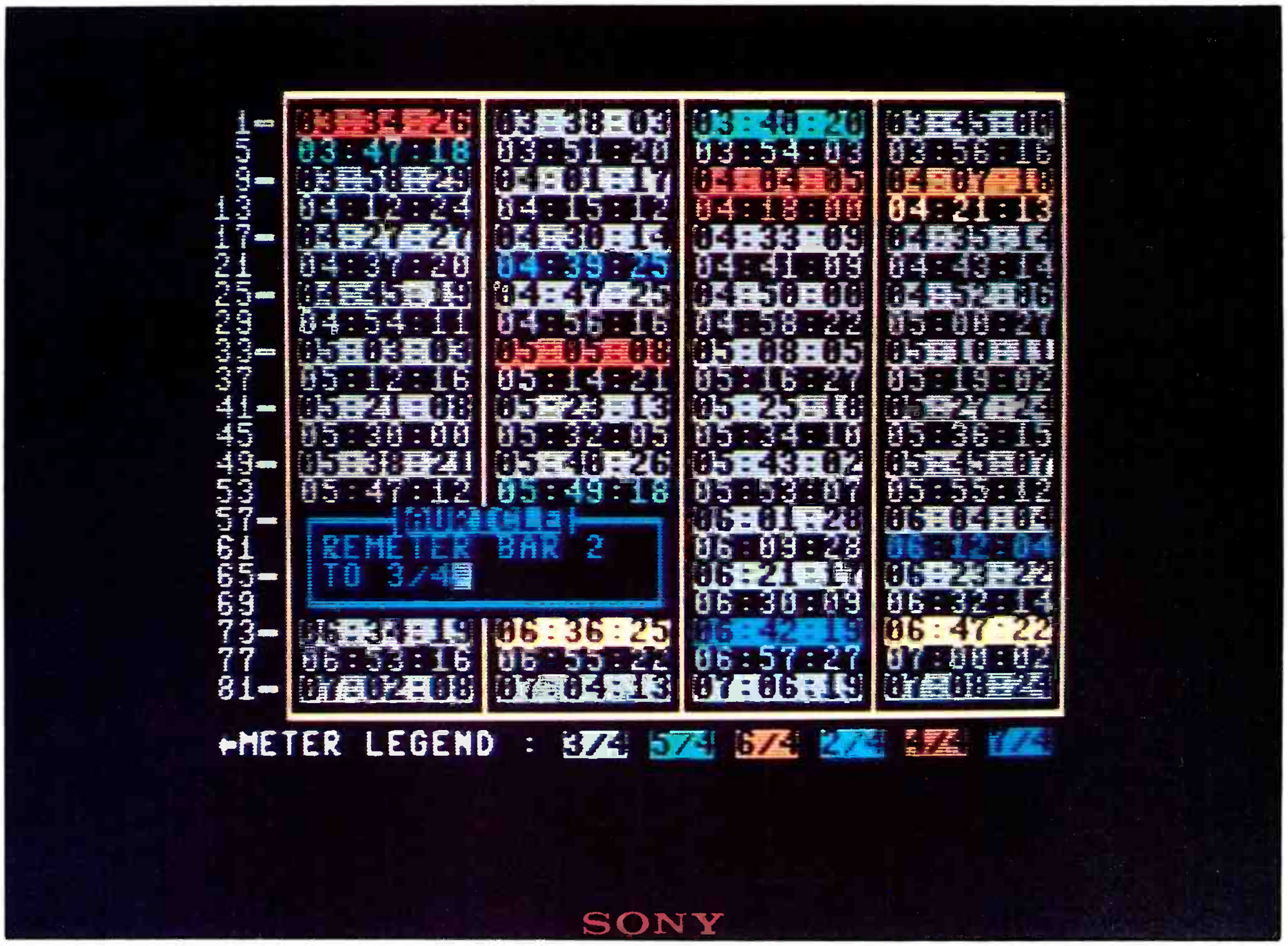
11. The delivery medium.

This is a matter of convention within various areas of the industry and within various markets. The question is: How can the end user of the mixed audio product best deal with the product: 2-track with code, 2-track with center-channel code, 1-inch type C, 4-track split track and so on? The answer may not lie within the basic system, for it is at the convention and the whim of your clients and your market's preference.

Anyone can buy equipment, but it is the operator that makes the difference. The basic system described above is nothing but a tool. Nonetheless, and for all its simplicity, it is a very powerful tool in the hands of an experienced and imaginative operator.

RE/P

Hands On: Scoring Software Update



Whatever computer system you use, there is a variety of software programs available to simplify the scoring process.

By Charlton Pettis

When I was young, I was told that music and mathematics were both conducted by the same part of the brain, and so were, on some deep level, intimately related. When I was a bit older, I was told that creativity thrived under structural limitations, the more limiting the structure the more creativity flowered.

No wonder then, as I sit before a SMPTE-burned workprint of a rough cut of "Return of the Psycho Fur Pig" (no, not a real movie) with a massive click book on my knee and crumpled wads of score paper on the floor around me, I feel I have been betrayed and lied to.

Scoring for film is, historically, a product of nine parts tedium, math, and frustration ("If only that shot were 10 frames

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longer..." "What do you mean the director cut two feet out of the scene...") and one part creative joy.

Bring on the computers! There now exist several products on the market whose sole objective is to free the composer from the tedium and rigor of clicks and counts and the myriad gyrations involved in melding music to an edited picture.

There have always been different methods employed by composers. Some, working with counts supplied by the film editor, have completely mapped out their work in advance before the actual composing begins. Others have preferred to improvise to picture, trying to capture the feel of the cue, before going back to fit the music to the hits and cuts precisely. (Hits are specific visual cues to be punctuated by the musical score, cuts are changes in the scene or camera angles).

Software has developed to facilitate both these methods of composition, and those methods combining the two. The two primary types of products, mirroring the historical duality of methods, are the click track/layout variety (like Cue and Auricle), and sequencer programs (like 48 Track PC and Performer), which have been expanded to include film scoring and SMPTE lock utilities.

While this makes the transition from traditional to computerized methods comprehensible and relatively painless, it does not, I hope, shut the door on a new way of confronting the compositional process, a way that uses the power of the computer to integrate the creative and logistical aspects of the task.

But until such a hybrid evolves, there are several programs on the market that admirably serve the composer and markedly lessen the gap between realization and actualization, between the art and the science of scoring to picture.

This article will be subdivided by computer model, so the reader can quickly evaluate the programs available for the various machines.

Macintosh programs

The Mac seems to have garnered the lion's share of the film scoring market. It alone has two state-of-the-art dedicated film click programs, in addition to another for SMPTE-based automation and sound effects and several impressive sequencers with their own film and video hit-mapping capabilities.

As if this wasn't enough, through MIDI files, click tracks and meter maps from the click programs can be dumped into the sequencers. First we'll look at a couple of the click/layout programs, then a couple of the composition-oriented sequencers.

Opcode Systems Cue

Cue is becoming pretty much a standard for film music programs. With its Master Cue Lists, Performing Rights Cue Sheet, and score printing, it is clearly geared for the working professional. This is not to say that it is difficult to use, because it isn't. It does a great deal, so it takes a while to learn, but it is an extremely well-designed and friendly piece of software. It is so friendly, in fact, it allows for several different ways of approaching the problems of film scoring.

A full review of the program by Bill Cavanaugh appeared in the July issue, so to avoid gross redundancy, I'll limit my discussion here to a brief overview of the program's capabilities.

Basically, Cue does just about anything you could ask it to, and a few more you probably never thought of. You enter hits in the Input window in any format you want, such as filmic (foot/frames), various forms of SMPTE, and EBU, along with optional descriptions of the hits and whether they are key hits.

Only key hits are included in the next stage, which is the Search Tempo function. Here, you specify a tempo range and an acceptable margin of error. Cue finds a tempo that most accurately catches all of your key hits. Alternate methods of entering hits include an onboard stopwatch, which you summon up from the Input window. This little utility allows you to grab hit timings when your workprint doesn't have a visual SMPTE window burn.

On the slightly more sophisticated side, the program locks to MIDI Time Code (MTC) so you can grab hits on the fly. Getting increasingly sophisticated, if you can get a workprint with Vertical Interval Time Code (VITC) and a VITC-to-LTC (longitudinal time code) converter, you can grab SMPTE numbers off still frames. In other words, you can find the exact frame of the hit, pause the picture and pop the frame number into Cue with a single keystroke, all in all a pretty nifty trick.

Once you've grabbed the hits, searched and found a perfect tempo, you can go on to add accelerandos and ritards, move cue points as the film edits change, make multiple versions of the cue, scale the total time of the cue (another handy trick, this one keeps all your temporal ratios intact but expands or compresses the total elapsed time of the cue), and retime within a specified area to make the key hits fall right on downbeats.

Once you are happy with your cue, you have to confront the vast array of decisions at the output stage of the program. If you actually write out your parts, you can print custom score paper with all the information from the Clicks Window displayed across the page, the times of all the clicks, the hits and their descriptions, tempo changes, and metric changes.

Having notated your piece, you can then conduct with Cue, providing either an audible click or a visual click with film-style visual streamers and punches right on the Mac screen. If you are working with sequencers, you can lock Cue up to MTC and drive the sequencer from Cue's MIDI Out. If you don't have a SMPTE-to-MTC converter and have only one Mac for sequencing, you can dump the beat map into an SBX-80 or equivalent (Garfield Master Beat, etc.) through a system exclusive dump.

Or, neatest of all, you can save the beat map as a MIDI file. A MIDI file is a new file format that allows information from different products to be exchanged. In this case, you can save your cue as a MIDI file that can then be read by any one of a few sequencing packages, notably Master Tracks Pro and Opcode's Sequencer. You can load the Cue file into the Conductor (or equivalent) track in the sequencer and all your tempo variations and metric variations will become part of the sequence. In the case of Master Tracks Pro, your hits will become markers so you can identify them (although they get moved to the top of the bar).

As a final handy little extra, there is a small MIDI event sequencer on board. One should by no means confuse this with a real sequencer package, but for an easy, efficient way to trigger sound effects or samples precisely, it is an invaluable little perk, and, of course, it means if you are just doing simple FX lay-ins, you can do the whole job without ever leaving Cue.

Cue summary

I have a hard time finding anything truly negative to say about Cue. It does what it's supposed to do in an elegant, straightforward manner. It allows for a variety of approaches to the process, from the traditional to the most high-tech. My only reservation would be a philosophical one. I feel the film scoring process really needs to be integrated.

In other words, the separation between the compositional process and the layout process seems to me artificial and restricting. Since Cue is a layout-only program and requires the electronic composer to use an external program for the composing, it only exacerbates this condition. But these are reservations to address in the next generation of software programming; in this "reel world," Cue is an exceptionally powerful and welcome tool.

Passport Designs Click Tracks

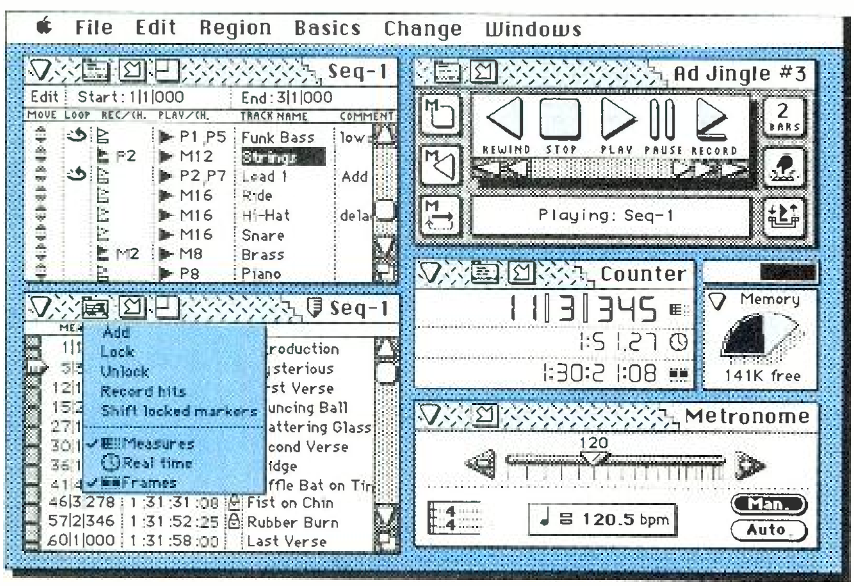
Click Tracks is Passport's entry into this market. It is similar to Cue, but its approach is more focused (or limited, depending on your perspective). Like Cue, the main idea is you enter your hits on a hit list. (These may be derived from an editor's cue sheet or a stopwatch utility while watching the picture.) Then you type in a range of tempos to scan or tap the Mac keyboard to enter a tempo sample.

The program then scans a range of tempos and displays the results with the seven best highlighted. Then you can change tempos or meters, see where hits fall (the Hit Map is excellent, by the way) and generally manipulate the data until you arrive at an acceptable layout.

The only thing missing from this stage of the process is a Retime command, something both Cue and Auricle have. This makes the process of perfectly lining up hits a bit tedious sometimes. Once the cue is laid out, you can either dump the beat map to a sync box using sys-ex dump or save it as a MIDI file to be loaded into Master Tracks Pro or Opcode's Sequencer, or anything else that reads them. The hits pop up as markers in Master Tracks Pro.

Click Tracks summary

My biggest reservation about the program is its lack of any real-time SMPTE features. But this is a philosophical difference of opinion rather than a criticism.



Sample screen, Mark of the Unicorn Performer software.

The designers were envisioning the product as a hit finder, not a hit grabber or a hit maker. The program is clearly best when used in conjunction with a compatible sequencer because it outputs no audible click on its own, so it is ill-suited as a conducting aid. But used with such a sequencer (particularly Passport's own Master Tracks Pro, which has Fit Time to make up for Click Tracks' lack of a Retime command), the program is an impressive and useful tool for circumventing the math and drudgery of laying out a musical cue.

Mark of the Unicorn Performer v. 2.31

Although not written exclusively for film work, Performer is a sequencing program that, in addition to being powerful as a straight sequencer, has a number of functions geared for the problems of working with picture.

For the purposes of this review, I will focus on the film application aspects. However, I should emphasize that the sequencing and editing features of Performer are as sophisticated as anything on the market. Though it does not read MIDI files like Master Tracks Pro or Opcode's Sequencer, Performer has enough SMPTE, MTC, tempo and metrical flexibility to function as a stand-alone film scoring tool, for those who work directly with picture and prefer not to work within the limitations of a layout structure.

Performer has three little extra goodies that merit its inclusion here: the Conductor track, the Markers window and the ability to lock to MIDI Time Code. The Conductor track is a special track devoted to tempo, metrical and key change data. For our purposes, the heart of the Conductor track is the Tempo Map. Here you can change tempos in a variety of ways: instantly (constant curve), in a linear, smooth manner (linear curve), logarithmically, exponentially or polynomially (a curve that starts gradually, is steep in the middle and ends gradually). Tempo changes are entered in beats per minute (BPM), and times can be displayed in real time or SMPTE time. Tempo changes are recorded onto the Conductor track. Further, using "markers," you can label all your hits and place them at the points where they occur in time and manipulate your tempo changes until the hits occur on the intended beats.

Alternately, on the Change Tempo window, you can change your start and end locations and let the program calculate the necessary tempo adjustments. This process, like Fit Time in Master Tracks Pro, allows you to conform your hits to picture relatively simply. This, taken in conjunction with the program's ability to lock to MTC, allows you to play to picture, mark your hits and then go back and fine-tune your tempo, to lock the hits to picture without losing the flow of the original performance.

Performer summary

My primary criticism is that Performer doesn't implement MIDI files, which seem to provide a spectacular breakthrough in terms of interfacing layout programs with sequencing packages. However, the guys at Mark Of The Unicorn say they plan to include MIDI files in future versions of the program, which should be available to owners at no extra cost (so send in those warranty cards). Also there is a little program floating around (PAN and possible other BBS) called Conformer that, yep, you guessed it, conforms MIDI files so they can be loaded into Performer.

Overall, I found the program to be exceptionally well written. It is among the strongest of the stand-alone sequencers I've seen for working with picture, given its MTC capabilities and its exceptionally sophisticated range of tempo change options.

Passport Designs Master Tracks Pro 2.1

Like Performer, Master Tracks Pro is an excellent stand-alone sequencer. Without going into the deep specifics of its sequencing functions, suffice it to say it does all of the things one has come to expect from a professional sequencing program and then some.

However, there are additional features that really make this program a gift for film composers. First of all, it reads MIDI files, which means you can load beat maps from Click Tracks or Cue into the Conductor track directly. As if that wasn't enough, it has a command called Fit Time that allows you to specify a beat where you want a marker to occur. If the marker (a hit to picture either entered directly or loaded via a MIDI file) is locked to a specific SMPTE location, the program will calculate the correct tempo manipulations to arrive at the correct beat.

In other words, you tell the computer where the hits are in SMPTE time and where you want them to fall musically, and the program takes care of the rest. Neat, easy and elegant (providing you don't ask for mammoth tempo corrections over short time spans, of course).

Version 3.0, which corrects some of the vagaries of marker placement when transferring from MIDI files, was shipped as of early August for the Mac and is scheduled to ship in September for the IBM and Atari.

Master Tracks Pro summary

Regardless of your approach to film work, Master Tracks Pro would probably suit your needs. If you work with a separate layout program, it has the necessary file compatibility to facilitate easy interaction with your companion program. If you work with just the sequencer, the Markers and Fit Time command allow you to conform your music to picture painlessly.

My only reservation was that version 2.1 did not lock to MTC, so it was extremely difficult to work directly to picture. However, version 3.0, I am told, will read MIDI time code.

IBM programs

While not blessed with the volume of programs written for the Mac, the IBM does have several excellent programs written for it. Also, with Yamaha's release of the C-1 (a laptop AT clone with some extra features), we can expect to see more good music software written for the IBM.

Among the programs already out, the most useful for the film composer are Roger Powell's Texture, by virtue only of its ease of use and ability to punch in while locked to MIDI Song Pointer; Voyetra's Sequencer Plus Mark 3, which has a Tempo track and comprehensive editing and sysex capabilities; and Robert Keller's 48 Track PC, which has so many film-oriented extras it merits a section of its own.

In addition, Passport is coming out with an IBM version of Master Tracks Pro 3.0 (which should be basically identical to the Mac version discussed earlier), and Auricle Control Systems is working on an IBM AT version of the Time Processor that does everything the Commodore version does, but on a larger scale and all at the same time(!). In other words, all of the displays can be on screen and active at once.

All in all, the IBM software market looks very promising, especially with Yamaha's endorsement of the format.

Robert Keller's 48 Track PC v.3.0

This program is unique in IBM land, in that it is actually designed for use by film composers. SMPTE is not just an added feature here; it is an integral part of the system. The key to the success of the program is its flexibility in terms of recording tempo changes. This sequencer allows the composer to play to picture without worrying too much about catching cuts and hits, then return, place markers at the SMPTE locations of the key hits and let the program perform the calculations necessary to place those hits exactly on musical beats.

As an alternate method, the pitch wheel of a synthesizer can be assigned to the tempo and ridden manually. All the tempo information and cue marks are stored on a separate track, somewhat like the conductor track on the Mac programs. The program does not lock directly to MTC, so it is, in effect, free-running after seeing a MIDI start command from a sync box, but its timing has proved amply accurate. The capability to post-fit music to picture simply by assigning a beat to a SMPTE location makes this an extremely easy-to-use and effective film scoring tool.

The only drawback I found in working with the program, a slightly awkward interface layout, has been corrected in the newest version, which should be available soon. The new interface has a SMPTE display right on the main screen and mnemonic commands in addition to the slower mouse commands and is generally laid out in a more natural way.

One footnote: It is strongly recommended you use this program with a hard disk, as it frequently refers back to the disk drive, which can be annoyingly slow on a floppy system.

Commodore 64 programs

Yes, the Commodore 64. In a classic example of the ascendancy of brains over brawn, or software programming over raw computing power, one of the most sophisticated and comprehensive film scoring tools on the market runs on the lowly C-64.

Lest you be tempted to skip over this section with an indulgent shake of the head and a dismissal of Commodore-based software as the realm of the hobbyist, let me assure you that this program would be first-rate on any machine. The fact that it does what it does on a 64K model is just a bonus. It's a convenience in a field in which an affordable dedicated click computer is a blessing.

Additional information

For more information about the software programs listed in this article, circle the appropriate number on the Rapid Facts Card in the back of this issue. Programs available for more than one system appear only once, in the section in which they were first mentioned; circle the appropriate number for all system versions.

Macintosh

Mark of the Unicorn—Performer (150)
Opcode Systems—Cue (151)
Passport Designs—Click Tracks (152)
Passport Designs—Master Tracks Pro (153)

IBM

Robert Keller—48 Track PC (154)
Roger Powell—Texture (155)
Voyetra—Sequencer Plus Mark 3 (156)

Commodore 64

Control Systems—Auricle (157)

Atari ST

C-Lab—Composer/Notator (158)
Hybrid Arts—SMPTETrack (159)
Steinberg/Jones—Pro 24 (160)

Control Systems
Auricle v. 1.8

The program's full name is Auricle: The Film Composer's Time Processor. Basically, the Time Processor is a click track generator/hit maker like Cue or Click Track (both discussed in the Mac section of this article). However, unlike these programs, which share a similar approach to the task at hand, the Time Processor has a unique, individual look and feel.

The interface between the user and the machine is a little window called the Auricle. All commands to the computer are typed into this box. There are no menus, mice, pull-downs or any of the other familiar means of communicating with the machine. Like any new interface, the Auricle takes a little bit of getting used to. However, once you become comfortable with it, it is a surprisingly natural and effective way of communicating.

The interface seems natural because it is patterned after that most-often-ignored model, human conversation. All of the commands are typed into the Auricle in (God forbid!) plain English.

If you want to see the Meter Map you type "meter map." If you want to add hits to the hit list, you type "add hits," and so on. Simple, huh? When the computer has to talk back, either to inform you of a mistake, ask a question, or confirm an entry, its responses pop up on the screen (again, in clear and civil English) and disappear as soon as you strike a key.

The effect is of a two-layered environment: on one level, the various maps and displays, and on an independent plane, the Auricle, in which you converse with the program. This layout allows the user to address any aspect of the program from any location rather than having to return to one screen and point and click through a succession of nested menus.

Users can further customize the working environment by assigning custom abbreviations or commands to the various functions via Hooks or Equates. Thus you could simply type "M" for Meter Map or "Mike Tyson" to add hits, and so on. All of the custom vocabulary can be saved on the program disk, so the machine will retain your vocabulary.

The basic procedure runs something like this: Let's assume that we are struggling with our cue for "The Return of the Psycho Fur Pig" once again. First, we would go to the add hits page, where hits are typed in. Hits can be entered in literally any format.

In addition to real time (mins./secs./hundredths) drop frame and non-drop SMPTE, there is a user-definable Optional Video format, which defaults to 25-frame EBU standard but can be set to anything. The system takes up to 45 hits. (If you require additional hits, you create another hit list.)

When your hits are entered, the system sorts them and the display indicates the bar and beat at which they occur and how early or late they fall (in hundredths of seconds and a graphic display). Now you can try different tempos. You can take a tempo sample by tapping the keyboard or simply type in the approximate tempo you want in either filmic clix or BPM.

The system does not accept fractional values, and, further, BPM entries are actually converted to the nearest Knudson (filmic notation, i.e., 12-0) click value. This can be inconvenient when you are working with a sequencer that functions in BPM as your master. However, you can trick the program into working with exact BPM values with the help of the Remeter and Retime commands if necessary.

Each time you alter the tempo, the system recalculates all the hits. Once you have arrived at the approximate tempo, you can increment and decrement the tempo by sprockets using the shifted minus and plus keys. You can also change the start of the cue with the "Set Start To" command.

This method is perhaps a little more cumbersome than Cue's Search Tempo, or Click Tracks' Scan, but it also allows you more quickly to assess the relative value of hits that are outside the acceptable hit tolerance and come up with a serviceable tempo.

So far, so easy. But, what if you can't come up with a click that comfortably catches all your hits? Several more options present themselves. Let's say there is a key hit that falls close to the downbeat of bar 24, but not close enough. You can use any of the three versions of the Retime command to shift the tempo of the piece subtly within a specified range so that the hit will occur exactly on the downbeat.

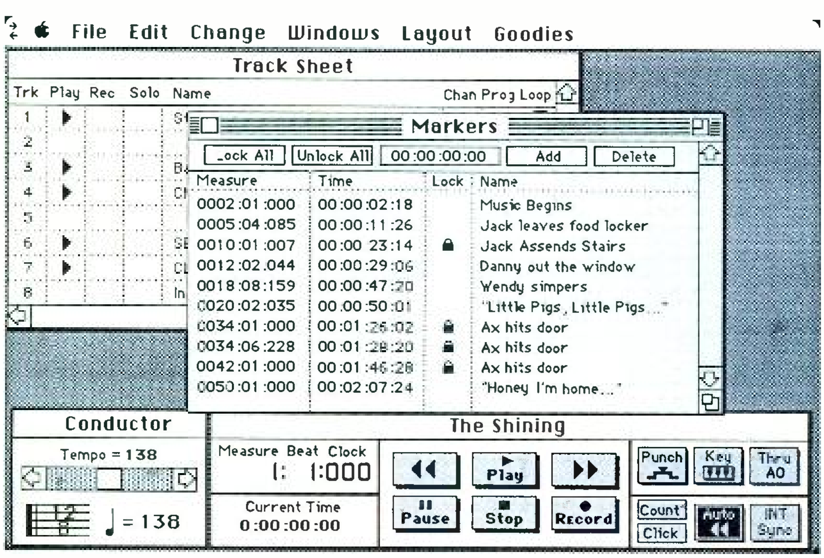
The versions of the command are Retime From Bar, Retime Inside Bars and Retime Over Bars. In the first two examples, the program simply calculates a new tempo for the specified range, i.e., bars 21 to 24, to allow the hit to land at the downbeat of bar 24. In the third example, Retime Over Bars, the program calculates the necessary tempo changes without altering any ritards or accelerandos (made with the Slope command) in the intervening measures.

On a more musical note, the program has the usual remetering commands to allow you to create any time signatures you require. There is an additional, extremely helpful command called Skip, which, when used in conjunction with the Remeter command, allows you to construct a meter map quickly with complex time signatures. For example, a repeating pattern of two 7/8 bars followed by a 4/4 bar.

Having constructed a beat map, a meter map and a list of hits, you can use the program as a click generator to conduct an orchestra, complete with warning clicks and streamers, or use the click to generate a beat map for an SBX 80 or equivalent. Or, if you have the MIDI option, the Time Processor can function as a master to drive whatever sequencer you are using. The program does not sync to SMPTE or MTC, but it can be set to start from a MIDI start command from any SMPTE-to-MIDI converter or any MIDI sync device such as the JL Cooper PPS-1 or equivalent.

This is, of course, not actual sync to tape, as the program is free-running once started, but it has proven to be extremely accurate so as long as the tape machine speed is correct.

Sample screen, Control Systems Auricle software.



Sample screen, Master Tracks Pro software.

Auricle summary

Overall, I've found the program to be exceptionally powerful and easy to use, a rare combination. It was clearly written from a musical rather than a "techno" point of view. Its greatest asset is that it runs on an extremely inexpensive machine, which makes it feasible to work on the musical and technical aspects of a cue simultaneously.

Its primary drawback is the lack of any SMPTE lock capacity. If the machine could lock to SMPTE or MIDI time code, it could function as a transparent link in the creative chain between the musician and the picture. The program's authors, brothers Richard and Ron Grant (a computer programmer and a composer respectively), are now completing the IBM version of the program, which promises to take full advantage of the larger machine's computing power. A beta release of this program is discussed brief-ly in the IBM section of this article.

Surprisingly enough, there are no other Commodore 64 programs that address the needs of film composers.

Amiga programs

The big brother in the Commodore family has a frustrating history in the music business, a history comprised almost exclusively of "if only." I too have looked longingly at this multitasking, potentially thrilling machine only to be left with vaporware dreams. When you think about multitasking long enough, especially as someone who works with picture, you find yourself salivating at the possibilities.

Already there are video programs that work well with the Amiga. Imagine adding a click/layout program and a sequencer, all RAM-resident and, in fact, all running at once(!), all interrelated and communicating with each other(!!), all in real time(!!!).

(Oh well, if only film editors could tap their feet in time...if only ice cream didn't get freezer burn....)

Atari ST programs

In marked contrast with the Amiga, the Atari software story has been one of "just wait." Anyone who criticizes an absence in the Atari market runs the risk of watching his complaints proved false by the time his words reach print. So, while I will say that as of this writing there are no programs specifically targeted at film scoring work, I do so with mingled trepidation and optimism; with confidence that such software will arrive.

Already, several powerful sequencing packages are on the market for the Atari. Notable among the group are Steinberg/Jones's Pro 24, Composer/Notator from C-Lab, Master Tracks Pro from Passport and SMPTETrack from Hybrid Arts. Pro 24 is a program that has dominated the European market for a while and is now becoming popular in America. It is one of the most comprehensive sequencers around. Of interest for the film composer's purposes is the Master track for recording tempo and meter changes, MIDI files capabilities (though perhaps not immediately useful, I'm sure it will be) and best of all, locking to SMPTE, if you have a SMP-24 or a Time Lock.

The C-Lab package also has extensive editing capabilities of data and tempos, and also provides score printing and exceptional ease of use.

SMPTETrack is another program with extremely comprehensive SMPTE lock capabilities and all the compositional freedom that it provides.

Finally, Master Tracks Pro, discussed in the Mac section, has a version for the Atari. Version 3.0, new and improved and just shipped for the Mac, is supposed to be released for the Atari by the time you read this and should be top-of-the-line stuff.

For the Mac and the IBM, there are several good programs. For the others, help is, we trust, on the way. Your choice in software really depends on your style of composition. You may want a sophisticated layout program and a simple sequencer, (or no sequencer at all if you plan to conduct from a score). Or a sophisticated sequencer that can conform your playing to picture, may be more appropriate.

The main thing is to work in a software environment that reduces, rather than increases, the amount of tedium and non-musical complications inherent in the process. RE/P

Facility Profile: NBC's Studio 1 in Transition

By Ron Estes

The NBC-Burbank studios were constructed in the early 1950s for live television as it was known then. The new control rooms will have state-of-the-art equipment, making them the most modern TV broadcast facilities anywhere.

The "Tonight Show" will move into all new audio/video control rooms by October 1988. The existing Studio 1 facility remains, and a new building housing the audience lobby, technical areas and offices was constructed where the old lobby once stood. The new control rooms will have state-of-the-art equipment, making them possibly the most modern television broadcast facilities anywhere.

Ron Estes is the audio mixer for the Tonight Show at NBC studios, Burbank, CA.

The NBC-Burbank studios were constructed in the early 1950s for live television as it was known then. Each studio has a 40-foot-high grid laced with counterweighted light and "carpenter" pipes. These pipes are divided into three sections across the studio for easy access. Studios 1 and 3 were originally designed for black-and-white and were converted to color in the late '50s.

The air-conditioning system for the entire Burbank facility is enormous; in the early days of color, the cameras were not nearly as sensitive as they are today and required much more light—sometimes as much as 400 foot-candles. Studios 1, 2, 3 and 4 are all somewhat similar except that Studio 1 has permanent audience seating that is still used today.

Photo 1. Close-up of the old console, showing assignment push-buttons under the armrest. Notice the rotary submasters and masters immediately above the arm rest. Next are the 22 vertical faders with 10 subfader groups above. The equalizers for the vertical faders are at the top.

Early audio consoles were designs that worked well in ratio. The consoles had 12 rotary faders that could be assigned to three submasters, feeding in turn into a board grand-master. (See Photo 1.) A Nemo (remote) master that had the same priority as the board grand-master made it possible to fade the entire show audio to "black" and come up on a commercial without resetting many different controls. No EQ or gain-reduction devices were in use at that time. The control room had a total of 24 mic lines coming in from the studio.

These consoles proved to be inadequate, as more and more live music was being used on shows such as "The Dina Shore Chevy Show" and "The Dean Martin Show," both of which used rather large orchestras. At first, RCA OP-6 (4-in/1-out) sub-mixers were stacked on top of the existing audio console to gain more inputs. Early Altec tube compressors were purchased by Bill Cole, mixer for "The Dinah Shore Show," for better control of high audio levels. It was time to rethink the entire audio console requirements at NBC. (See Figure 1.)

New design

Russ Nies, an audio/video design engineer, was assigned to find out just what the audio mixers at NBC really wanted in a new audio console. He drew up plans, made mockups, revised plans

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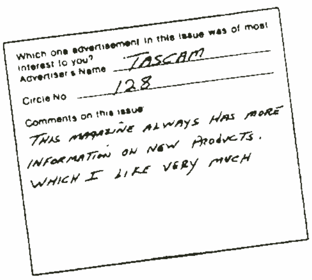
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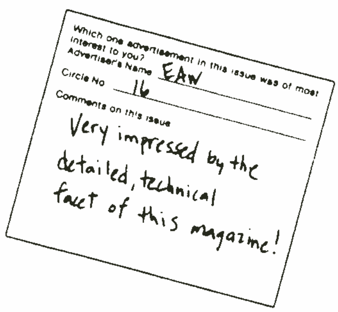
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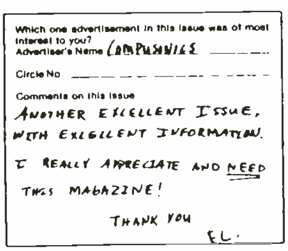
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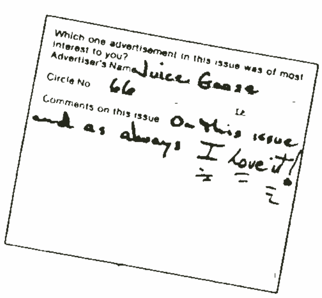
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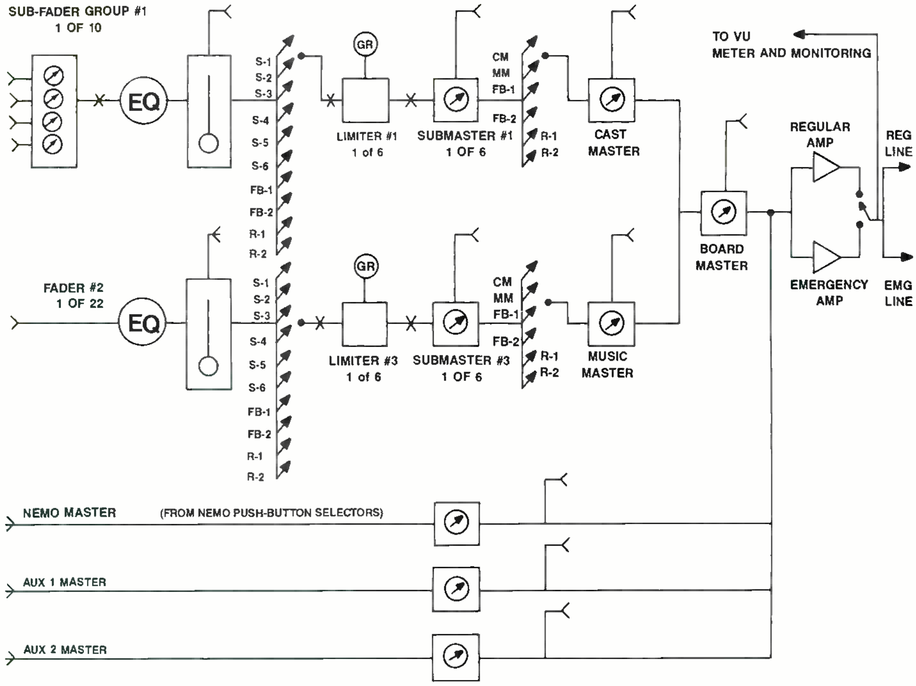


Figure 1. *Simplified diagram of the old audio console built by NBC in 1965.*

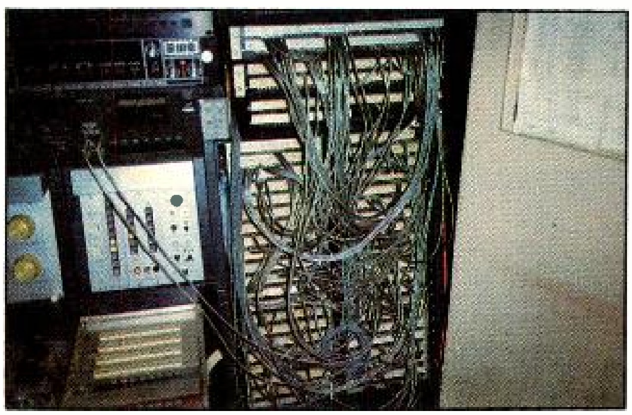


Photo 2. *Main patchbay for the old NBC console. The original bay had 960 jacks, but 384 more were added for the stereo expansion. More than 200 patch cords are used.*

Photo 3. *Present audio mix area, with old NBC console in center and the auxiliary Soundcraft 800B to the right.*

and finally built a TV audio mixing console from the ground up that was different from anything available at that time. His design became a series of 4-in/1-out subfader groups. (Remember the RCA OP-6.)

Each one of the 10 subfader group outputs normalled through the patch panel to a Langevin EQ 251-A passive equalizer and a Langevin AM-16 booster amplifier to make up for the EQ loss. These fed into vertical faders that were in essence a

For ease of operation, the console is configured in an L shape with the first 16 faders on the left side and the remainder in front of the mixer.

master for the subfader groups. Each of the pre-amps feeding the subfader group could be bypassed so that a higher input level (−22dB) could be fed into the group. Each of the subfader and vertical fader inputs had separate input mults so that prefader feeds could be sent to the PA system and stage monitors. Each of the vertical faders also had separate isolated outputs for postfader feeds that would follow the fader.

There were 22 vertical faders, with associated 4-in/1-out sub-fader groups above them. Ten others were single inputs for booms and hand mics, and two were for reverb return. The console design had 10 mixing buses, all mixing being done at the −22dB level. The first six buses again were routed through the patch panel and returned to six Teletronix LA-2A tube limiters.

The outputs of these limiters were routed again through the patch panel to six rotary submasters located just below the vertical faders. Each of the six rotary submasters (again with a separate prefader and postfader patches) could be assigned to the Cast Master and/or Music Master. This brought the entire show down to two rotary faders.

The Cast and Music Masters, like the submasters above, had prefader and postfader patch points and fed into the Board Master. There were four masters; the Board Master controlled all inputs to the console except those that were patched to the other three masters. Two of these were designated as aux inputs and could be used as independent masters.

The Nemo master was fed from a "job-switcher" that had various videotape machines, film chains and other remotes assigned. This relay switcher could be controlled by the video switcher when in the "automatic" mode, or by push-buttons on the audio console when in the "manual" mode. Three separate Channel Amps with VU meters and input selectors made it possible to select the input or output of any of the 12 rotary masters.

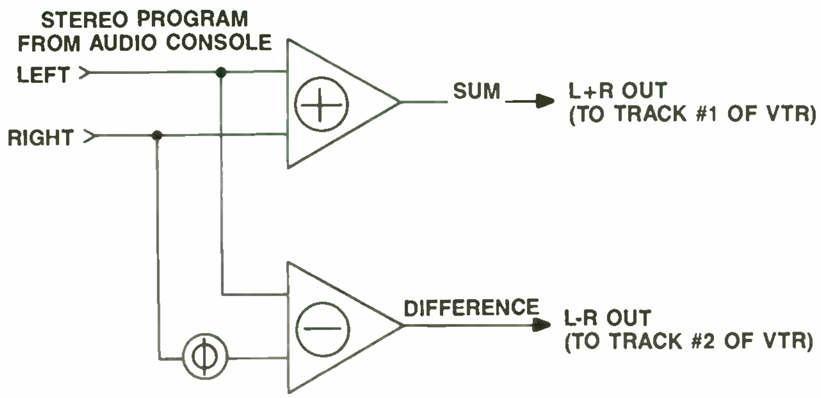


Figure 2. Encoding (sum and difference) matrix, an early "Tonight Show" recording method.

Looking back over this console, we find a 55x22x6x2x1 design, all in about 42 inches of width—an unusual and versatile blend of rotary and vertical technology in a limited space. The original patch panel contained 960 TRS TT jacks and is still in

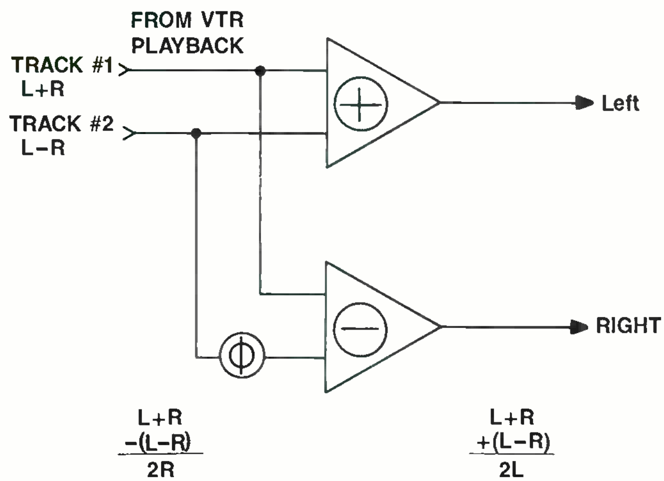
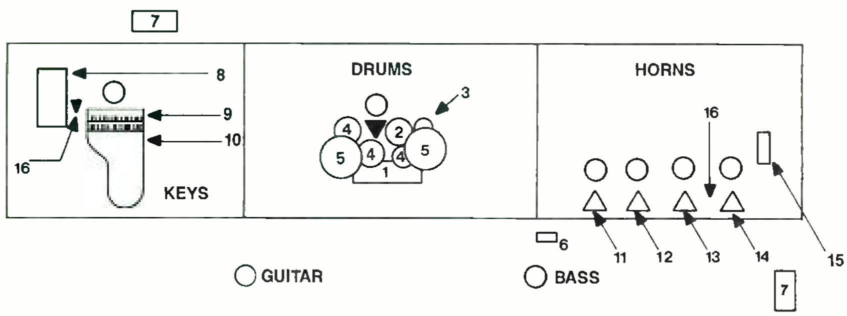


Figure 3. Decoding (sum and difference) matrix, used to decode early "Best of Carson" reruns.



1. Kick: Sennheiser 421.
2. Snare: (top) Altec tube; (bottom) Countryman ISO Max.
3. High-hat: Scheops CMC 5/MK 4.
4. Rack and floor toms: Scheops CMC 5/MK 4.
5. Overheads: Milab DC 96s.
6. Bass: (direct) Audiotek DB1 with Jensen transformer.
7. Guitar amps: Milab DC 96s.
8. Akai sampler: (direct) Audiotek DB1 with Jensen transformer.
9. Piano: (high) Scheops BLM-3 plate mic; (low) Crown PZM.
10. DX-7: (direct) Audiotek DB1 with Jensen transformer.
11. Baritone/alto sax: Milab DC 96.
12. Tenor sax: Milab DC 96.
13. Tenor sax: Milab DC 96.
14. Trumpet: RCA 77 DX.
15. EVI: (direct) Audiotek DB1 with Jensen transformer.
16. Vocals: AKG 535s.

Figure 4. Sample setup for Billy and the Beaters, a guest band.

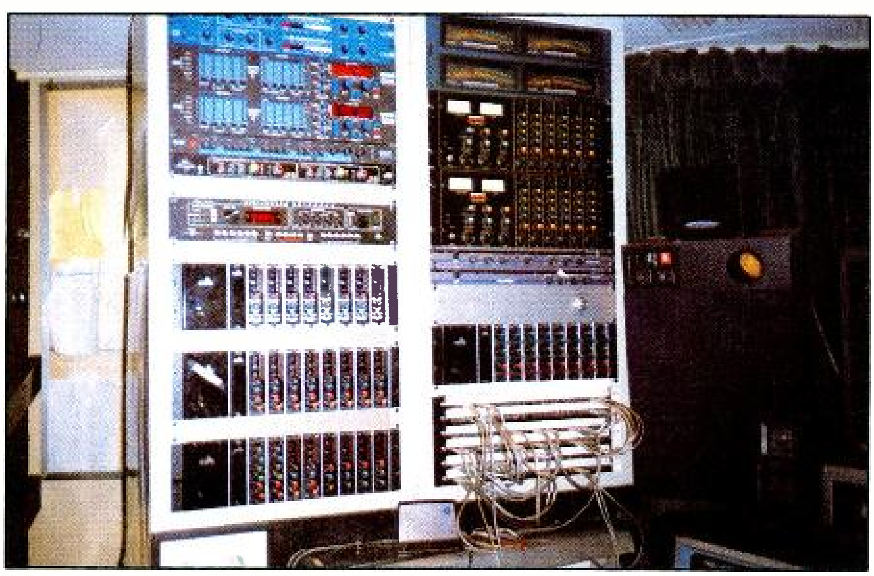


Photo 4. Two racks of outboard equipment and auxiliary patch panels adjacent to main console.

use today. (See Photo 2.) Five of these consoles were built by Russ and his construction group for NBC's larger studios. The last one to be replaced is the console in Studio 1, "The Tonight Show" studio.

Because of the extreme flexibility of these consoles and the extraordinary way they were constructed, some have lasted more than 25 years. The fader assignment buttons were recessed in the armrest/script area of the console with heavy plastic covers. It was not at all uncommon to stand on the console to change light bulbs in the fixtures overhead. All modules were removable, making for easy troubleshooting. Modules were interchangeable with those in other studios.

An extender frame and test-fixture were built for checks and alignment. Even the knobs for the rotary faders were machined in the NBC mechanical shop from blocks of solid aluminum. Each console contained more than 100 Langevin AM-16 preamp/booster amplifiers and five Langevin AM-17 program amplifiers. Separate regular and emergency program lines fed the transmission/videotape areas. Three pairs of power supplies provided redundancy for all amplifiers. The console was "bulletproof" and would not clip until +37dBm output!

Stereo from a mono console

With some idea now of how the hardware works, let's take a look at how a stereo program such as "The Tonight Show" has been mixed on this old mono console with no pan pots.

As mentioned, some of the 55 simultaneous inputs to the console are patched to the 10 4-in/1-out subfader groups and others directly to the remaining 10 vertical faders. Orchestra sections of brass, reeds, keyboards, etc., are premixed on the subfader groups—with the equalizer, and its associated vertical fader becomes the master for this section. Any of the 22 vertical faders can be assigned to any of the six submasters, and these submasters can be assigned to the Cast and/or Music Masters.

Here is where the stereo split is made. The Cast and Music Masters became (by definition) left and right outputs of the console, each with its own program amp, VU metering and monitoring. These two masters "sum" to a mono mix at the Board Master.

The six submasters are renamed as follows:

#1—Cast/Left.
#2—Cast/Center.
#3—Cast/Right.
#4—Music/Left.
#5—Music/Center.
#6—Music/Right.

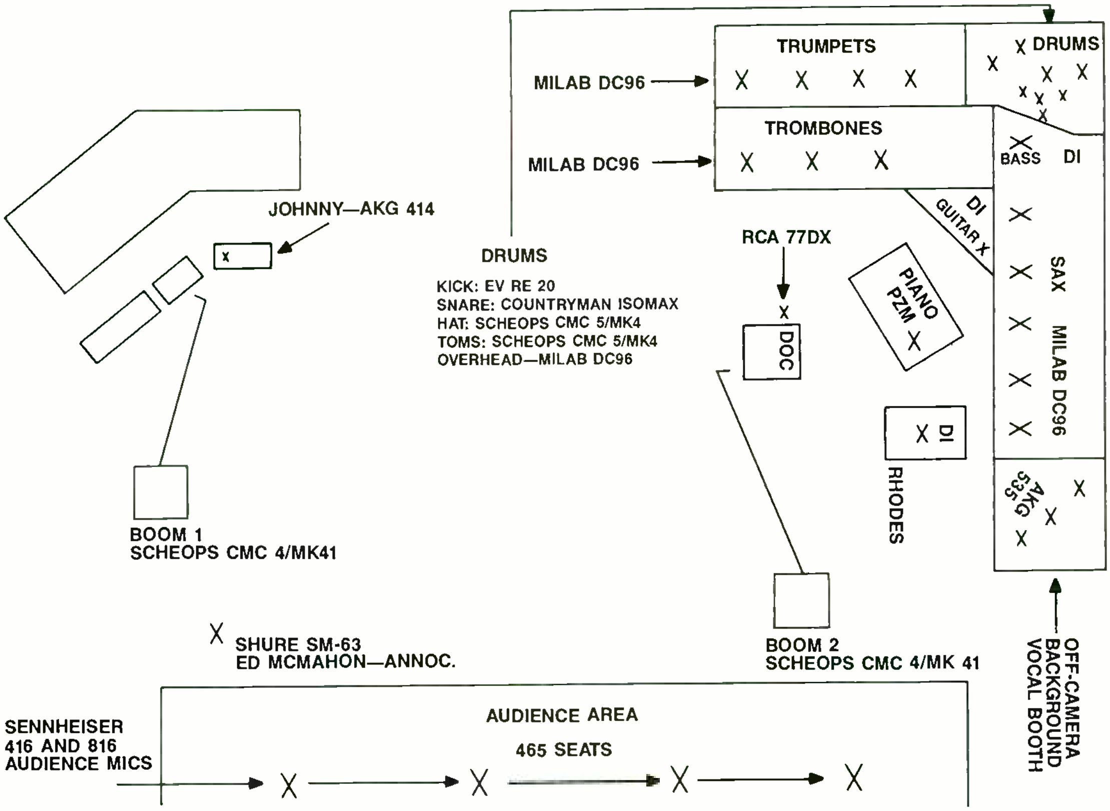


Figure 5. *The standard "Tonight Show" setup.*

The #1 submaster is assigned to the Cast Master (left channel) only. The #2 submaster is assigned to both Cast and Music (Cast/Center). The #3 submaster is assigned to the Music Master (right channel) only. Subs 4, 5 and 6 are assigned in the same was as 1, 2, and 3. Now there is the option of assigning any of the vertical faders hard left, center or hard right while keeping them in the Cast or Music domain. Dialogue and applause are assigned to 1, 2 or 3, while music is assigned to 4, 5 or 6. This configuration solved the placement problem without pan pots or an auxiliary console.

To help minimize the obvious extremes in "panning," the brass section, which is assigned hard left, has its reverb return assigned hard right; and the reed section, which is assigned hard right, has its reverb return assigned hard left. Any instrument assigned music/center has its reverb return center also. The different phase characteristics of the dual channel AKG BX 20 spring chamber make for interesting effects. Drums, piano, guitar and bass are all center (subs #6, #4 and #5 went to the left and right, in this particular case).

Perhaps one of the more unusual approaches to panning was achieved by patching a microphone to the first input (A) of two adjacent subfader groups, whose vertical faders are assigned hard left and hard right. Another mic was then patched into the second (B) input of both groups and so on through input four (D). By setting the vertical faders to the same gain and adjusting the relative gains of each pair of inputs, a sort of "variable pan pot" was created. This arrangement was used mainly with background singers to position them slightly off-center.

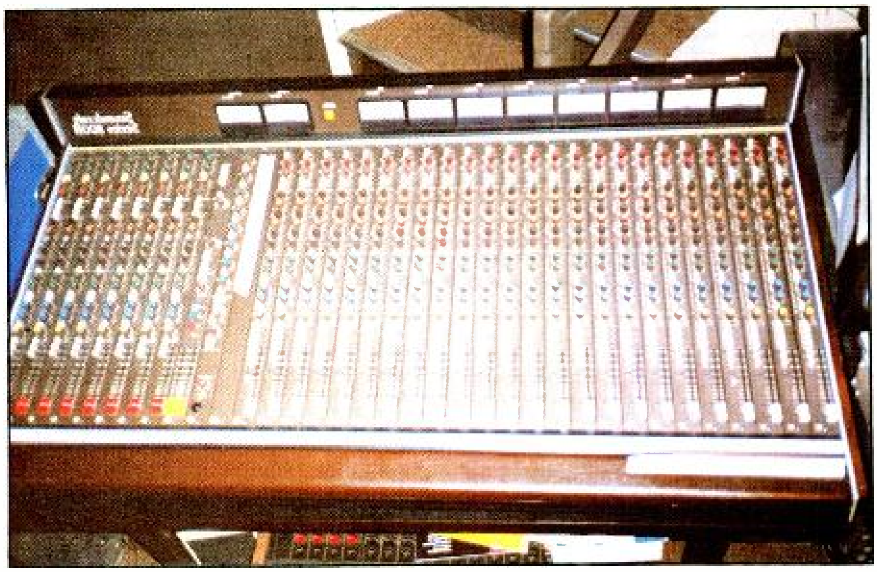


Photo 5. *The Soundcraft 800B auxiliary console.*

"The Tonight Show" was mixed in stereo on this console using these techniques for some time. For the first few years, this stereo output was heard only on the monitors in the audio booth because the show was still being recorded on 2-inch quad videotape with only one good audio track.

Later, as the show moved to 1-inch type C videotape with two good audio tracks, both channels were recorded—but not as left and right. Stereo television was still only being talked about and no system had been approved by the FCC. The networks had only a one-channel delivery system to the affiliates, and any stereo shows had to be "summed" to mono. It

Equipment List

Mixers

1 NBC custom mono audio console, 55x22x6x2x1 (built in 1965).
1 Solid State Logic 6048 TV production console, 80 mono/8 stereo x 32 buses x 3 x 1 stereo.
2 Solid State Logic submixers, 18x4 stereo.
1 Soundcraft 800B auxiliary mixing console, 24x8x2.
1 Soundcraft 400B monitor mixing console.
1 Studer 962 NAB stereo cart machine mixer, 16x2.
2 Yamaha M406 stereo submixers, 6x2.
1 Yamaha 916 stereo mixer, 16x4.
1 Yamaha M1532 PA mixer, 32x4.
1 Yamaha PM 430 sound effects mixer.

Outboard gear

12 dbx 903 compressor/limiters.
10 dbx 904 noise gates.
10 dbx 905 parametric equalizers.
1 Eventide H949 Harmonizer.
1 EXR EX III Exciter.
1 EXR EX Exciter.
3 Aphex Compellers.
4 Teletronix LA 2A tube limiters.
2 Lexicon 95 Prime Time II DDLs.
2 Effectron II ADM 1024 DDLs.

Reverb devices

1 AMS RMX 16 digital reverb.
1 Lexicon PCM 60 digital reverb.
1 AKG BX 20 E1 reverb.
1 Yamaha REV-5 digital reverb.
1 Yamaha REV-7 digital reverb.

Tape machines

1 Otari MTR 90, 24-track with autolocator.
2 Ampex ATR 104 4-tracks.
1 Studer A710 cassette recorder.
15 ITC stereo NAB cart machines.

Other equipment

1 Dolby Cat. 43A noise gate.
1 Skotel TCG-80 time code generator.

Photo 6*. The new SSL 600 console for Studio 1.*

was unrealistic to ask the videotape department to mix the two discrete tracks together to avoid broadcasting the left channel only.

Compatibility was maintained, however, by encoding a left-plus-right signal on channel 1 of the videotape and a left-minus-right signal on channel 2 of the videotape. Channel 1 (left-plus-right) was sent out to all the affiliates and broadcast in the normal way. Channel 2 (left-minus-right) contained the "difference" signal that, when decoded, would return the program to full stereo. A matrix encode/decode box was designed and constructed by John Strain of the maintenance department to accomplish this task. (See Figure 2.)

"The Tonight Show" was recorded in this format for more than two years. None of

None of these shows ever ran in stereo the first time they aired.

these shows ever ran in stereo the first time they aired. They were later decoded, however, and were aired in stereo as "Best of Carson" reruns some years after being taped.

Stereo commitment

NBC made a commitment to be first with network MTS stereo. During the start-up period, many decisions had to be made and much new equipment installed. All of the studios in Burbank were slated for new audio/video control rooms. Studio 11 was constructed for daytime dramas. The studio itself is 100'x180' and houses the first of the new generation of audio consoles—the Solid State Logic 6000 video production series.

Later, a new control room building was constructed for studios 2 and 4. They also received new SSL consoles somewhat larger than the one in Studio 11. About this time, the FCC approved the MTS format for stereo broadcasting. KTLA, Channel 5, a Los Angeles independent, began stereo broadcasting shortly afterward. WNBC in New York, the NBC owned-and-operated station, had a "hand-built" stereo modulator and was testing stereo transmission in the early morning hours.

All this time "The Tonight Show" was being recorded in the sum/difference format in Burbank but not being broadcast in stereo. Word leaked out that ABC was planning to broadcast the opening ceremonies of the 1984 Olympic Games being held in Los Angeles on its owned and operated station, KABC.

NBC, not wanting to be upstaged, decided to run one of "The Tonight Shows" that was being mixed in stereo on WNBC in New York. The program was fed out of "The Tonight Show" studio through a "patched-up" satellite system to New York, where it aired on WNBC. Doc Severinsen made a special announcement at the beginning of the program about this "first" network broadcast of MTS stereo. On July 26, 1984, NBC beat ABC with stereo by two days.

It was soon evident that to continue mixing "The Tonight Show" on the old console would compromise the stereo. New control rooms were to be constructed for the studio 1 and 3 complex, but that was several years down the road. The interim solution was to add another mixing console to supplement the existing console. (See Photo 3.) Extensive outboard gear such as compressors, gates, better equalizers, digital delays, pitch shifters and more reverb devices were installed in racks adjacent to the main console. (See Photo 4.) The console selected was a Soundcraft 800 B, 24x8x2. (See Photo 5.)

All inputs, outputs and insert points were brought out to added patch panels installed in the racks that house the outboard equipment. Timing of all this new equipment was to coincide with the first broadcast, July 23, 1985, of "The Tonight Show" in full-time stereo. Maintaining tolerances in the sum-and-difference format proved in tests to be too difficult for day-to-day operations, and discrete left and right recording methods were adapted. "Best of Carson" reruns, however, were regularly decoded and aired in stereo for the first time.

The new stereo console

The new control rooms for studios 1 and 3 occupy a recently completed building addition fronting on Alameda Street. This building stretches the entire length of the two studios and was designed exclusively for control rooms and a new studio 1 lobby and audience holding area. The second floor houses "The Tonight Show" offices. All new equipment, including cameras, switchers, monitors and audio equipment, is being installed in these control areas.

On July 26, 1984, NBC beat ABC with stereo by two days.

The new audio console is again the SSL 6000 video production console equipped with the Total Recall computer. Many of the studios at NBC have rotating shows and different people are assigned to mix, as needed. With Total Recall it is possible to take an electronic "snapshot" of all settings of the console modules (fader assignments, reverb send levels, EQ settings, etc.) and record them on floppy disk.

When the same show returns to the studio, this information is read back from the disk and gives the mixer a color graphic representation on a video monitor of each fader module and where every control was set when the data were recorded. This computer notepad simplifies setup of ongoing shows that move from studio to studio.

For ease of operation, the console is configured in an "L" shape, with the first 16 faders on the left side and the remainder in front of the mixer. (See Photo 6.) The monitor/computer/output section is followed by eight stereo faders.

In addition, all of the short (monitor) faders directly above the long faders have external pre-amps so that they can be used for submixing. Two SSL mini-mixers, consisting of 18 inputs and four stereo outputs, occupy the left half of the extended left-wing area. The main console has 40 fader inputs with 4-band EQ, compression and gating on each input; 40 small fader inputs for premixing; and 36 inputs to the two mini-mixers. Most of the recently purchased peripheral equipment used in the old control room (extra gates, EQ, compressors, DDL's, etc.) is also being installed.

The audio control room is approximately 25'x23'. Acoustical design of all the new NBC control rooms is by Chips Davis, and he recommended and supervised the construction of the rooms and air conditioning. The console sits in front of four large

Photo 7. *Music room adjacent to main audio control room, with 15 NAB stereo cart machines and a Studer mixer.*

Tonight Show Microphone List

Trumpets, trombones and saxes: Milab DC 96.
Drums:
Kick: EV RE-20.
Snare: Countryman IsoMax C.
High-hat: Scheops CMC 5/MK 4
Rack toms (2): Scheops CMC 5/MK 4.
Floor tom: Scheops CMC 5/MK 4.
Overhead (2): Milab DC 96.
Piano: Crown cardioid PZM.
Rhodes: Direct, Audiotek DB1 with Jensen transformer.
Bass: Direct, Audiotek DB1 with Jensen transformer.
Guitar: Direct, Audiotek DB1 with Jensen transformer.
Doc Severinsen: RCA 77 DX.
Johnny Carson: AKG 414 EB P48.
Ed McMahon: Shure SM 63.
Boom mics (2): Scheops CMC 4/MK 41.
Vocal mics: Sennheiser 431.
Additional microphones: Neumann KM 88; Sennheiser 416, 421, 441 and 816; AKG 45a; RCA 77 D; Scheops assorted Collette series and plate mics; Milab DC 96 and DC 63; Altec M 51 tube mic; Countryman IsoMax C and B.

video monitors and two UREI 813 C speaker systems. Boulder 500 power amplifiers drive the main speakers. Two small Auritone cubes are mounted on the meter bridge for TV-type monitoring. A separate switching bus of the Grass Valley 300 switcher is installed in the audio console enabling the mixer to "punch up" any of the video feeds on a separate video monitor.

Behind the console are five low racks, which contain the patch panel and some of the outboard equipment. All the SSL consoles have their internal patch points integrated into the main patch area and are not above the producer's desk. The three racks of patch rows contain 3,360 mini TRS patch points—a rather large number for a broadcast control room!

Adjacent to the audio control room is another room designated the Music Room. (See Photo 7.) This room has 15 stereo NAB cart machines and a Studer stereo mixer. Tie lines are available for the Otari MTR-90 24-track tape machines, which can be used for recording split-outs from the audio console. A small announce booth is in the corner for voice-over recordings.

Eight broadcast service panels are mounted around the studio itself—two at either end of the studio and three on each of the two side walls. Each one of these panels has 24 microphone inputs as well as numerous high-level trunks, power amplifier lines to drive speakers, intercommunications and video tie-trunks. Twenty additional mic lines terminate at various locations in the grid for boom-drops and applause mics. More than 225 mic lines appear on the control room patch panel.

Built into the patch panel is a custom-made Jensen transformer microphone-splitting system. Each of the 48 inputs has a loop-through to pass phantom power from the audio console. Each has two secondary outputs—one designated for feeds to the PA console and the other for feeds to the foldback or monitor console. There is also a +4 input, which is padded to mic level for splitting of line level feeds.

A total of 48 tie trunks run from the audio console patch panel to the PA console at the rear of the studio. Two loop-through connectors allow these lines to appear at panels on either side of the studio if a different PA console location is needed. The same is true for the 48 trunks to the foldback console. They normally terminate in front of the audience area where the foldback mixer is located; but by using the loop-through connectors, the trunks can appear on either side wall as needed.

At this time, "The Tonight Show" is recorded on four 1-inch tape machines. There are two sets of stereo lines leaving the studio—regular and emergency. Three of the tape machines record from the regular line and one records from the emergency line. The regular line is fed a composite stereo feed. The emergency line gets band only until after Ed McMahon has introduced the guests and says, "Heeeere's Johnny." At that time, the emergency line is switched from the band-only feed to the composite feed. If one of the scheduled guests does not appear on the show, this band music is edited into the spot where Ed billboards the name of the "bumped" guest. Since all four tape machines have the same time code, it makes for a simple edit.

The transition

NBC will carry the 1988 Olympic Games from Seoul, South Korea, this September. "The Tonight Show" will be pre-empted, and the move into the new control room complex is to be completed at that time. The first show from the new area will be Monday, Oct. 3. Plans are now being completed for a smooth transition to the new facilities.

In a way, I'm really going to miss the old control rooms and equipment. They served faithfully for many years. When I recently visited NBC in New York, I could feel the ghosts of the past. Here, as well as there, TV history was made. And I am still in awe of the enormously creative pioneering efforts made by so many during those early years of live television.

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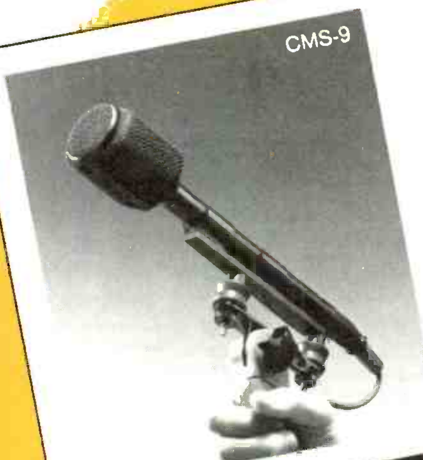

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Circle (28) on Rapid Facts Card

Flim & the BB's in Concert

By David Scheirman

Precision microphones and dual-channel signal analysis help make live sound "as good as the record."

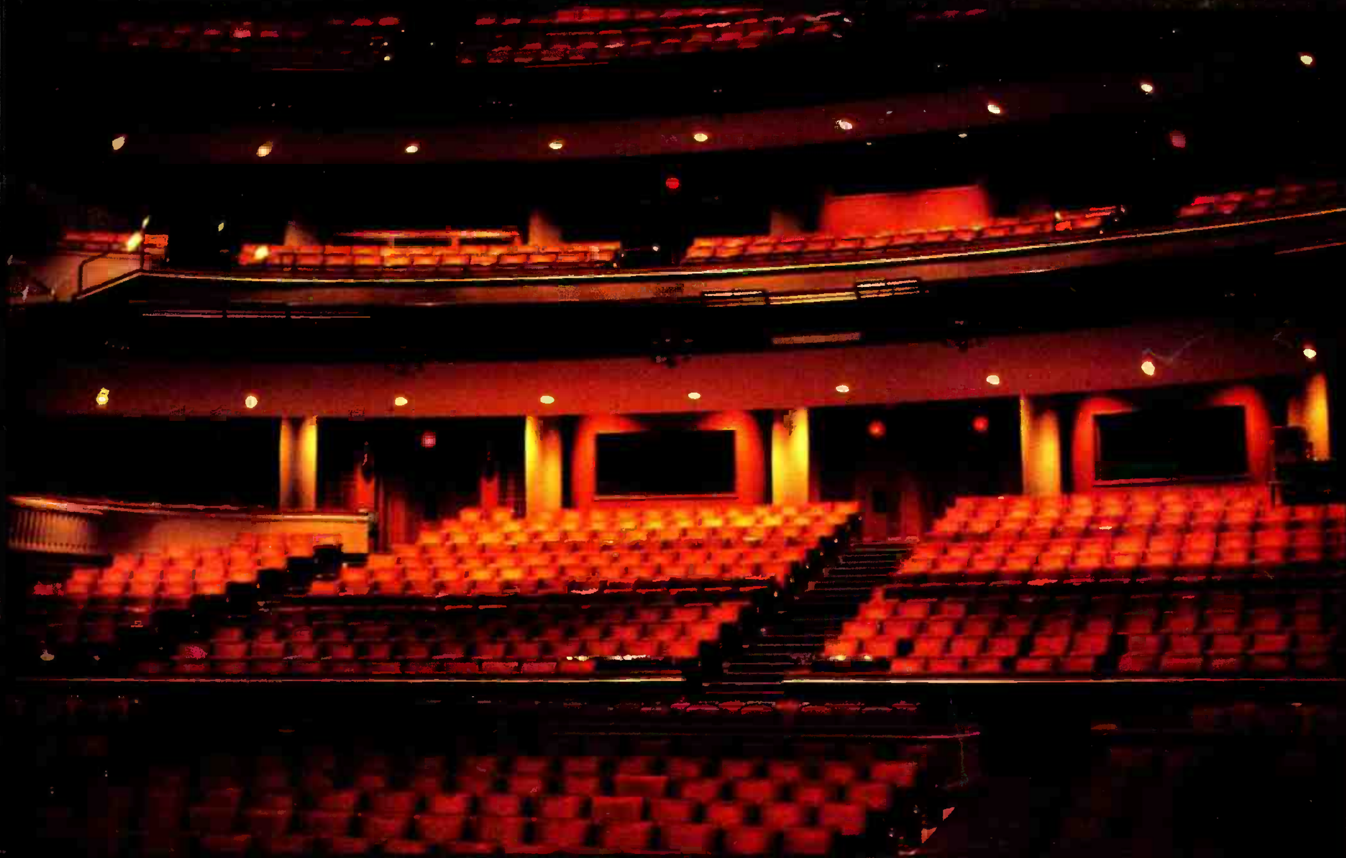


***Figure 2.** The Ordway's Main Hall can accommodate 1,815 persons, and features several high balconies.*

In the early 1970s, a group of musical friends regularly met to jam at a small nightclub in Minneapolis. At the same time, 3M technicians were working on the research and development of that company's first digital audio recorder. Sometimes, the technicians would invite the musicians into the plant to play live music through the newly developed audio circuits.

In 1978, a collaborative effort resulted in a direct-to-digital recording, using a prototype machine. Then, in 1982, producer Tom Jung invited the group to do a compact disc project for the newly formed DMP label (Digital Music Products). Released in 1983 and entitled "Tricycle," the resulting CD was attributed to a mysterious group, Flim & the BB's. It received *Audio* magazine's award for jazz compact disc of the year.

Because of the group's use of musical passages that offered an exceptionally wide dynamic range, the disc became a standard favorite of many sound system technicians and was often used to demonstrate stereo equipment by in-store

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

Figure 1. Exterior view of the Ordway Music Theatre, St. Paul, MN, site of the Flim & The BB's concert.

salespeople. It was also used extensively at pro sound industry events.

With the CD earning so much praise, many people began to wonder about Flim & the BB's. Were they actually a band? Where would they be appearing in concert? Stumbling across a Flim & the BB's concert was no easy task. In the six years that have passed since the recording of "Tricycle," the group's live appearances can almost be counted on one hand.

"We all have our own projects going, and much of our work takes place in the recording studio," says drummer Bill Berg (one of the BB's, along with keyboardist Billy Barber, plus bassist Jimmy "Flim" Johnson and saxophonist Dick Oatts). "Getting us all together in one place at one time is quite a feat. I guess we've averaged about one concert a year, you could say!" [Editor's note: The group has scheduled a rare concert appearance in Los Angeles to coincide with the fall AES convention.]

Live at the Ordway

The most recent live event took place in June at the Ordway Music Theatre in St. Paul, MN. (See Figure 1.) Because the group members had met and lived in St. Paul in the past, the idea of a hometown concert was appealing, particularly in light of the fact that the group's latest recording ("Neon") featured three of the musicians' fathers. A cut titled "Fathers And Sons" gives us a clue. Bill Barber Sr. played acoustic piano, Cliff Johnson (Flim's father) played acoustic bass and Jack Oatts played alto saxophone. "The Dads" agreed to participate in the live concert at the Ordway Music Theatre. Regularly used by the Minnesota Orchestra and the St. Paul Chamber Orchestra, the Ordway's main hall is a well-equipped multipurpose facility with a proscenium stage and 1,815 seats in an audience area that features several high balconies. (See Figure 2.)

Completed in December 1984 with a $45 million construction budget, the Ordway Music Theatre's grand opening took place on Jan. 1, 1985. Since then, the facility has hosted a wide variety of music and dance events. The theater's in-house sound system, designed by consultant Jim Gundlach through Lawrence Kirkegaard & Associates, handles most events that take place at the Ordway. It was installed by AVC Systems of Minneapolis.

"There are at least 60 dates a year that we use nearly all of our available sound gear for musical concerts," says Jim Pfitzinger, the theater's sound department head. "We have something going on here several days a week, but not every show requires full sound reinforcement. We have quite a classical musical schedule; there are perhaps 200 classical dates, but at least half of those need no sound system at all."

Pfitzinger notes that the sound department is required to respond to the needs of many different types of musical performances. "For the big bands, things are pretty simple. They usually need about three monitor mixes and a little bit of reverb on the vocalists. Nothing complex there."

Country music shows also make use of the Ordway. "Another concert room in town recently closed down, and we are picking up those bookings now." Pfitzinger says. "For those shows, we rent a monitor board, a power distribution system and stage stacks, and a few outboard effect devices. We are starting to get more complex shows in here, where they want maybe eight stage mixes. So, we have to find rental items when a show is booked in to use house sound. Because of the nature of this facility, rock-oriented performances are uncommon."

House sound system

The Ordway's house sound system includes left-, center- and right-cluster arrays installed over the stage in the proscenium. These arrays are driven by signals originating in the 32-channel console's output mix matrix. A total of one dozen 18-inch JBL 2245 bass speakers are installed in vented boxes, and handle audio program material up to 200Hz. A JBL model 6230 dual-channel power amplifier is bridged to mono and connected to each driver.

Reproducing lower midrange frequencies in the room is no problem, with a total of 15 Community Light & Sound M-4 units installed, five per array. A single channel of a JBL 6260 amplifier is connected to each M-4. For frequencies from 2kHz to 12kHz, 15 JBL 2445 compression drivers (mated with 2380 and 2385 horns) are available. For high frequencies, 27 JBL 2404H tweeters units have been included.

"The overhead clusters do a pretty good job of getting the sound out there, but we find that some small fill units on stage are necessary to keep the 'image' of the show sound in perspective", says Pfitzinger. "I know that the owners of sound systems in public buildings like this prefer to see everything 'nailed down' and installed, but some portable speaker units that can be used in various configurations are essential, in my opinion. It is a real challenge to do contemporary, popular shows in a room like this with an installed house sound system."

For the Flim & the BB's concert, additional rental sound equipment was

Figure 5. *Richard Craig, B&K field applications engineer, observes the acoustical response of the sound system in the room. Meyer CP-10 parametric equalizer is used for EQ adjustments.*

Figure 12. Jimmy Johnson (Flim) adjusts his AMS stereo digital delay.

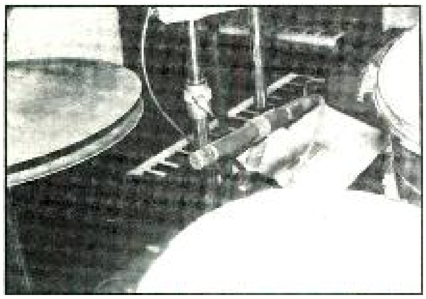


Figure 11. A model 4011 directional microphone was positioned near the snare drum's top head and the high hat.

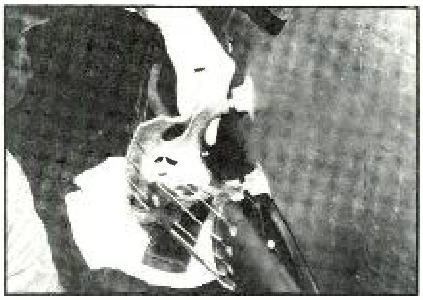


Figure 13. A B&K accelerometer was attached to the bottom of the bridge of the acoustic bass. Note tiny transducer head near fingertip.

brought in from Anicom Sound of Minneapolis and Pi Audio/Design. Four subwoofer enclosures were used on stage along with Meyer UPA-1A boxes. The JBL Concert Series sub-bass units, each box loaded with one or two 18-inch loudspeakers, were driven by JBL 6290 amplifiers operating in a bridged mono mode. A Rane crossover was used to supply the subsystem with an 80Hz low-pass mono feed, with a selective low-bass mix obtained from an auxiliary send on the house's Wheatstone/Audioarts mixing console. (See Figure 3.)

"When Flim & BB's first began talking about doing this hometown concert, and I got involved, my first concern was the sound reinforcement system here," says soundmixer Scott Rivard. Contacted by the group to handle the live mix for the concert, Rivard first was exposed to the musicians seventeen years ago.

"My background is primarily in multitrack recording," he says. "I got started when 4-track was evolving. I guess I first did a session with Jimmy Johnson [Flim] and Bill Barber in 1970 or 1971. Then, I worked on the direct-to-disc project for 3M, with Tom Jung. I've been interested in the group and their music ever since."

Currently, Rivard is a partner in a 24-track facility [74th St. Studio] in Minneapolis, and is technical director for Minnesota Public Radio. He has been responsible for the audio on "A Prairie Home Companion" with Garrison Keillor.

"In checking out the system here at the Ordway, I felt that we'd really be fine in terms of the primary sound reinforcement system, but I was concerned about reproducing the low bass, so we've added this auxiliary system," Rivard says. "My other concern was the input to the system. I really wanted to get things as clean as possible.

"We've had some real help from Bruel & Kjaer on this production; because the group used B&K microphones to record their music, we've gone ahead and obtained the same devices for the live show here. Bruel & Kjaer also has been using Flim & the BB's as something of a 'field-test' project, so for this concert, they brought in the Model 2032 FFT dual-channel signal analyzer to look at the system's response in the room." (See Figure 4.)

Most often, the equalization process for live sound reinforcement systems includes the use of broadband pink noise, and this obtrusive test signal must be used prior to the audience's entrance into the auditorium. Thus, not only is the annoying noise required prior to showtime, but the audience will actually change the acoustical environment, making much of the previous EQ work less valid.

According to B&K users of such devices as the 2032, FFT (Fast Fourier Transform) analysis can give excellent results when this measurement technique is applied to the investigation of the acoustical performance of a particular speaker system in a given room environment.

Traditional ⅓-octave measurement devices average the acoustical energy found within a ⅓-octave band; the high-resolution 2032 accurately depicts the acoustical energy in very narrow bands (6.25Hz resolution when using the 5kHz range, for example). This analyzer, when combined with an accurate and versatile parametric equalizer, enables very fine adjustments to be made on the sound system while it is in use during the musical program.

"It is worth our time to bring this equipment out into the field and work with live music, particularly when the group is so concerned with audio quality," says Richard Craig, a field applications engineer for B&K who lives in Minneapolis. (See Figure 5.)

"We learn a bit more about how effective this measurement process can be each time we use it," he says. "Dual-channel FFT analysis has been used by engineers and scientists for many years for research and development in a wide variety of industrial fields; applying a known technology to the concert scene has proved to be exciting for us."

During the band's sound check in the main hall of the Ordway, Craig observed the sound system's acoustical response in the room. A few room resonances were obvious, along with some "holes" in the sound. (See Figure 6.) Some initial equalization adjustments were made with a Meyer CP-10 complementary-phase parametric equalizer. (See Figure 7.) After filter adjustments were completed, another room-sound analysis showed the extent to which these anomalies had been corrected. (See Figure 8.) Throughout this process, the band was performing live music on stage, and no test signal was audible. The music was the test signal.

People not familiar with this method of system/room equalization are often curious about the validity of these measurements. Isn't the music (and thus the test signal) different at every instance during the process? What about background noise such as audience applause?

To get around these potential problems, system operators average several sets of data, which help to eliminate the random time-varying signals. The 2032 also provides the system operator with the ability to check measurement reliability by looking at the coherence function. A low coherence reading means that measurements are not valid and further averaging should be performed. [For more information on this subject see "Acoustic Measurment Techniques for Sound System Equalization" in the June issue.]

Stage instrumentation

With only four musicians on stage in a contemporary jazz format, Flim & the BB's typically has minimal microphone input requirements. With the addition of "The Dads" for the Minneapolis concert, several additional instruments were on stage, but the entire ensemble still required less than 24 stage lines to the mixing console. Of these, a total of 13 were open microphones, with the balance being direct inputs. All microphones used were B&K 4000 series microphones. (See Figure 9.)

"Traditionally, omnidirectional microphones have not worked well for live sound technicians, particularly when a full rhythm section is involved," says B&K's Adrian Weidmann. "This has been perhaps more a function of the frequency-response characteristics of those omni mics used in the past, rather than the omni-directional pickup pattern of the mic itself. The 4000 units used here, such as the 4003 and 4007, are prepolarized condenser units, and they have a great ability to handle high peak sound pressure levels. For example, the 4007 can take 155dB before clipping its internal pre-amplifier system."

Drums and percussion

Drummer Bill Berg's kit, located center stage, was picked up with only four microphones: a 4007 for the kick drum, a unidirectional 4011 for the snare/high hat, and a pair of 4003s for the toms. (See Figure 10.) One 4003 was positioned between the 8-inch and 10-inch toms; another was located near the floor tom.

"I really enjoy the sound I can get from smaller toms," says Berg. "These are a suspended-mount and have a cushioned metal frame, with rubber grommets so the drum is attached by the frame instead of the drum shell. I'm using a 12-inch floor tom here; I often also use a 14-inch, but didn't have room to bring it with me on the plane!" (See Figure 11.)

A pair of 4011s, with console inputs panned to the left and right, was positioned over the hand percussion table, located near the drum set.

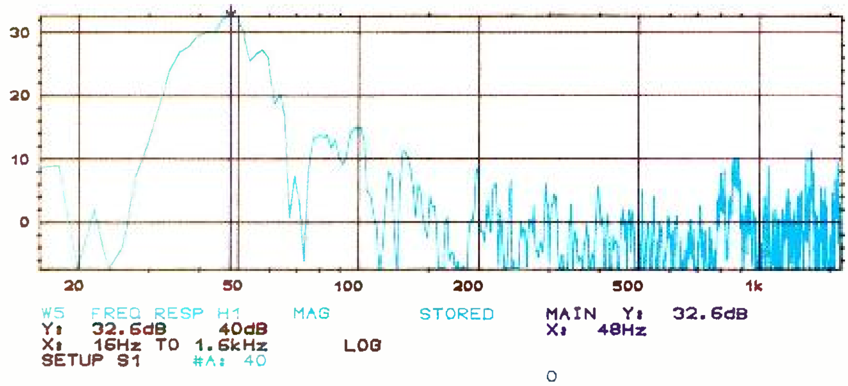


Figure 6. *An example of the system's response in the room, viewed from 16Hz-1.6kHz. Cursor is set at the center of a response "peak," 48Hz.*

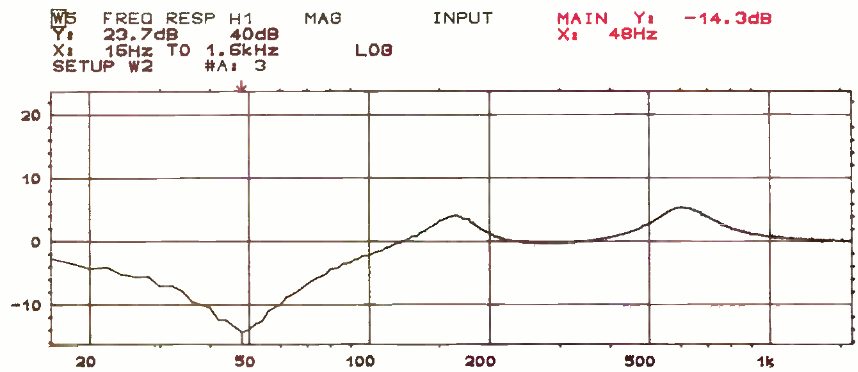


Figure 7. *System printout of the corrections introduced by the Meyer CP-10 parametric equalizer. Note that a narrowband correction has been introduced at 48Hz.*

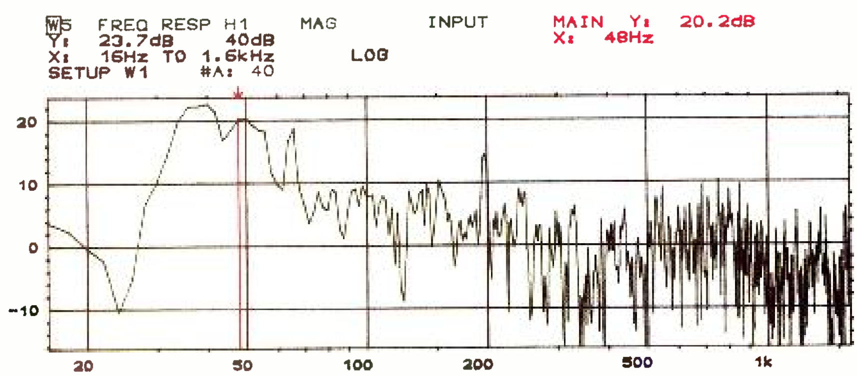


Figure 8. *The acoustical response after previously shown corrections.*

Bass

Flim's Alembic 5-string bass was enhanced with an AMS dmx 15-80S stereo digital delay, located onstage. (See Figure 12.) An active direct input box from Simon Systems with power supply received left and right summed signals from the Alembic IN-2 input module/power supply unit. The output of the AMS was padded down to guitar level, and mixed with the padded dry level out from the IN-2. The DI's output went straight to the house mixing console, and Johnson monitored his own playing through a foldback speaker onstage.

"It's a pretty odd way to wire it up, but it works well for me," says Johnson. "I can carry this setup with me easily on the plane. I tend to travel light. I do have a

Figure 9. *Microphone list.*

Input	Function	Mic
1.	Kick	4007
2.	Snare/hat	4011
3.	Toms/cymbals	4003
4.	Fl. Tom/cymbal	4003
5.	Percussion	4011
6.	Percussion	4011
7.	Acoustic Bass	Accelerometer
8.	Electric Bass L	DI
9.	Electric Bass R	DI
10.	Sax, Alto (Dad)	4006, modified
11.	Sax, Alto and Tenor	4007, modified
12.	Sax, soprano	4007
13.	WX7 Yamaha (wind)	DI
14.	Piano Low	4003
15.	Piano High	4003
16.	Synthesizer	DI
17.	Synthesizer	DI
18.	Synthesizer	DI
19.	Synthesizer	DI
20.	Synthesizer	DI
21.	Synthesizer	DI
22.	Vocal	4011
23.	Vocal	4011

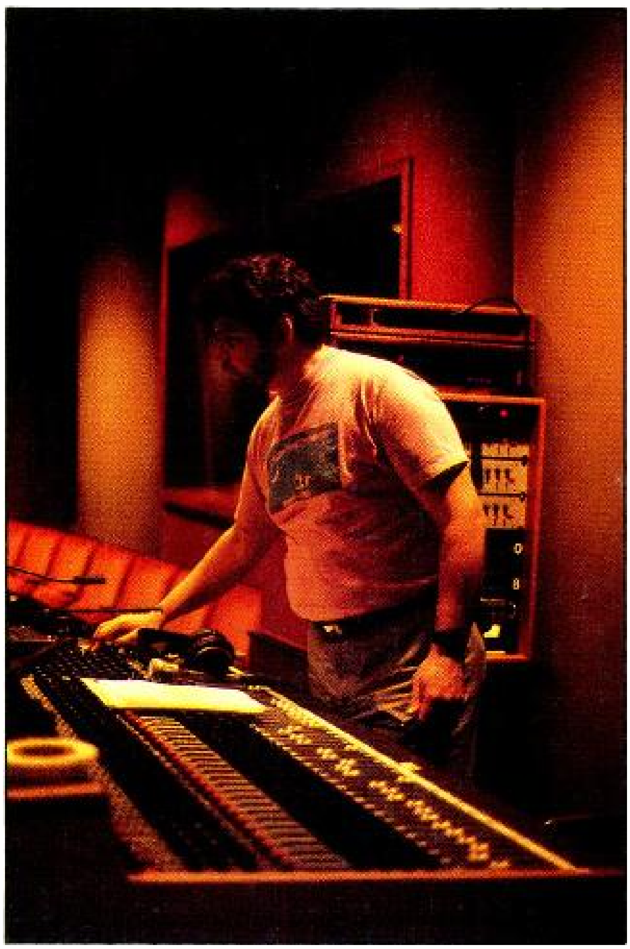


Figure 3. *Soundmixer Scott Rivard at the facility's 32-channel Audioarts/Wheatstone mixing console.*

couple of bass rigs with speaker cabinets to take with me on the road, but when you stop and think about it, playing in concert situations with a large, full-range PA system gives you the chance to use the PA for your bass rig. If you have a separate amp and speaker on stage that's just for you, but it can clutter up the stage sound as well. I like to rely on the main system."

Johnson's father played a vintage upright acoustic bass that was equipped with a Bruel & Kjaer "accelerometer" (a precisely calibrated, pressure-sensitive transducer used in critical acoustical measurement applications). Mounted beneath the wooden bridge, the tiny transducer head was pointed at the instrument's body. (See Figure 13.)

"So far as we know, this is the first time this transducer has been used for the digital recording of a musical instrument like this," says Weidmann. "This was the unit that Tom Jung found really satisfied him during the recording of 'Neon.' "

Keyboards

Billy Barber's grand piano (for the Minneapolis concert, a 7-foot Baldwin) received a pair of 4003 microphones. (See Figure 14.) The pair of mics was factory-matched with respect to frequency response, phase response and sensitivity (within 1dB). B&K's 3530 A-B stereo microphone mount held the matched pair above the strings. In addition to the grand piano, electronic keyboards were also in use.

"They're getting closer, but the various electronic keyboards still haven't replaced the acoustic piano," says Barber. "Not even the Kurzweil. You're talking about a lot of digital information to accurately reproduce all the subtle nuances of a grand piano."

Saxophones

Dick Oatts' saxophone was equipped with a modified 4007 mic element, removed from the microphone case and suspended in front of the instrument's bell. (See Figure 15.)

"The omni mic cartridge gives us higher sound quality," says Oatts. "It works really well for my tenor sax. For the alto sax, I am playing into a modified capsule from a 4007, with a measurement microphone pre-amp. Putting just the mic element at the saxophone gets the weight and mass of the rest of the mic unit away from the instrument, so I have a lot of freedom as I am playing.

"I've really been happy with the way the instruments are sounding now. The soprano sax, though, can present an interesting challenge. You can't just put the mic down in the bell because the sound radiates from the body of the instrument in that frequency range."

For the second alto sax, played by Dick's father, Jack Oatts, a 4006 (with 16mm capsule) was used. (See Figure 16.)

"The outputs are slightly different for the 4006 and the 4007," says Weidmann. "The 4006 has a bit more output and is more directional. Having two instruments playing together live, with the two different units, gives us a good chance to compare them to each other."

On some selections, Oatts also performed with a Yamaha WX7 electronic wind instrument.

Although Flim & the BB's perform instrumental jazz, a pair of vocal microphones were placed onstage for announcements and acknowledgments. A 4011 was used for this purpose. A directional microphone with a 19mm-diameter element, this cardioid mic is transformerless and will accept a 158dB peak sound-pressure level before clipping.

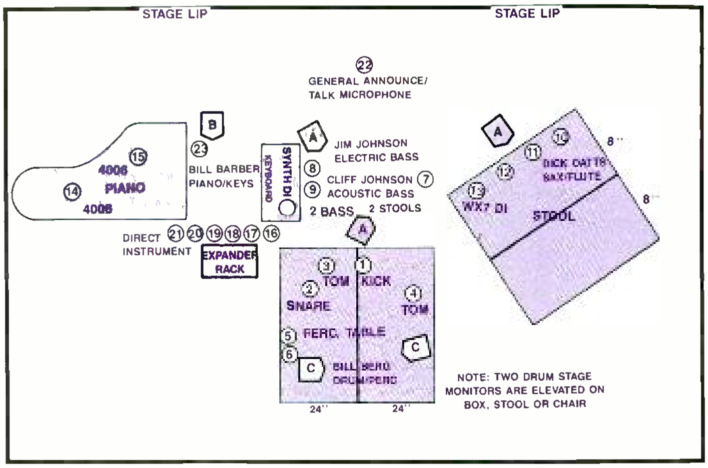


Figure 10. Stage chart.

Mixing the show

During soundcheck, soundmixer Scott Rivard took care to locate the optimum position for each microphone on stage, and mic pre-amp sensitivities were set. As the sound of each instrument was "dialed in," it became apparent that the input to the sound system was extremely clean. Between numbers, every tiny move of the drumsticks and each squeak of the piano stool was audible throughout the audience area, testifying to the sensitive mics and high system gain potential.

"We're working with pretty low monitor levels on stage, so as to not contaminate the house sound any more than necessary," Rivard says. "If too much reinforced sound is present, it will really be evident in the mix, particularly on the drum sound."

Because Flim & the BB's is primarily a recording studio group, no effort was made to make a critical live recording of the concert.

"I'm running a cassette tape recording, of course, for the group to listen to after

Figure 4. The Bruel & Kjaer model 2032 dual-channel FFT signal analyzer.

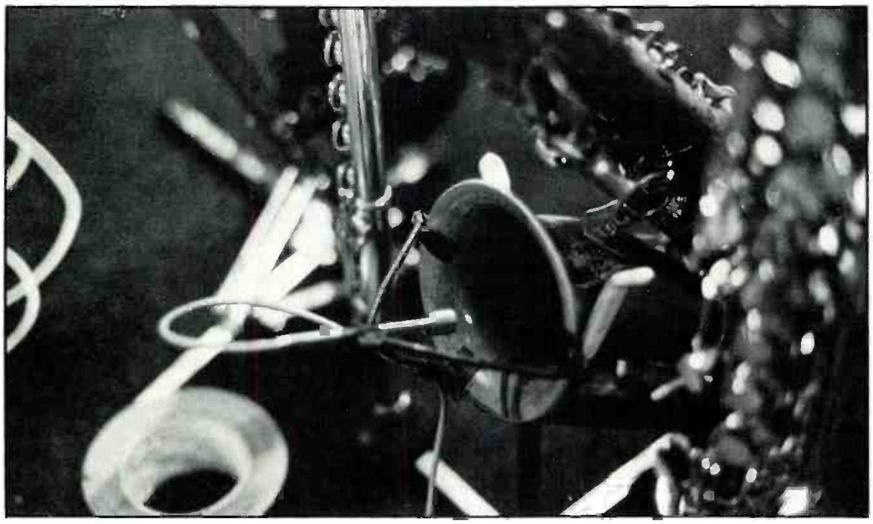


Figure 15. A B&K microphone element was removed from the mic housing and suspended in front of the saxophone's bell.

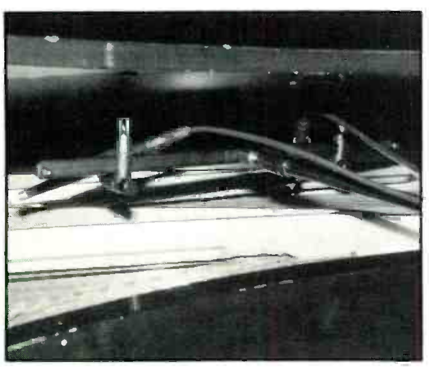


Figure 14. A matched pair of 4003 microphones were suspended above the piano strings (lid set on short stick).

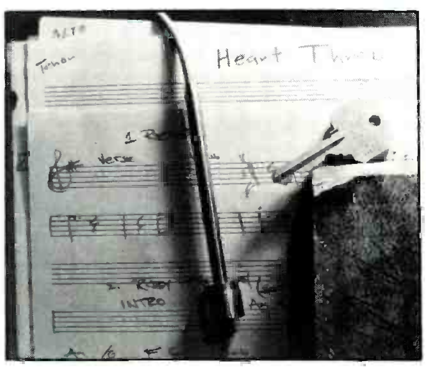


Figure 16. Mic element used for saxophone pickup rests against sheet music; key shown for scale.

the show," Rivard says. "Doing a live recording for commercial purposes is not really a priority, though. And, the EQ I'm using for the house mix is somewhat inappropriate for doing a recording. Because the available console does not have extra outputs to do a separate record mix, I'm not really concerned about documenting the show as much as I am attempting to let the audience receive the same impact from the live show as they do from hearing the recorded product at home."

Before the event, I found myself thinking that I did not envy Rivard his appointed task of attempting to interface a small stage-stacked system with a permanently installed house system, even one that was apparently capable in terms of room coverage and headroom. Many installed cluster-type systems are compromise design projects that are meant to serve many functions, and just do not work well for all contemporary musical events. With a live concert on tap that was of such a critical nature with regard to audio quality, I was eager to witness the attempt to present the group's music live.

The excessive amount of attention that had been paid to microphone selection and placement, and to system equalization, paid off. While this particular setup would probably have been much more difficult to use if the level of the program material had been higher, I felt that this group's jazz format worked quite well in the Ordway Music Theatre.

Jim Pfitzinger agreed. "It sounds best in here when we have an act on stage that doesn't expect to get 115dB coming out of the stage monitors," he says. "In this room, too much stage sound washes out the audience mix, and nobody hears anything but noise."

This particular concert event proved to be a success. It was an interesting chance to see what sort of results can be expected by a musical group when relying primarily on a permanent in-house sound system at a performing arts center. For many acts that do not travel on a regular basis, or do not tour with a consistent production package, this approach to concert sound can work very well if special attention is paid to detail.

Microphones and signal processing gear can be compact enough to be carried with guitars and other band gear on an airplane; such specialized equipment as detailed here can make the difference between a mediocre event and a memorable one. RE/P

Photos by David Scheirman.

This article has been written with reader interest and education in mind. The mention of specific brands of manufactured sound equipment is not to be taken as an endorsement by the author, *RE/P* or Intertec Publishing.

Producer Interview: Patrick Leonard

By Jeff Burger

In six years, Pat Leonard has gone from being Chicago studio musician to one of the top-selling record producers in the world.

Madonna. Bryan Ferry. Peter Cetera. Julian Lennon. Jody Watley. Kenny Loggins. Hiroshima. Fee Waybill. Christopher Cross. Carly Simon.

What do these artists have in common other than being consistent chart-toppers? Production by Patrick Leonard. And almost every one of those relationships also includes co-writing credits.

Just six years ago, Leonard was working the Chicago jingle jungle while honing his songwriting and production chops. The opportunity that shifted his musical endeavors into high gear was his role as keyboardist on the Jackson's 1984 "Victory" tour. The biggest stepping stone of his career presented itself the following year in the form of musical director for Madonna's "Virgin" tour. That relationship blossomed, as he found himself co-writing and co-producing six songs on Madonna's album "True Blue."

After working on the scores for "At Close Range" and "Nothing In Common," Leonard garnered BMI's award for Most Performed Motion Picture Song of 1987 for "Live To Tell." Next came the production and co-writing of Bryan Ferry's 1987 "Bete Noire" LP, and numerous writing credentials for other artists.

Last fall, Leonard opened his own private studio, Johnny Yuma Studios, Burbank, CA, to fill his production needs. While the live micing areas are certainly respectable, the main attraction is a MIDI/keyboard production center behind the traditional engineering station. A custom MIDI patchbay provides access to approximately 150 MIDI outlets located around the facility. A bank of synthesizers and drum machines line the rear wall, all controllable from Leonard's master keyboard/sequencing station.

This custom world (complete with weight room, sauna, entertainment center and security parking) has become a comfortable second home for Peter Cetera and Julian Lennon while creating their new albums. Up next: you guessed it, another Madonna LP.

RE/P: To what do attribute your success?

PL: It's the old 15-year instant success story. There are a lot of things that people attribute success to. A lot of people will say it's luck or being at the right place at the right time and having the talent to back it up. But I've always done just one thing and I'm still doing it—I write songs and record music. That's what I attribute it to. I've always been a writer. I've never done other people's songs.

I've always thought that creative work was my strongest thing. Whether it's an artist looking for a producer or songwriter looking for a partner or a person looking for overdubs, they know that when they call me, they're going to get something creative. If they want something that's generic, don't call me because I don't know how to do that.

RE/P: Is there anything in particular that helped you make that transition from player/writer to producer?

PL: It was sort of a natural thing. Artists were paying a lot of money to other producers and then looking me in the eye and saying "the demo was a lot better!"

Obviously, "True Blue" didn't hurt. Doing those songs and that record with her [Madonna] happened because she said, "Oh, yeah, this will definitely be better if we do it together than if we get some other producer to do it." From that, I started collaborating with other people.

The nice thing for me is that the records don't resemble one another. The Bryan Ferry album sounds completely different from Madonna's album. Peter Cetera's album has nothing to do with those other two and Julian's is a street rock 'n' roll record.

It's coming from different backgrounds. It's loving Jethro Tull, Led Zeppelin, Chick Corea, the Beatles.

RE/P: What differences do you find in working with such a wide variety of artists?

PL: Because they're all complete artists, it's a relationship based on who they are as people, and who you are. If you don't give them room to be who they are, it's over real quick. I'm just letting them be who they are and finding out what I need to do—what they need from me.

It's easy to do with somebody like Julian because he came to me with finished material. I said, "OK, I see what you're trying to do with this and I think I know what we need to do to make it happen. He's very soft in his approach. Peter, on the other hand, is much more direct in terms of ideas and says "I hate it" or "I love it" or "Let's try this."

Bryan is much more esoteric, so the approach is not direct. Once I saw that he doesn't approach things directly, I learned not to approach them directly. I don't say, "We need to do this." I just sort of let it evolve. It's much less focused music, but that's Bryan Ferry. He's unlike Madonna, who is very focused. That's just the way she is.

RE/P: How do you select the songs for a record?

PL: By feel. By what I like. On Julian's

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

record, my wife and I went of vacation and I had his demos. I kept playing them in the car and she'd say, "I like that one" or "I don't care for that one very much." I listened to what she said, put it together with what I thought and what I could see past what the demos presented and came home with a list. I said to Julian, "Out of the 20 you played me, I think we should cut these eight." He agreed and what he didn't agree with, he didn't mention until we got into it.

We started recording and working with the songs I had selected, and on a couple of them we said, "It ain't happening." So, we tried doing a couple of the other songs I hadn't picked in a different way, and they worked.

RE/P: *Some producers will purposely pad the hit songs on a record with lesser songs that make the hits stand out, while others seem to go for as many hits as possible. What's your approach?*

PL: I don't look at it that way. I hope that we have singles and usually we do. With Bryan Ferry, I thought we had singles, but felt that radio just wasn't going to play them no matter what they were. It's a shame that really good creative music doesn't get past Debbie Gibson, Tiffany or Run DMC—and if you want a list I've got one! We just hope there are singles, but I would never ever try to do an album that was just singles. With Madonna, for example, that wasn't what we were trying to do. We were just writing songs and recording them.

I'm not a singles producer and I don't get called for that. There are people who do that and that's what you expect from them. Not that there's anything wrong with that. I just don't see myself that way, and I don't want that pressure on me.

RE/P: *Do you do all your work here at Johnny Yuma now?*

PL: Other than mixing, I do everything here. There are no limitations as far as my record making. It's the room I wanted, I built it the way I wanted, it sounds the way I wanted it to sound, and the gear is wired and set up the way I wanted it to be. This is pretty much my home. I had the studio in my house before, so I knew what I didn't want and what gave me a hard time there.

RE/P: *Where do you mix?*

PL: I leave that up to the engineer. The first day of a mix, we go in and get oriented. This is what this project sounds like in this studio. I always feel it's best that we're all there for the first mix, even if we sit in the lounge and watch TV most of the time, just so we get used to what the music is going to sound like in a different room.

That way, we don't come in for the first time and the engineer says, "Here's the mix, it's finished." He may have been working for two days and we say, "What is this?" The first day in a new environment is so disorienting.

While I'm back at my studio working on something else, the mix engineer will send DATs over from the other studio when he feels he's at the right point for us to listen. The artist and I will make a bunch of notes and send them back or call or go over there, depending on how critical it is. I don't need to be there for all the mixing, especially if it's the same engineer that recorded the tracks.

RE/P: *Do you do any engineering yourself?*

PL: For the writing stages, which is where a lot of the stuff gets recorded, I do all the engineering myself. It's just the artist and myself. As things go along, it depends on the project. For the live rhythm section on Julian's project, I didn't even play keyboards. I brought in a keyboard player because I didn't want any responsibilities except listening. I didn't want to deal with my part and my sound. I wanted to listen to the whole thing. So, on that record I didn't really do any engineering at all.

With Peter's album, I engineered all of the basic tracks that he and I recorded in the writing stages. Then I brought in a different engineer to do the drums, a different engineer to do the tracking and a different engineer to mix. This was done because each engineer had a different forte.

RE/P: *Are you picky about monitors?*

PL: Yeah, I've gone through so many things. I have a shelf in the storeroom here with 10 sets of expensive speakers. Like a lot of other people, I keep trying to find the missing link and it doesn't happen. A set may be OK for six months, then I realize there's something about them I just don't like.

The last speakers I had were KRKs handmade by some guy here in town [Los Angeles]. I loved them, but they're so big that I can't use the big speakers when they're set up. The engineer I did Bryan's album with uses a combination of an Auratone and a Radio Shack speaker hooked together in series.

There's absolutely no reason why they should sound any good at all, but if you get things to sound good on them, it really translates brilliantly! [Editor's note: For more information on setting up small speakers in series, see "Close Field Monitoring" by Jeff Blenkinsopp in the May issue.] I go through phases of speaker fanaticism where I'm pulling my hair out trying to find the speaker of my dreams, and I wind up with Yamaha NS-10s every time.

RE/P: *What are you using for larger speakers?*

PL: UREI 813C with Boulder power and White EQ.

RE/P: *Do you see things through the mastering phase?*

PL: At some point, mastering becomes sort of a gray area for me, because nowadays you're mixing to digital, and you're not so concerned with cutting a lacquer disk anymore. Lacquers are tough because you need to EQ a lot, change levels, compress and de-ess to get it right for vinyl.

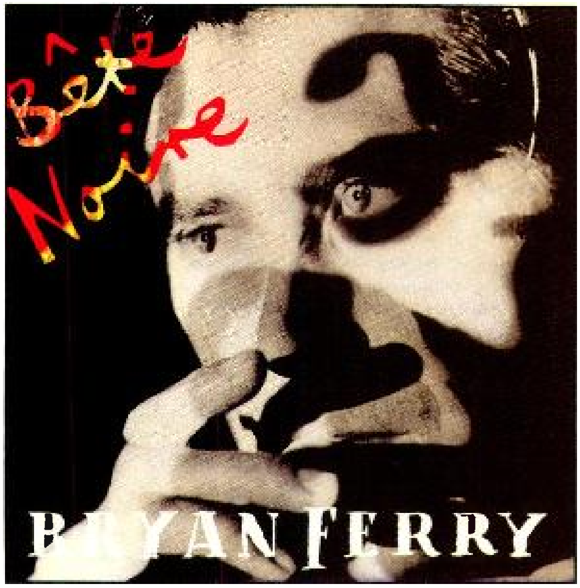
Other than that, get them to CD and leave my tapes alone. The engineer I'm using now puts an EQ across the 2-mix bus, so when we're mixing, he's EQing the overall track anyway—and doing kind of what might be done in mastering. It seems to work real well.

RE/P: It looks like you're fond of DATs.
PL: DAT is such an amazing format. I don't use cassettes at all. DATs at home, in my car and here in the studio. It's also great for field work—samples and that type of thing.

Cassettes are so inconsistent. Judging mixes from cassettes is so ludicrous, especially considering everything's going to CD anyway. By next year I don't think records will even exist.

That's the pain in the ass about making records—the records themselves. Albums are terrible. Plus they're dusted at the manufacturing plant. They have this fine powder that is put on the record to keep the surface noise down and make it go into the sleeve faster—so they can run the machines faster. The dust also cuts off everything above about 12kHz because all the high end is at the bottom of the groove and that fills up with dust. On the records I make, I call up the plant and say "You dust my records and I'll be down there with a shotgun!"

RE/P: Let's chronicle the evolution of a project start to finish.
PL: We typically lay SMPTE time code first with a 2-bar count-off starting at 20 seconds. Then I usually use an SBX-80 running a drum machine with a sidestick sound because it's easier to play to than a metronome going "dink-dink-dink." It depends. I'm seriously thinking of trying the next Madonna album with a rhythm section and no click. Tape rolling...good, here we go...count one, two, three, four.



We can always erase it. I think people are starved, by about now, to hear *people* playing music.

RE/P: *What's next?*
PL: The usual is to start writing the songs and do them as sort of drum machine/synth/MIDI demos—putting the stuff on multitrack. In the case of Madonna, it'll all go down in one day, each with lyrics and everything.

Once all the material's written, we look at which songs should stay synth songs and which ones need to be cut with a rhythm section. Everything gets real drums. If it's drum machine stuff, it's in addition to the drummer. I don't use drum machines even though it sounds like it. Next, we go through the process of overdubbing whatever's needed.

We'll spend a few days doing guitars. We'll bring in a guy we like and have him play on all the songs that are right for him. Then we'll bring in another guy and do the same thing. There's never one person who is good for everything. The lead vocals and background vocals are next. Periodically I'll do a synth overdub that just pops up.

I usually work from three tapes. First is the master, then you make a multitrack slave of it bouncing the master material down to, six tracks on the slave—drums, guitars, bass, keyboards. The second reel is for overdubs. I'm a track-eater; I like to do things in stereo. I also like a lot of little things going on and I don't like them on the same track. If you've got two guitar players and they each do two passes, 24 tracks don't last long. Once the second tape is full, we make up a third tape, which is used exclusively for vocals. When we go to mix, we take tapes two and three and bounce them to a digital 32-track—then mix from the 24-track analog master and the 32-track digital.

The theory is that the first 24-track master hasn't been run since the day you cut it and made the work slave. The stuff that has been run quite a bit you're bouncing to digital so that you're not wearing it out anymore as you're mixing. And, this way you also have automatic safeties.

RE/P: *What kind of guidelines do you use when interacting with musicians?*
PL: With a drummer I usually say, "Play this kick pattern, this snare pattern, this high hat figure, then put fills where you feel they're appropriate." The guitarists and I usually sit together and come up with a part. It depends on the situation again. If it's a real rock 'n' roll thing I just say, "Play and have a good time until you find something that's killing all of us." Sometimes I just say, "Play anything you want to play front to back, five tracks, thank you, goodbye," and then we comp it together. It just depends on the kind of feel that's needed. If it's loose rock 'n' roll, then I don't tell anybody what to do. If I want something esoteric, I might have 10 people play what they think without telling them and then pick a note from this guy and a note from that guy, etc. Then you listen to it and say, "What is that?" That's the Bryan Ferry school; you don't even know what you're hearing. That's because it isn't anything that was done by any one person. It's something that was done by a lot of people, so there are a lot of thoughts flowing in and out of each other.

RE/P: *It also sounds like you're still fond of your Minimoog for bass.*
PL: Yes. I have a really good bass patch on the Super Jupiter that we used on just about everything on the Madonna record. We would blend it with different things and EQ it differently to make it appropriate for each song. But it would never stay stable—sometimes you'd get 0VU levels, sometimes −3 or −6, and then you'd see it come back up. I have since had my Minimoog MIDIed. Now the Super Jupiter bass patch has dust on it.

The Minimoog is consistent—it never dips. I was in search of the ultimate bass sound for the past few years and now that the Minimoog's up and running again, that's it.

RE/P: *Do you have any other favorite pieces of gear?*
PL: Yeah, I love the TC 2290 digital delay/chorus/sampler. I have two of them linked together in stereo with a fast trigger modification and 22 seconds of sampling in each one. We can fly-in stereo guitar parts, background vocals...anything. They have 48kHz sampling rates, so the frequency response is CD quality. They're quiet as a mouse, with signal-to-noise of 100dB and the trigger time is around 4s. They're amazing.

They save us a lot of time with background vocals, lead vocals and doubles. The new Eventide H3000 Ultra-Harmonizer is an unbelievable machine, too. I love the Akai MPC-60 for writing, sampling and replacing drums. There might be a little section in a tune where there's a flam or a long tom-tom fill and a couple notes are just not quite dead-nuts on. If you start putting other things around it, you can really feel the flaws. So we'll sample the whole kit into the Akai and just do a running punch. You never hear a change because the sampling's that good. It's saved me on a couple things.

RE/P: *Let's talk about the role of machines in the music-making process.*
PL: What do you want to know—how much I love them or how much I hate them?

RE/P: *Both.*
PL: That's a difficult one. Philosophically, I hate them. Technically, I love them.

I don't think they've done anyone a favor or helped the state of music—they've hurt it considerably. That's a very loaded statement for someone who's made records with machines for five years. I don't think there's any reason to listen to something a machine did more than once to hear how cleverly it was done because there's no performance value. It's not a person doing it, so it's not a performance in the real sense. Not if it's quantized. Not if the dynamics have been corrected. Not if all the note values have been changed by a computer. Not if you can't make a mistake. A performance is all those things.

Machines also make it very simple for anybody to copy anything. All you've got to be able to do is tap. Bass line, the same thing. It's giving people who can't really play the ability to copy what others who can play are doing. Consequently, the radio is full of people copying what other people have done. In that sense, it's bad and I think it's taken music three giant steps back. The problem for me is that I can't listen to mechanical music. I don't think music is a machine—music is a person.

The upside is obvious. You have a lot of people who are really capable of doing brilliant things because they can get the stuff down so quickly. Their minds are just flowing with ideas and, bang, you get these brilliant arrangements.

RE/P: *How do you reconcile all that in a room full of machines?*

PL: I don't use machines to make records. The sequencers and drum machines are writing tools. I do use synths on everything. But I don't use factory sounds. I do all my own programming so that the sounds fit the parts.

Sometimes I'll use the Macintosh computer as a tape recorder so that I can do layers of things and hear them without recording them and maybe have three or four parts of a synth arrangement all go to two tracks because they've worked out individually. Sometimes I'll do a part that I think needs to be quantized and I will quantize it. It's not that I can't play it, it's that I think the quantization is more appropriate musically. In that sense, it's great. If you want a drum machine sound, use a drum machine for effect. If you want drums, use a drummer.

RE/P: *Do you have real drummers triggering electronic sounds?*

PL: On every record. I have trouble with drum machines and sequencers, not synthesizers. That's mostly because I can't listen to the radio. I can't take it. There was an article in last week's *Billboard* saying that everybody's copying everything everyone else is doing and cloning it exactly. What they didn't mention is the reason they're able to do it. It's all done on machines, you just do the same program that was used on someone else's last hit. If you're not even remotely capable of it, you just use step time. It's a pretty scary thing.

Does this sound familiar? "You've got nice legs, so we'll use a drum machine and sequencers to copy everything that's already been done, we'll fix your vocals and you're going to be a big star." The drag is that it works because people are listening to the radio saying, "Oh, yeah, I've heard this before. I relate to that—that's 'that' snare sound I love." You listen to the radio and you're digging something and then about halfway through you realize you're listening to the snare sound because the song is terrible. The song isn't even a song, but the sounds are so good and so precise that you listen to that. Now, whether you're going to buy the record and listen to it again depends on what you think music is.

RE/P: *Is there anything that we haven't covered that you'd like to say?*

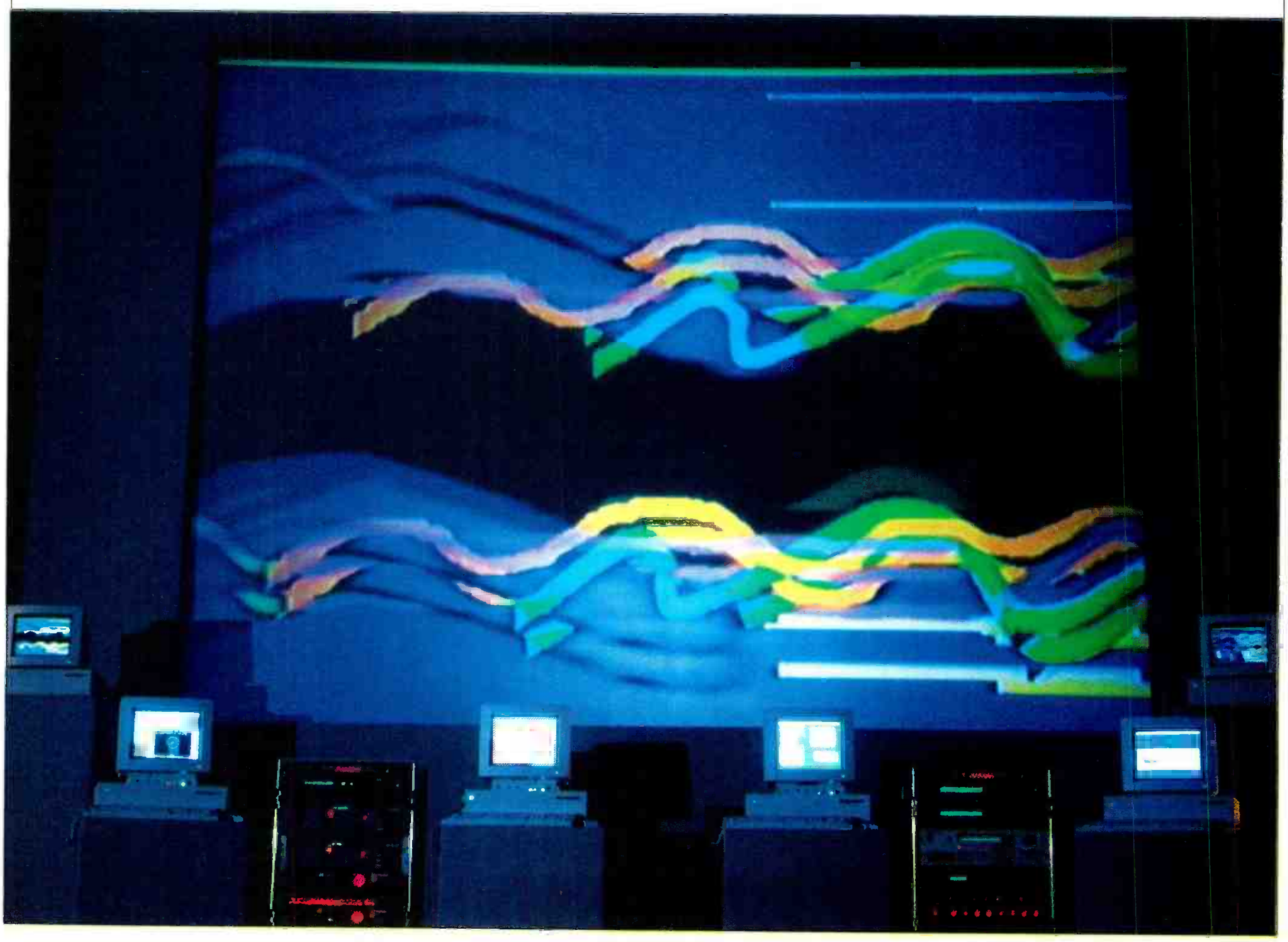
PL: Yeah...wanna buy a Dr. Click?

RE/P

Glass Apples: Philip Glass Uses Macintosh Computers for Concert

By Rosanne Soifer

Held in conjunction with the recent New Music Seminar, this concert marked Apple Computer's official entry into the music business.



Music and technology continue to demonstrate the strength and success of their marriage in yet another format at this year's New Music Seminar. On July 19, Philip Glass' "Music in Twelve Parts" was "performed," so to speak, via Macintosh computers. Co-sponsored by Apple Computer and Sam Ash Music, the concert officially premiered Apple's entry into the music business.

Says Caroline Donahue, Apple district sales manager for the New York area, "We looked at the music market and saw that the demand for our computers was already there—we didn't even have to try creating interest."

It's no surprise to anyone even remotely connected to music, audio, video, or recording that computers are increasingly entering our professional lives. *RE/P* incorporated a regular computer column into its monthly format last March, recognizing the need for those in professional audio to understand basic computer applications and terminology. The pro audio community, in terms of computer use, appears to be developing a vertical market. In other words, all levels of the audio industry are employing computers in some aspect of their work.

Apple's initial involvement with Philip Glass seems to substantiate this observation. Says Donahue, "We signed on with Sam Ash Music to teach them about computers and found they knew more than we did! Most of the sales staff were computer freaks and MIDI experts. Sam Ash manager Paul Rice had previously worked with Euphorbia—Philip Glass' production company—and that's how Apple got involved with him and the New Music Seminar."

Glass, incidentally, had been using Macintosh computers for at least four or five years, according to his producer Kurt Munkacsi. Munkacsi, who's been with Glass since 1971 (they met when he was working for John Lennon and Yoko Ono as an engineer) says, "Most studios are a bit behind the times, probably because it takes a while to develop really good software. For example, when it first came out, Mark of the Unicorn (which was used for the concert) couldn't operate in multiple time signatures.

"On one hand, automated consoles and digital delay lines have been around for a comparatively long time, yet the same studios using them are not picking up on using basic computers."

"Music in Twelve Parts" was composed between 1971 and 1974 for live performance on conventional instruments. Because it was originally written in "open score" (meaning any combination of instruments can be used), the instrumentation is somewhat flexible. The 1974 Virgin Records recording (scheduled for release later this year) features electric organs, soprano sax and flute, alto and tenor sax, and voice.

The July 19 concert demonstrated another version of the piece, this time arranged for and sequenced with 10 synthesizers.

Technical notes

The music was recorded into the Macintosh using the Mark of the Unicorn Performer 2.3 sequencer software. Says sound designer Miles Green of Euphorbia Productions, "The computer is capable of playing all of Michael Goldray's synth parts, which he had played and recorded individually."

All together, four Mac II computers were used during the reproduction processes. One of the Mac IIs was linked to two racks

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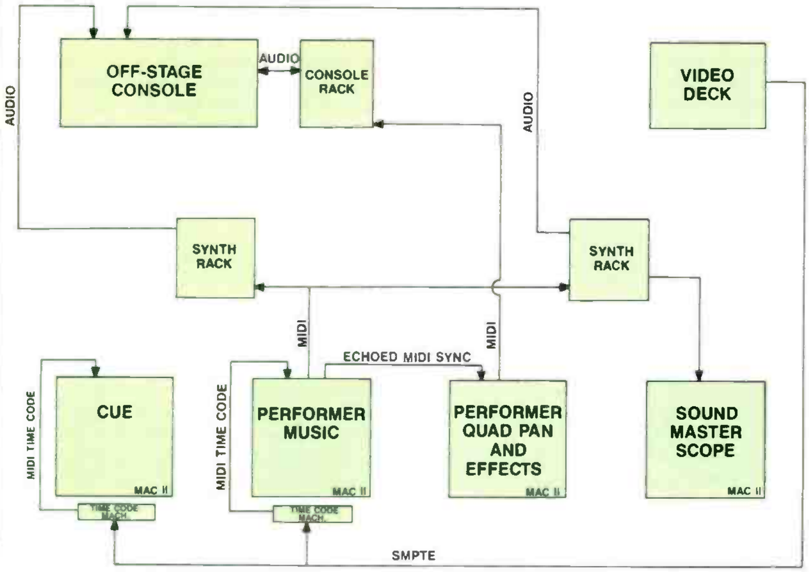


Figure 1. The Philip Glass Non-semble system configuration.

Equipment used

Software

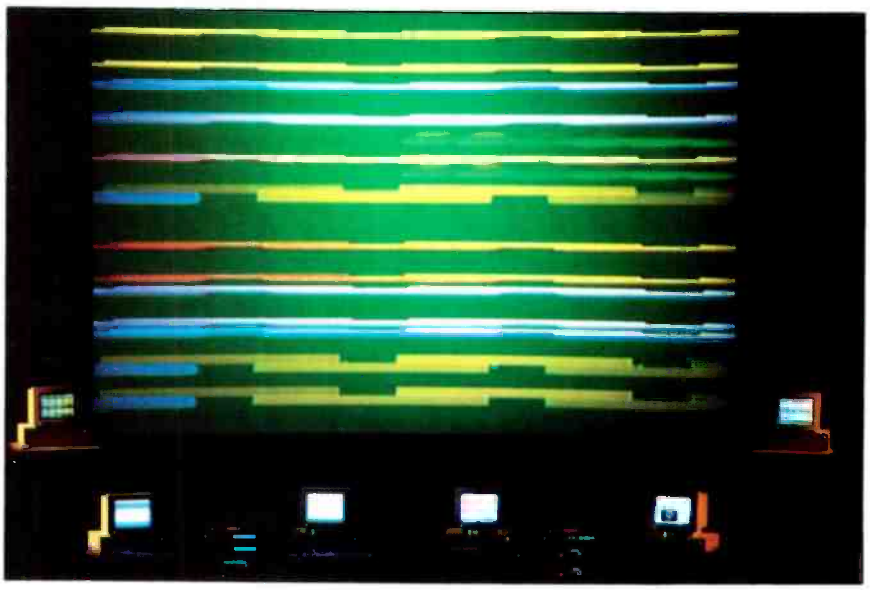
Mark of the Unicorn: Performer 2.3.
Opcode: DX-7 II/TX, Matrix 6R and D550 patch editors.
Opcode: Cue.
Digidesign: FX Designer (PCM 70).
Kurzweil: MidiScope.
Musically Intelligent Devices: MegaMix.

Synthesizers

1 Yamaha TX-816.
2 Roland D550s.
2 Oberheim Matrix 6Rs.
2 Oberheim DPX-1s.
2 Akai S900s.
1 Roland MKS-70.

Signal processors

2 Lexicon PCM-70s.
2 Alesis MidiVerb IIs.
1 Korg SDD-3300.



of synthesizer modules, which received the incoming note information and turned it into sound. In general, the independent modules are paired to produce a complex composite synth sound for each of the instruments in the score.

As the piece progressed, Performer sent patch change and volume commands to the synth modules, thus changing the timbres and relative levels of the instruments. For quadraphonic panning effects, several synths were patched into Megamix—a bank of MIDI-controllable VCAs—which in this application controlled the feeds to the quad system. Also used for quad flash were two Lexicon PCM 70s set to change their parameter values in response to MIDI controller information sent by Performer.

Using computers in pop music for orchestrating, sampling, editing and tapeless recording is becoming fairly commonplace. Now that computers and classical/non-pop music have discovered each other, the ramifications for both industries, as well as for the synthesizer-assisted live concerts of the classics are becoming more frequent, and concert music composed for such combinations as flute, violin and computer are no longer novelties. It may now be up to the pro audio field to educate both the computer industry and the classical/non-pop music community about each others' potential. Pro audio stands to gain much, both economically and in terms of good will.

Apple Computer is evidently ready to enter this market at a rather allegro tempo. Says Donahue, "People are really doing incredible things with some of our productions." RE/P

Photo by Andy Freeberg.

Hands On:

Kurzweil 1000 Series

By Paul D. Lehrman

The 1000 PX is a rack-mount version of the K1000 sampling keyboard, containing sounds described as "Kurzweil's greatest hits," such as grand piano, orchestral strings, choir and baritone horn.

When Kurzweil Music Systems started delivering the K250 a little more than three years ago, it caused a tremendous sensation, followed by equally great controversy. Many who used the multitimbral orchestral-simulation machine praised its sound, but loud grumbles arose about many aspects of its operation.

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

It was hard to program new sounds (very hard if you wanted to do your own samples); off-loading sounds onto a computer was a tedious and unreliable process; the sequencer was small and clumsy; there weren't separate outputs for different instruments; the MIDI implementation was a disaster and only 12 voices were available.

Worst of all, the price was almost universally agreed to be too expensive, and Kurzweil was one of the only American synthesizer manufacturers in history actually to *raise* prices after a model's initial release.

Because of its price, the K250 was attractive to only a couple of markets: the rich hobbyist with $15,000 to blow on an elaborate, high-status, high-tech toy and the well-heeled production or recording studio with a need for convincing orchestral sounds—at any cost. Over the years, Kurzweil has addressed many of the K250's initial problems with software options and various add-on devices, but the instrument is still beyond the reach of most studios. Competition from companies like Roland, Korg, Ensoniq and E-mu Systems is closing the performance gap quickly.

The 1000 series

The 1000 series is Kurzweil's answer to the low-cost sampling synthesizers. It maintains the high quality of the K250's sounds, offers improved functions for modifying the sounds, substantially increases the number of available voices and brings the MIDI implementation out of the dark ages. And it does all this at one-sixth the price of the original.

What's missing is the internal sequencer, which most users will never miss, and the ability to sample new sounds, which is probably the most important contributing factor to the low price.

There are five models in the 1000 series: the K1000, a full-size keyboard instrument containing a collection of sounds that can be described as "Kurzweil's greatest hits," such as grand piano, orchestral strings, choir and baritone horn; the 1000 PX, a keyboardless rack-mount version of the K1000; the 1000 SX, a rack-mount that contains a wide variety of solo and ensemble string sounds; the 1000 HX, a rack-mount with brass and saxophone sounds; and the 1000 GX, a rack-mount with acoustic and electric guitar sounds. Their prices range from about $2,000 to $2,600.

Architecturally and operationally, the five models are essentially identical. Our review unit was a PX, but most of what follows applies to the other models as well.

Architecture

The 1000s are a cross between samplers and wavetable synthesizers. Each unit contains 50 "soundfiles," which are the basic building blocks of the sounds. A soundfile can be a sample (actually, most are multisamples, covering a range of four octaves or more), or it can be a digital representation of a useful wave, such as sine or sawtooth. Most of the samples are present in several variations, such as grand piano, hard-strike piano and mellow piano, which give plenty of variety.

The soundfiles reside in ROM, and they are what differentiate the various models. They can be changed only by swapping out or adding chips. Kurzweil has announced an optional upgrade for PX owners, with the installation of an extra ROM board (about $500) that will add a number of new Soundfiles, including flute, drum kit and Rhodes piano.

When a sound is generated, a soundfile is passed through a set of effects, such as chorus and vibrato, and then through a set of envelope and performance parameters, much like a conventional synthesizer. The result is called a "layer." Many of the stock programs consist of just one layer, but as many as four layers can be combined into a single program. Because different attack times, transpositions, velocity sensitivities and volume levels can be assigned to each layer, as well as different soundfiles and effects, complex split and layered programs can be created.

There is a price for using multiple layers, however. The *total* polyphony of the unit is proportional to the number of layers used in each sound. So while the 1000 PX has 24 voices available for single-layer sounds, only 12 voices are available for two-layer sounds, and so forth. (The HX, SX and GX models start with only 20 voices total.) Some of the effects, such as chorusing, create an extra "invisible" layer, so that just using one such effect can cut the total number of available voices in half.

Depending on the model, between 100 and 120 programs are provided in ROM, with 64 slots in non-volatile RAM for user-defined programs. (In an unusual approach, Kurzweil varies the size of its programs according to their complexity, and, thus, it is possible to create programs so complex that the program RAM is used up before all 64 slots are occupied.) Because the number of programs exceeds those available through MIDI program changes (128), a software map to assign the internal programs to incoming MIDI program numbers is provided.

The envelope stages

Generally speaking, each sample soundfile comes with its own native amplitude envelope, and when using the samples in their natural state, these envelopes do not have to be dealt with at all. However, envelopes can be imposed on the soundfiles so that new types of sounds can be created, and these envelopes can be very complex.

Up to seven attack segments can be described, each with a time (10ms to 10s) and level (0% to 100%) parameter. At the end of the attack segments is an Attack End segment that can be set to sustain, decay immediately or loop over the previous segments, either forward or bidirectionally. The number of segments that get looped and the number of times the loop occurs can be specified.

Following the attack segments (unless Attack End has been set for immediate decay), up to seven release segments are available, also programmable for time and level. At the end of the release segments is a Release End segment that can also be set to decay immediately or to loop—but not to sustain.

After all that is set up, you can modify the *overall* attack and release rates in real time over MIDI or use one of the 1000's internal control sources, which we'll address later.

Effects

Each layer of a voice can be assigned one effect. These come in two flavors: "compiled," which are factory-preset combinations of delays and LFOs, set up as vibrato, chorusing, echo and Leslie-type effects; and "modular," which are user-definable.

Modular effects let you change a program's sound in real time by specifying control sources to change various aspects of the sound. These sources include many MIDI commands: 64 different controllers, channel and polyphonic pressure, pitch bend, note-on and note-off velocity, note number and "key state."

They also include a number of internal functions: two "local" LFOs (which means they trigger with individual keystrokes); two "global" (constant) LFOs; two local envelope generators (which are separate from the ones described above); two global envelope generators; two inverters; two negators; the complex amplitude envelope described above; and two mixers for combining multiple control sources.

Control sources can be assigned to a number of destinations, which include the overall attack and release rates described earlier, pitch and amplitude modulation depth (there are two pitch modulators and one amplitude modulator per layer), and a control that lets you crossfade two layers.

A simple "modular effect" would be to assign MIDI modulation wheel (controller 1) to the depth control of one of the pitch modulators. But the control sources do not have to go directly to the destinations; much like the modules in an early analog synthesizer, they can modify *each other* long the way.

An envelope generator, for example, with its rising and falling segments, can be set up to control the rate of an LFO, which, in turn, controls pitch modulation, with the result that the vibrato rate of the sound evolves over time. Channel pressure can be sent to a slow LFO and then negated and sent to a second LFO so that it creates a phasing effect.

A square wave can be sent to a pitch modulator and that same square wave sent to a balance control so that a siren effect, which alternates channels, is produced. By using multisegmented envelopes and various delay and repeat functions, many note patterns can be triggered from a single keystroke, with waves evolving over time to other waves, changing in pitch and moving across the stereo field.

The modular effects can be so complex that it seems even Kurzweil's own engineers aren't sure how to take full advantage of them, and even the wildest factory presets use only a small fraction of their power. In some ways, they might even be considered overkill. In the LFOs themselves, the variety of waveforms that can be generated is positively staggering, but I doubt that many users will have much call for some of them, such as green noise.

Multiple timbres

A more immediately useful feature of the 1000 series is its multitimbral capabilities. Kurzweil virtually invented the concept of a multitimbral polyphonic instrument with the K250—the company even devised a new, unofficial MIDI mode called "Multi"—and the 1000s follow that philosophy with a vengeance. The 20 or 24 voices can be spread over as many MIDI channels as you like, with each channel set to its own program and capable of receiving individual controller and program changes.

Limits on the number of voices each channel can play can be set manually, or the instrument's built-in "stealing" algorithm can allocate voices dynamically. The stealing algorithm seems to be similar to the method used by the K250 (although it is not as programmable as it is in the older model), and is completely unobtrusive in normal use.

The multitimbral capability means that the device can be the equivalent of 16 separate polyphonic synthesizers. The only drawback is that 16 separate audio outputs don't exist; there are only two. Programs, or even individual layers, can be panned hard left or right so that they only show up in one output channel, and thus can be processed individually, but this is useful only to a point.

Also, hard-panning to one side neutralizes the unit's various stereo effects, such as the clever "auto pan" feature, in which the low notes of a program appear at the left output and, as you go up the scale, the image moves to the right.

But much of the inconvenience of cramming everything onto the same audio outputs is made up for by the total independence of the channels when it comes to incoming MIDI data. Each channel responds separately to controller No. 7 (MIDI volume), for example, so that mixing and balancing the voices can easily be done externally.

One software design mistake does exist, however, which shows up when you want to use the 1000 in conjunction with other MIDI synthesizers. In the "Multi" mode, you can program the unit not to respond to the MIDI channels your other synths are set to, but when you turn the unit off and then on again, it forgets which channels you have turned off, and so you have to go through the procedure all over again. Admittedly, it's minor, but it is annoying.

There are many other tricks you can do through MIDI. You can enable or disable sustain (controller 64), sostenuto (66) and Hold 2 (69, called "freeze" in the manual) pedals, and set the value by which the volume will be decreased (or even increased) when the Soft (67) pedal is pressed. A "keyboard tilt" feature assigns a progressive change in volume from one end of a layer's range to another, so, for example, the lowest keys can be 18dB louder than the highest ones, or vice versa. Note that all of these assignments are by *layer*, so within one program, a MIDI command can affect various parts of the sound quite differently.

You can adjust the velocity response of the instrument to whatever keyboard controller you happen to be using. Provided are several "velocity maps" for use with the K1000 keyboard; a Yamaha DX-7 (which doesn't cover the full MIDI velocity range); or a generic keyboard. You can also set up your own, either automatically—the unit prompts you for a "ppp" keystroke and then an "fff" one—or manually, in which case you assign a MIDI velocity number to each of eight dynamic levels.

In addition to the MIDI-receive program-change map described earlier, there is a MIDI-*transmit* map, in which the unit sends out a MIDI program change different from the one it has received. This can be helpful when you are using the 1000 with a MIDI effects unit that has fixed program locations and you want to associate a particular effect with a particular 1000 sound. Finally, a special combination of button presses turns the unit into a "Midiscope," and displays the last incoming MIDI event in terms of command type, channel and data values.

Alternative tunings

A fascinating feature of the 1000 series is microtonal tuning. Within an octave, each semitone can be detuned in values of a cent (1/100 of a semitone). Seventeen alternative tuning tables are provided in ROM, including various Arabic, Indian, Balinese, Pythagorean and Just tunings, and you can create your own and store them in non-volatile RAM. Although the default reference note for all tunings is C, that can be changed so you can move your alternative tonalities around to different keys freely. In fact, the reference note can even be changed *in real time* over MIDI, using the bottom octave of MIDI notes (0-11) to set it. No existing MIDI controller can actually play these notes, but many can be set to transpose down into that range. Unfortunately, separate tuning tables are not available for different instruments or channels; whatever the current tuning table is, the whole instrument uses it.

Using the 1000

With so much power packed into a small box and with a limited front panel, you'd think the 1000 series would be hard to use. In some ways, you'd be right. Fortunately, the stock sounds are superb, and I would imagine a good proportion of users will never feel the need to go beyond them.

Those who wish to delve into programming the device, however, will find a few confusing aspects. First, just getting to the right menu can be difficult. Taking an unfortunate lesson from the Japanese, Kurzweil has equipped the 1000's front panel with only eight multifunctional buttons, and a two-line LCD for moving through and editing a couple of hundred parameters on more than a dozen menus. (There is also an output-volume control, and the K1000 has a data slider.)

The difference between some parameters (such as local LFOs and global LFOs) is subtle, and yet must be thoroughly understood if the modular effects are going to be used intelligently. Two parameters, "Modulation Source" and "Modulation Depth," seem as though they should do exactly the same thing. Even the company's own technical support staff had trouble explaining the the difference between them.

Many functions appear in more than one place and can get in each other's way if you're not careful. For example, you can specify a pan position, both inside a particular layer and at the "channel editing" level. The latter can either take precedence over the former or not, depending on the setting of a "pan override" switch. The same pseudo-redundancy shows up with upper- and lower-note limits, volume controls, voice allocations and other parameters.

The manual

The manual, as is unfortunately typical for Kurzweil products, is a problem. Instead of providing an overview of the intricacies of the various menus, it simply serves as a slave to them. The main body is a "dictionary" of menu items, listing them in the order they come up, which is not at all the order in which you'd want to use them. There is no index, so if you want to find out about a specific function, you have to figure out what menu it's on and then find it. The various functions of the modular effects are explained fairly well, but no examples exist on how to use them.

When the original K250 came out, a Kurzweil employee and a user got together and wrote an article, which described the unit and contained an invaluable "road map" for the various menus. Someone should do the same for the 1000. (Actually, one of my college students last semester—who was working part-time in Kurzweil's warehouse—started such an article. He should be asked to finish it.) Kurzweil says a new manual is being prepared and may be out by the time you read this.

Kurzweil makes available, for a $50 charge, a Macintosh software program called Object Mover, which is useful for off-loading user presets onto disk and sending them back into the 1000. (It will soon be available also for Atari and IBM computers.) With only 64 RAM presets (or fewer) available within the unit itself, serious programmers will find this utility a big help. There are some editing facilities within the software, but they simply duplicate the unit's own front-panel control. They, therefore, represent no real advantage. Also, some more sophisticated functions are reportedly buried in the program, such as the ability to change volume curves and LFO wave shapes, but they are not documented.

Much talk has arisen, both within Kurzweil and without, of third-party developers writing sophisticated editing software, but so far nothing appears imminent. One developer reportedly made an attempt but gave up when confronted with the 1000's unique system-exclusive format.

Is it for you?

I'm glad to say that the sound quality, the main criterion for deciding whether one or more of these boxes should go into your studio, is just fine. The sounds are not identical to those of the K250, but they're very close. Some users have complained that the piano is a bit dull, but I think that's only because they haven't experimented enough with the alternative piano samples. There's plenty of bite in many of them. (There's even one particularly obnoxious program called "Chain Saw Piano"—so named because "it cuts through anything.") Long string pads are sometimes a problem because you can occasionally hear the sample looping. Used judiciously, however, and with some good processing, the sound is entirely workable.

Kurzweil set out to bring the quality of sound that its products are noted for into the smaller studio. With the 1000 series, it has succeeded admirably. Although the user interface is almost as confusing as the K250's, the combination of multitimbrality, superior MIDI implementation, excellent sounds, high potential for creative control and reasonable price makes the 1000 series all winners.

Within the limitations imposed by the single pair of outputs, they are tremendously flexible machines that can perform the work of a number of more-expensive devices put together. I am particularly looking forward to the new sound block for the PX, which will make it even more useful. Kurzweil's perseverance has paid off, and we are all the beneficiaries.

RE/P

Specifications

Note channels: *1000 PX: 24. 1000 GX, HX, SX: 20.*

Dynamic range: *100dB.*

Sample format: *16-bit floating point. 7X oversampled, with 20-bit sample accumulation. 17-bit D/A converters, processed with pre-/de-emphasis.*

Outputs: *line-level stereo.*

MIDI: *In, Out, Thru. Omni On, Omni Off and Multi Modes (independent program assignments for all 16 MIDI channels; Multi-timbral with dynamic voice allocation). Programmable destinations for all MIDI control sources.*

Programming RAM: *Battery-backed RAM for up to 64 user-defined programs. Programs may consist of up to four layers/splits.*

Front panel: *Eight multifunction buttons. Audio Output Knob. Power Switch. 32-character LCD.*

Programmable functions: *Two program editing levels: Compiled and modular effects for each of up to four layer/splits. Compiled editors include preset configurations of stereo chorusing, vibrato, tremolo, phasing, stereo Leslie simulation, echo delay, doubling and amplitude envelopes. Modular editors include two Local LFOs per layer; two Local ASRs per layer; two Global LFOs per program; two Global ASRs per program; two Envelope Generators with eight attack and eight release segments; loop functions; attack and release rate controls; two mixers for combining the signals of various modular effects; two inverters and two negators for inverting and reversing the function of a parameter; Pitch Control (source and depth); Amplitude Control (source and depth). Master parameters include MIDI Program Mapping, basic MIDI Channel, Master Tune, Transpose, Velocity Maps, Intonation Tables and MIDI Channel Override for Layer parameters.*

Computer interface: *Object-Mover program for off-line storage of RAM data on Apple Macintosh or Atari computers, including programs (patches), velocity maps, MIDI program maps and master parameters. Also provides graphic display of the 1000 series' front panel for remote control via keyboard or mouse.*

Dimensions: *17"x14"x3".*

Weight: *16 pounds.*

Consumption: *30W.*

[Editor's note: Specifications are taken from manufacturer's literature and have not been independently verified.]

Studio Policy, Rates and the "Arts"

By David Porter

Operating a studio is no different than running any other business except for the perception that the studio's business policies should not interfere with the "art."

Many of us enter the recording business with little or no business sense, armed only with the desire to make a living doing something we truly enjoy. We often make some bad assumptions about how to operate our businesses and suffer the consequences. This article will offer the studio owner/manager some guidelines based on data and experience derived from actual operations, combined with some hypothetical models to help explain how rates are arrived at, how billing procedures and policies are developed and, most importantly, how to get paid.

I'm always amazed when I visit my attorney's office and look at his capital investment as compared to my own. But what's more important is his return on investment as compared to my own.

In his office, you will find the obligatory law library, a sophisticated PC-based word processing and accounting system, some very nice furniture and a photocopy machine. The total capital investment for his office, for two attorneys and three staff members, is approximately $80,000, and his billable rate is $150 an hour. Amazing!

Determining studio rates is always a controversial subject, one that will often bring otherwise stable human beings to the brink of despair. If one is to use the attorney's office as an example, then the ratio of $80,000 yielding $150 and hour could be applied to a studio investment of $1 million, yielding $1,875 per hour!

Obviously, we can not use the legal profession as a guideline for establishing studio rates, but it should be noted that studio rates are often undervalued with respect to how capital-intensive they are. A more practical method with respect to what the market will tolerate is one based on the actual cost of operations including debt service and the desired return on investment.

The reality, of course, is that rates are most often a function of what the studio across town is charging and are not based on economic principles. My company, like most other studios, initially established rates based on what the competition was charging. We soon found that we were not making even a reasonable return on investment and set out to remedy the problem by analyzing our expenses and our historical, not hypothetical, studio usage.

We found that our studios, on average, billed 5.6 hours a day, based on 27 work days a month. From this calculation and the actual expense data, we determined what hourly rate provided the break-even point. We were then able to set rates based on our own profit goals rather than what the studio down the street was charging.

Table 1. Expenses expressed as a percentage of gross.

Advertising	1.5
Amortization	2.5
Auto	0.6
Depreciation	6.1
Entertainment/Promo	0.4
Equipment Rental (Leases)	4.6
Interest	1.8
Insurance (General)	1.6
Insurance (Health)	1.4
Janitor	0.3
Legal and Accounting	1.1
Maint. and Repairs	31.3
Officer Salary	4.8
Office Supplies	0.7
Postage-Freight	0.4
Rent	9.7
Taxes (Property/Business)	1.3
Taxes (Payroll)	2.1
Phone	3.8
Travel	0.3
Utilities	3.2
Total % of Gross	81.7

Determining expenses

Table 1 lists some sample expense guidelines. No two facilities have the same structure so keep in mind the percentages and ratios described are not necessarily applicable to all markets and situations.

A practical rule of thumb: Gross in 12 months what the studio cost to build. As an example, if the studio cost $250,000 to build, it should gross at least $250,000 in the following 12 months.

Conversely, if you spend $100,000 on new equipment, the gross in the following 12 months should show a $100,000 increase. We use the last five years of "cost-to-build" (equipment plus leaseholds) with no provision for depreciation as a basis for

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this calculation. If you are buying an existing operation, you might apply this rule to the purchase price. Did, or can, this operation gross in 12 months what you paid for it?

Example: Assume one room and a $250,000 investment and expenses are 81.7% of gross. Total monthly expenses are $17,020.

$17,020 divided by 27 days = $630.37/day, divided by 5.6 hours/day = $112.57/hour to break even.

Rate: A rate of $125/hour x 5.6 hours/day = $700/day x 27 days = $18,900 and yields a nominal 10% profit ($18,900 x 12 months = $226,800).

A rate of $150/hour x 5.6 hours/day = $850/day x 27 days = $22,680 and yields a 25% profit ($22,600 x 12 months = $271,200).

Try substituting your own historical data into the formula and test your profit margin. You may be surprised to see where the economic reality is and where the competition left off. If you are starting a new venture or buying an existing one, these models may prove useful to you when the "blue sky" approach to projecting revenue and income has you coming up with numbers that look too good to be true.

Billing and credit policy

We have all heard the phrase "credit is a privilege," but most of us make some fundamental assumptions about where credit is available. Do you expect credit at the department store without a charge card? Is there, then, an assumption that recording studios should, as a matter of course, extend credit without the same credit checks that Visa or Mastercard might run?

> ***We were able to set rates based on our own profit goals, rather than what the studio down the street was charging.***

A comprehensive credit report should include a minimum of three vendor evaluations, a report from a bank and information as to the structure of the company (sole proprietorship, corporation or partnership). See that they include their federal ID number, state resale license number, who the principals are, where they live and their social security numbers. Make sure to supply a bank information release form, to be signed by one of the principals, allowing the bank to give you information as to how they manage their account.

Keeping this information on file will be valuable to you in the event you need to take legal action. You may also request personal guarantees from the principals. It has been my experience that loose credit practices are often a result of desperation on the part of the studio manager to keep the studio filled with clients even when a client may be a bad credit risk.

What happens then when a session books on short notice and there is no opportunity to do a credit check? Just like the department store, you should demand payment either in advance or upon completion of the session, especially if the client is an individual.

Many companies use purchase orders when buying studio services. A PO may be for a predetermined amount, or it may have a "not to exceed amount." On some regular accounts, you may obtain a blanket PO that is good for a given period of time for ongoing projects. If you don't get one of these, be careful of small unauthorized amounts such as cassette copies and safeties that are requested by the producer. Unless they are paid for individually or you get it added to the PO, you may have a difficult time getting those charges paid.

It is advisable to call the purchasing department and ask what the policy is for processing your invoice. Sometimes at ad agencies, the line producer must have the creative director sign off on the invoice before the accounting department will make payment. If you are not familiar with their procedures, you may have to wait longer than your stated terms for payment. In other words, when the line producers says, "Don't worry about it, I'll turn it in for payment," you should be asking, "To whom?"

Find out the hierarchy of each company you extend credit to because many invoices may live in what I call "A/P purgatory" waiting for approval. This same problem occurs at record companies, as well. Here too, be especially careful when billing for services that exceed the initial purchase order agreement. It may be necessary for an additional PO to be issued to authorize payment of the excess charges. Many times, a well-meaning record producer has led the studio to believe the extra charges would be, according to Joe Isuzu, "no problem." Setting policy, therefore, is part of the economic reality.

Many studio operators, in highly discounted and competitive markets, are so thrilled to have any business at all that policy is often the last consideration. One area that is a perennial enforcement problem is getting the engineer to have the client sign the work order. This procedure becomes especially important in disputes regarding start and stop times, setup charges or equipment rentals.

The problem increases exponentially as the engineer and client become more familiar with each other. Engineers need to be reminded that your ability to pay them is based on the studio charging and getting paid for all the materials and services it supplies. Work orders that are filled out the next day from memory with no client signature are useless in the event there is a dispute.

Cancellations

Charging for a cancellation is another area where a dispute may arise. Do you enforce the cancellation policy on a long-time client? That long-time client may be habitually booking and cancelling studio time with little or no notice. Do you get tough and risk losing him, or do you enforce policy and charge for the time booked? A meeting with the studio manager or a published policy document followed by a letter of warning may need to be sent. Sometimes that "good client" can be hurting you more than you think.

Operating a studio is no different than running any other business except there is a perception that the studio's business policies should not interfere with the "art." It is the true professional who can implement good business policy while maintaining an atmosphere that is conducive to the creative process. RE/P

Sample business letters

Our company uses the following standardized letters when establishing credit with potential customers. If you are not using something like these letters, you should seriously consider doing so. Having standardized letters makes the process much easier on you, and ensures that you have the same information on everyone.

Letter No. 1 is addressed to the accounting department, and requests credit information for your potential client:

"Recently, I received an application for credit from (insert name). They have listed your company as a credit reference and have authorized us to look into their account records. I would appreciate it if you could fill in any information that pertains to this company. A self-addressed envelope is enclosed for your convenience.

"Thank you for your help."

The form that we enclose includes the following questions:

When was account established?
Sales terms?
Discounts?
Recent high credit?
Credit limit?
Current balance?
Current past due?
Date of last purchase?
Last payment?
Are payments: Prompt, 30-45, 45-60, Over 60?
Any problems?
Recommend this account?
Comments.

Letter No. 2 is sent to the client to whom you have decided to establish credit:

"Thank you for your recent application for credit. We are happy to inform you that your request for an account has been approved and a credit limit of (insert amount) has been set.

"Our terms of sale are net 30. No discounts are allowed. A service charge of 1½% per month will be added to all accounts after 30 days.

"We appreciate your business and look forward to serving you in the future."

Letter No. 3 is used if you deny someone credit. With this letter, be especially diplomatic.

"Thank you for your recent application for credit. We appreciate your cooperation in taking the time to complete the application. Unfortunately, we are unable to extend credit to you at this time.

"We look forward to doing business with you in the future, but ask that all studio sessions and all duplication jobs be paid in full upon completion. As you continue to work with us, we may establish a line of credit on a trial basis. If this term of trial basis becomes acceptable, we will establish an account for you with 30-day terms.

"Thank you for your cooperation."

Letter No. 4 asks for a state reseal card and goes to the client's accounting department:

"We currently do not have a state reseal card on file for your company. Please take a few minutes to complete the enclosed card and send it back to us in the envelope provided. Without it, your file with us is incomplete, and we will be required to charge you sales tax.

"Thank you for your cooperation.

"This does not include tax applicable to materials."

An important reminder: the key to determining and granting credit is to have a polite, but firm tone. State your business as concisely as possible and finish.

STUDIO UPDATE

Talkback

Restoring Old Records

A three-step method to restoring old records. This sequence of photos shows a record with a paper sticker over certain cuts being readied for radio airplay. The method also works for restoring records for a variety of uses. Upper left: Wash the record in warm, soapy water. Upper right: The paper has been removed, but the glue remains. Lower left: Spray WD-40 over the affected area, and work the glue loose using a brush. Lower right: Wipe away the WD-40 and apply warm soapy water using a circular motion.

A chief engineer at a San Diego radio station has stumbled on a way to resurrect old records. Although primarily beneficial for radio stations, the method is helpful for engineers in a variety of applications, such as archiving sound effects from album to a digital format.

The engineer, Jack Dobbs at KSON AM/FM, needed a way to restore album cuts that once were banned but now are considered classics. At the time, the offending cuts were covered over with adhesive stickers. Through the years, the glue hardened, and now had to be removed without damaging the record. After testing several cleaning products, Dobbs discovered a three-step process using WD-40.

1. Apply warm soapy water to lift the sticker's paper off the glue base. Using a clean cloth, gently wipe away the loose paper.

2. Spray the glue area with WD-40, let it soak for a few minutes, then work the WD-40 into the glue using a toothbrush or hard nylon or plastic brush. According to Dobbs, this will not harm the record and is often necessary to work the glue loose.

3. With a clean cloth, wipe away the WD-40, then apply warm soapy water into the record grooves using a circular motion.

Dobbs adds that WD-40 can also be used as a record restorer, enhancing the quality of the music, reducing unwanted noise and making the music suitable for professional use. RE/P

STUDIO UPDATE

Studio News

Northeast

John Alberts, audio mixer at **Howard Schwartz Recording** (New York) was awarded two 1988 International Teleproduction Society International Monitor Awards for audio work. In the entertainment specials category, he won for "Bette Midler's Mondo Beyondo Show." In the musical performance category, he won for "Graceland: The African Concert." *420 Lexington Ave., #1934, New York, NY 10170; 212-687-4180.*

Sound On Sound Recording (New York) has installed Digital Creations' Diskmix Moving Faders automation system on its Neotek Elite console. Other equipment additions include an AMS RMX-16 reverb, AMS DMX-15-80S delays, an Atari 1040ST computer with Creator sequencer software, E-mu SP-12 drum machine and an Akai S900 sampler. *322 W. 45th St., New York, NY 10036; 212-757-5300.*

Southeast

Disney-MGM Studios (Orlando, FL) has announced two appointments. Ted Kaye has been named vice president of film and tape production. Robert M. Allen has been named director of film and television productions.

Memphis Sound Productions (Memphis, TN) has purchased a Neve V Series console with Necam 96 automation. *315 Beale St., Memphis, TN 38103; 901-525-5500.*

Miller Recording (Starkville, MS) has expanded to 16 tracks with the addition of a Sony MCI JH-16 recorder and Neotek Series II console. *Route 5, Box 447, Starkville, MS 39759; 601-323-0728.*

Midwest

Master View Recording (Otsego, MI) has announced two appointments. Richie Dekker has been named chief engineer, and Brent McDonald has been named music production coordinator. *2236 Jefferson Road, Otsego, MI 49078; 616-694-6322.*

Freedom Sound & Light (Owatonna, MN) has expanded to 16 tracks. New equipment includes a Tascam MS-16 recorder, Tascam AQ-65 autolocator, Soundcraft 600 24x8 console, and an Atari 1040ST computer. *1520 Ninth Ave. NE, Owatonna, MN 55060; 507-451-9085.*

Sound Images (Cincinnati) has named Terry Dean as vice president of sales for New England and the Middle Atlantic. The facility hopes to expand its services to include East Coast clients, and Dean will head an East Coast sales office based in New London, CT. *602 Main St., Cincinnati, OH 45202; 513-241-7475.*

Southwest

Omega Audio and Productions (Dallas) has celebrated its 15th year in business. *8036 Aviation Place, Dallas, TX 75235; 214-350-9066.*

Southern California

Intercut Editing Studios (Hollywood) has announced a new Betacam-to-1-inch CMX A/B roll day rate, which also includes Chyron, DVE and add-ons. The facility added the rate to allow clients more flexibility and creativity without sacrificing quality because of budget considerations. *6363 Sunset Blvd., Suite 716, Hollywood, CA 90028; 213-466-0104.*

Editel/LA (Los Angeles) has promoted Bill Frazee to vice president/general manager.

Devonshire Audio/Video Studios (North Hollywood) has installed a Neve V Series 60-input console with GML automation in Studio 3. Studio 4 has been updated with a Neve 8128 56-input console with Necam 96 automation. Both studios have custom George Augspurger monitoring systems. *10729 Magnolia Blvd., North Hollywood, CA 91601; 818-985-1945.*

Northern California

Savage Studios (San Francisco) has recently opened 24-track facility. Construction features a spacious control room and variable studio acoustics. Steve Savage is the owner. *372 Brannan St., San Francisco, CA 94107; 415-546-1374.*

Mobius Music (San Francisco) has installed a Neve 8068 console in its recently remodeled studio. *1583 Sanchez St., San Francisco, CA 94131; 415-285-7888.*

Dave Wellhausen Studios (San Francisco) has purchased an E-mu Systems Emax with an internal hard drive and a library of more than 3,000 voices. *1310 20th Ave., San Francisco, CA 94122; 415-564-4910.*

Northwest

Spectrum Sound Studios (Portland, OR) recently shared three IBA awards for work completed with the Spaghetti Factory, Portland Adventist Medical Center and agency Wagner, Wiecks, Smith & Lapel. The studio also received two Telly Awards and a Clio for work on Norm Thompson with Brown-Dugan. *1634 SW Alder St., Portland, OR 97205; 503-248-0248.*

Manufacturers' announcements

Soundmaster International has installed Soundmaster Integrated Editing Systems at ABC TV, Hollywood; Cherry Beach Sound, Toronto; Cinar Films, Montreal; Magno Sound & Video, New York; Mix Magic, Los Angeles; Sound Venture Productions; Ottawa, Ontario; Video One, Boston; Weddington Productions, Hollywood; and Zenith/dB Studios; Chicago.

AKG Acoustics has sold ADR 68K reverbs to Yale University School of Music, New Haven, CT; Lakewood Studios, Jamesville, NY; and Alpha Star Studios, McKeesport, PA.

Sony has installed an MXP-30306 console and two APR-5002 recorders at the University of Miami's recording studio in Coral Gables, FL. RE/P

NEW PRODUCTS

Analog Solutions 18-bit D/A converter chip

The ZDA1801A provides three functions: a digital/analog converter, a deglitcher and reference function, simplifying the design and manufacturing of devices needing an 18-bit chip. The built-in proprietary deglitcher circuit offers guaranteed front-to-back THD+N of 98dB, from 20Hz to 20kHz, and the chip is able to perform 4x oversampling. The module is packaged in a electrostatic and electromagnetic shield; dimensions are 3"x1.6"x0.45".

Circle (176) on Rapid Facts Card

Amek ESM32 interface

Designed for Amek's BCII consoles, the interface allows external control from all major video editing controllers and switchers through RS-422 or -232 connectors and cable. The rack-mount unit plugs between the video editor and the console. Audio levels from the BCII can be assigned as crosspoints from the host editor, with communications between the two via ESAM II, the serial interface protocol used by Ampex, CMX and Paltex editing systems. A reprogrammed version of the unit can be used with Grass Valley and FOR-A systems.

Circle (177) on Rapid Facts Card

Tannoy AV Pro monitors

The AV Pro is a reference monitor 15 inches tall and is constructed around heavy vertical and horizontal internal bracing. Using Differential Material Technology, the monitor's high frequency unit has deep drawn duralumin diaphragms and skirt with a separate silicone based suspension, giving the piston rigidity associated with titanium but without HF breakup modes in the pass-band. Magnetic flux density is less than 1 milli-tesla outside the cabinet, making the monitor suitable for use with video equipment.

Circle (178) on Rapid Facts Card

Carvin EQ2029 graphic EQ

The 2029 is a ⅓-octave graphic equalizer that provides smooth band-pass filtering for low distortion and transparent operation. The 29 bands are centered on standard ISO frequencies, and the unit has switchable ultrasonic and subsonic filters to eliminate ultrasonic noise, oscillations, stage rumble and wind noise.

Circle (179) on Rapid Facts Card

Digital Music Services DMP11 PRO software

DMP11 PRO is an interactive intelligent mixing assistance program for use with the Yamaha DMP 11 digital mixer and an Apple Macintosh. The program graphically displays setting for every mixer parameter, and functions can be controlled with the Mac mouse or from the DMP11. Also included are functions not found on the mixer, such as automation assistance, solo and muting functions, naming of channels, disk storage of mixes, and filtering or adjusting of MIDI sequences. The program is compatible with MultiFinder or any MultiFinder-compatible sequencer and can run simultaneously with it on the Macintosh, eliminating the need for an additional sequencer.

Circle (180) on Rapid Facts Card

D&R Stylyx console

The Stylyx starts with a basic frame and can be customized to fit individual needs using various modules, including mic/line, stereo line, subgroup, master, meter bridge, rack-mount power supply and automation. The automation package is a soft muting system that can be manually, sequencer- or time code-controlled.

Circle (137) on Rapid Facts Card

Neutrik plugs and jacks

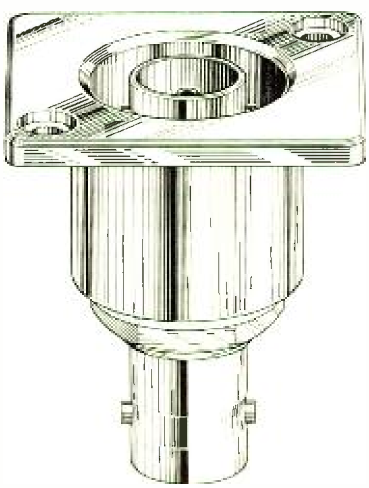
Connectors the company recently has introduced include Bantam plugs using hard plating material formulation, ¼-inch-long frame phone plugs with solid brass shells and a latching phone jack available in either chassis-mount or cable-connect.

Circle (111) on Rapid Facts Card

AKG D90S, D95S dynamic mics

The 90 cardioid and the 95 hypercardioid both are designed for studio and sound reinforcement purposes. Both come with integrated wind and pop filter, lockable on-off switch and polar response pattern. Frequency range is 70Hz to 18kHz; SPL is 130dB; and sensitivity is 1.3mV/Pa.

Circle (102) on Rapid Facts Card



Canare 75Ω flush-mount BNC connectors

The new line features a recessed design that protects the housed connector, allowing users to punch one hole for audio and video panel mount connectors. Other features include low VSWR characteristics, correct impedance match when using 75Ω coax cable and compatibility with other BNC connectors.

Circle (118) on Rapid Facts Card

Audio/Digital PAD-300/18 delay system

The system, said to be the first 18-bit delay for alignment and synchronization applications, features a dynamic range of 105dB. Other features include one audio input and three audio outputs, RS-422 ports, front-panel bypass control, and failure detection and reset/bypass control. Delay range is 0ms to 650ms, in 15μs steps; frequency response is 10Hz to 22kHz, and distortion is less than 0.01% at 1kHz.

Circle (105) on Rapid Facts Card

Bryant Electric studio wiring

Studio-Tech is a line of NEMA-configuration wiring devices for studio and stage uses. Features include color coding, nylon materials, special wire adapter sleeves and locking devices.

Circle (115) on Rapid Facts Card

MicroAudio MIDI POD ⅓-octave equalizer

MIDI POD is a 28-band, ⅓-octave equalizer programmable by MIDI controllers. Taking up a single rack space, the unit has no front panel controls and features a 100-year, non-volatile memory with no internal batteries. It uses combining filters to give a smooth response and a noise floor greater than 90dBm.

Circle (116) on Rapid Facts Card

Sound Genesis "Essential Percussion," Vol. 2

For use with the Fairlight, the sampling collection contains four 60-Mbyte data streamers of percussion samples. Each sample has been tuned, mapped to the keyboard, assigned function curves, and tested for consistency of color, tonal quality and sonic realism. The company adds that all sounds are fully protected, eliminating the threat of copyright infringement.

Circle (184) on Rapid Facts Card

MORE Musician's Organizer

Musician's Organizer is a personal organizer designed for musicians and other professionals in the music/production industry. Sections include a monthly calendar, personal resources, player resources, itinerary planner, notation writing pad, venue resources, instrument inventory, studio resources, floppy disk holder and a zip lock envelope.

Circle (185) on Rapid Facts Card

Korg Q1 MIDI workstation

The Q1 16-track/16-channel, MIDI/-SMPTE multitrack sequencer acts as a full-function MIDI control center and has a 16-bit high-speed CPU and 512Kbyte of RAM. Two mergeable MIDI Ins can record data onto any channel of the 16 tracks. Sequences can be recorded in Patter mode (drum-machine style) or in continuous play/record mode (tape-recorder style). Two separate MIDI transmitters and two pairs of parallel MIDI outputs allow users to address up to 32 channels.

Circle (114) on Rapid Facts Card

Audio-Technica 900 series headphones

Three units make up the series. The ATH909 and ATH911 are open-back designs allowing users to hear outside sounds while monitoring program material. The 910 has a closed-back design that allows on-mic operation at high levels without feedback. Specs include 96dB sensitivity, 4Ω to 24Ω matching impedance and 30Ω actual impedance. Frequency response is 20Hz to 20kHz for the 909 and 910, and 20Hz to 25kHz for the 911.

Circle (109) on Rapid Facts Card

Gefen Systems cataloging software

The company has released two programs for Apple Macintosh computers for cataloging and searching production-music and sound-effects libraries on compact disc. The programs integrate the listing catalogs and offer selection in one location. Retrieval can be by key words, category words and synonym descriptions. Interfacing with a multiCD changer to audition and play the desired selection is also possible. Hardware requirements are 128K ROM, 1Mbyte of on-board RAM and two floppy drives.

Circle (123) on Rapid Facts Card

NEW PRODUCTS

Publications

SSL G Series Computer manual

The operator's manual is designed for both new users and those who have already worked with the computer system. Included are explanations of the concepts and operations of earlier versions of the computer and an explanation of new operations specific to the G Series. Sections include basic computer operation, the mix system, Total Recall, the Synchronizer system, programmable equalizer and references.

Circle (132) on Rapid Facts Card

Klark-Teknik audio system designer handbook

Titled "The Audio System Designer—A Technical Reference," the handbook contains more than 100 pages of technical data needed for maximizing the performance and characteristics of audio installations. Sections include general acoustics, sound insulation and absorption, room acoustics, psychoacoustics, speech intelligibility, sound system engineering and physical data.

Circle (133) on Rapid Facts Card

Sencore Electronics Tech Tips

Tech Tips are technical bulletins designed to provide information on Sencore products and also to provide new troubleshooting applications. Topics include microprocessor and VCR troubleshooting, and capacitor testing.

Circle (135) on Rapid Facts Card

3M Audio Setup Card

The card is a laminated reference that lists optimum bias selections and operating levels for all 3M analog mastering tapes. Included are graphs, cross-reference charts, written explanations, and information on proper bias settings, speed, record head gap and 10kHz overbias information.

Circle (136) on Rapid Facts Card

Solid Support Industries tape recorder stand

The TR-10 is designed for rack-mount tape recorders and features a two-position top piece that allows for recording or editing positions. The top piece is 14 rack spaces high with an open back, which allows larger machines to fit within the confines of the rack. The stand is constructed of 1″×2″ heavy-wall steel and comes with four casters (two of them locking). Overall height is 30 inches.

Circle (119) on Rapid Facts Card

Fluke TL20 test lead set

The TL20 features long, heat-resistant test leads, interchangeable safety-designed alligator clips with retractable jaws and stainless-steel needle-point test probes. The set consists of black and red pairs of alligator clips, needle-point probes and test leads with right-angle connectors at both ends. Test probe tips are 19mm long and have sharp points for piercing insulation coatings.

Circle (103) on Rapid Facts Card

Digidesign Turbosynth software

A Macintosh program, Turbosynth performs a variety of digital-synthesis and sample-processing functions using a graphically oriented user interface. The program produces 16-bit samples that can be transferred to most brands of samplers for performance. Users design sounds by using a palette of modules, any number of which can be placed on the screen and arranged into any configuration.

Circle (112) on Rapid Facts Card

Passport Designs MIDI Transport

Passport's MIDI Transport is a Macintosh interface that bridges MIDI and SMPTE for recording applications. For sequencing, the program provides two sets of MIDI inputs and outputs, one set running from the printer port and the other from the modem port. For studio work, the SMPTE read and write capability enables sequences to lock to audio or videotape. The transport sits beneath the computer and provides modem and printer through ports and corresponding front panel switches to enable modem or printer operation without reconnecting anything.

Circle (113) on Rapid Facts Card

Keyboard Technologies GZ keyboard series

According to the company, the GZ-1000 is the first electronic keyboard to contain true piano hammer action. Percussive key sensors and the MIDI operator control system allow users to design multiple keyboard response parameters simultaneously transmitted over MIDI. An edit footswitch transforms the keyboard into a control panel for data entry. Also included is a 3.5-inch disk drive to dump and save MIDI setups and patches.

Circle (107) on Rapid Facts Card

Celestion SR series

The SR series is a sound reinforcement system with an electronic controller. A single-driver design produces full-range frequency response with no intermodulation distortion. Phase and time coherence problems are eliminated because there are no separate tweeters, compression drivers and crossovers. The controller monitors amplifier distortion, voice coil temperature and cone excursion.

Circle (100) on Rapid Facts Card

Otari T-650 R-DAT loader

Said to be the first fully automated R-DAT cassette loader, the unit winds, cuts and splices 0.15-inch metal particle or barium-ferrite pancake onto empty, preleadered, R-zero cassettes. A rotary-type splicer provides consistent, precision splices with no overlap. Cycle time is less than 39 seconds at a winding speed of 8 meters per second for a two-hour tape. Loading speed is variable and provides three preset positions.

Circle (122) on Rapid Facts Card

Yamaha FMC1 digital format converter

The unit is a dual-channel, digital-audio-format converter that transforms the Yamaha-proprietary format to the three commonly used digital audio formats: unbalanced SDIF-2 (Sony), via three BNC connectors; CD/DAT (S/P), via an RCA-type pin jack; and AES/EBU, via an XLR connector. The unit converts the digital outputs of the company's DMP7 and DEQ7, which allows it to be used in existing digital recording and processing systems. One rack space high, the converter includes an on-board master word clock, switchable to 44.1kHz or 48kHz.

Circle (117) on Rapid Facts Card

Words & Deeds studio management program

Archie is a recording studio software system that automates studio paperwork. Functions include tracking clients and producers, controlling inventory, creating tracksheets, generating instant timesheets, producing invoices and statements, analyzing studio performance, and tracking payables and receivables on a 30-, 60-, and 90-day basis. The system is designed to run on Macintosh computers with more than 1Mbyte of memory and a hard disk.

Circle (125) on Rapid Facts Card

FANSIM software from TUTSIM Products

The program provides frequency analysis of open-loop and closed-loop response, finds transfer functions of real or simulated systems and finds poles and zeros. Users can take real, simulated or internally synthesized data to find overall frequency response. System requirements are IBM PCs or compatibles with at least 330K of RAM (512K recommended). The program supports CGA, Hercules and color EGA cards.

Circle (121) on Rapid Facts Card

Southworth Max Audio processing cards

Three cards, all using the Motorola 56000 signal-processing chip, have been released for use with the Macintosh II NuBus system. The Max Audio Analog Card performs all A/D and D/A conversions using a 20-bit proprietary converter. The Max Audio Quad 56000 DSP Card provides additional signal processing for reverberation and effects processing, frequency domain audio processing, sample playing and additive synthesis. The Digital Audio/SMPTE Card sends and receives data in the AES/EBU format and provides a SMPTE time code reader/generator.

Circle (120) on Rapid Facts Card

Sunn SPL series monitor systems

The SPL1282 and SPL1285 employ vented-baffle, low-frequency sections for clear bass response and a HF horn for smooth reproduction and controlled dispersion of high frequencies. The SPL1282 has a 12-inch woofer with 2.5-inch voice coil; the SPL1285 has a 15-inch woofer with a 3-inch voice coil. Both feature ribbon-wire voice coils and polyimide Kapton voice-coil bobbins with cast alloy baskets.

Circle (106) on Rapid Facts Card

NEW PRODUCTS

Studer A730 CD player

The unit allows direct access to track and index, minutes, seconds and frames, as well as elapsed and remaining track time. The Autocue feature determines and stores both start and end of modulation information. Also included is a disc recognition feature, using non-volatile memory for up to 100 CDs. It can be used either on a desktop or flush-mounted.

Circle (101) on Rapid Facts Card

Korg M1 Music Workstation

The M1 is a 61-note digital synthesizer that the company says is the first instrument to use all-digital processing with high-resolution sampled sounds, which enables it to combine sampling and synthesis technologies in a single workstation. A total of 4Mbytes of 16-bit ROM-based sounds are available, with additional libraries available on ROM or PCM cards. An eight-track sequencer is included, with room for 10 songs, 100 patterns and 7700 notes.

Circle (182) on Rapid Facts Card

Audio-Technica AT4051 mic

The 4051 is a cardioid mic that is transformerless and externally polarized, with interchangeable head capsules to meet a wide range of micing applications. The polar pattern can be changed by replacing the unit's element with the A-T 4049 omnidirectional capsule or the 4053 hypercardioid capsule. Other features include an 80Hz high-pass filter, windscreen and turned brass housing plated in black chrome.

Circle (183) on Rapid Facts Card

Fane Acoustics coaxial speakers

The CX10 and -10PA incorporate a two-way, full range transducer that incorporates the bass driver and a concentrically mounted, independently driven tweeter in a signal unit. The design allows the speakers to be suitable for both stage and studio use, according to the company. Power handling for the CX10 is 150W into 8Ω, and 200W into 8Ω for the CX10PA.

Circle (187) on Rapid Facts Card

Orban 787A mic processor

The 787A is a programmable mic processor designed for mic or line-level applications where the ability to store and recall processing is needed. The unit features a three-band Constant-Q equalizer, compressor with adjustable release time, de-esser, noise gate and/or compressor gate, and effects send and return ports. Control setups are stored in 99 memory registers. A Jensen transformer mic pre-amp with 48V phantom power is optional.

Circle (189) on Rapid Facts Card

JBL G-718 subwoofer

Featuring the JBL E115-4 18-inch transducer, the G-718 is designed to be added to a sound reinforcement system as a subwoofer to augment bass performance. The unit is rated at 600W continuous, and the vented-box enclosure is constructed with high-grade multi-laminate cross-grain material. Dimensions are the same as the G-732, 33"x24¾"x17¾".

Circle (190) on Rapid Facts Card

Wheatstone TV-500 console

Designed to address the stereo routing requirements for television, the unit includes four stereo subgroup buses, two separate stereo master buses, a mono bus for SAP and mono sum outputs. Four stereo aux buses are also included for foldback, mix minus and special effects. Mainframe configurations range from 16 through 56 inputs.

Circle (104) on Rapid Facts Card

Hardware, software updates

Retrofit for Korg DSM-1

The DSM-HDK retrofit allows an SCSI interface port to be connected to the DSM-1, allowing sample data to be loaded and saved to hard disk drives. With one or more SCSI-compatible hard disk drives connected, full megaword DSM-1 systems can be loaded and saved in less than 14 seconds. The retrofit, which is available from authorized service centers, supports up to seven drives, and has been tested with Macintosh-compatible 20-meg drives from General Computer and CMS.

Circle (200) on Rapid Facts Card

Passport update of Master Tracks Pro

Version 3.0 now syncs to MIDI Time Code and SMPTE time code as well as MIDI Song Position Pointers. A SMPTE window provides continuous time-code readout while synchronizing sequences to audio or videotape. For audio-for-video work, the Marker window now displays each marker's song position and SMPTE time, allowing it to act as a cue list that can be printed out. The update is available for the Macintosh, Atari ST and IBM PC computers.

Circle (129) on Rapid Facts Card

New accessory for AKG 68K

AKG has introduced a Master-MIDI ROM cartridge for the ADR 68K, which consists of 50 presets, each containing all the settings necessary to use the full power of one of the unit's programs and MIDI control features. With the cartridge, users can create a useful MIDI setup quickly, without a lengthy learning process. Examples include keyboard notes trigger samples, modulation wheel controlling DDL feedback gain and aftertouch (channel pressure) controlling chorus rate and depth.

Circle (130) on Rapid Facts Card

Hardware, software updates

Ampex 672 industrial audio cassette

The cassette now features an improved tape formulation that delivers superior frequency response and higher output, according to the company. The tape also has a new pad design that improves azimuth tracking, and a color-coded leader tape is included in bulk product, allowing easy play length identification.

Circle (191) on Rapid Facts Card

ART ProVerb memory update

The ProVerb upgrade allows owners of older units to upgrade memory capacity from 100 presets to 200, and also includes 90 new reverb presets. The retrofit involves adding an auxiliary circuit board and an EPROM that replaces the RAM system with a RAM chip with a built-in battery backup board. Units must be returned to ART to be updated.

Circle (192) on Rapid Facts Card

Graphic Notes software upgrade

Version 1.1 of the company's Music Publisher program has been released, which allows users to play back the compositions on MIDI devices and also accepts input from MIDI keyboards. Also upgraded are part extraction and transposition functions. For use with Macintosh computers, the upgrade is free to registered users.

Circle (131) on Rapid Facts Card

Sunn SPL 7000 power amp

The SPL 7000 has received UL approval, according to the company. Features include combination circuit breaker/on-off switch, efficient cooling system and logic circuits in each channel that protect the unit from improper loads.

Circle (126) on Rapid Facts Card

RAM cartridge for Kurzweil 250

The new RAM cartridge enhances the 250's user memory facilities, and contains ¼-megasample of RAM for up to five seconds of storage at 50kHz, 10 seconds at 25kHz or an addition 30,000 notes of sequencing storage. The cartridges fit into the slot directly beneath the keyboard in the center of the machine.

Circle (195) on Rapid Facts Card

General Devices cooling module

The company's redesigned D-4118 cooling module now has finger guards. The module fits a 19-inch cabinet or rack and contains nine fans, all of which are user controlled. Other features include an aluminum extruded frame, black enamel front panel and circuit breaker protection.

Circle (196) on Rapid Facts Card

Passport Master Tracks Jr. for Apple IIGS

The sequencer program is now available for the Apple IIGS, and it includes the same features as the Macintosh and Atari ST versions, including 64 recording tracks, a Conducter track for continuous tempo control, automated punch-in and punch-out recording, and tape recorder-style controls.

Circle (128) on Rapid Facts Card

Software enhancements for dbx RTA-1

Version 1.5 of the professional real-time analysis system includes enhanced room-response curve capabilities, improved microphone calibration and customized printouts. The price has also been reduced, and present system owners may upgrade by retrofitting the new software.

Circle (127) on Rapid Facts Card

Digital Music Services software updates

The company has released new versions of the Macintosh editor/librarian software for the Yamaha TX81Z and TX802. TX81Z PRO 2.0 and TX802 PRO 1.1 feature faster running times, MIDI echo options and new multiple selection features. PRO 2.0 also has an on-screen graphic envelopes and an expanded utility editor. PRO 1.1 now reads Opcode format files.

Circle (197) on Rapid Facts Card

JL Cooper update of MSB Plus

Revision 2.0 of the 8x8 MIDI switching box includes a programmable program change manager, a program advance mode capability using the Panic Button foot switch jack and the ability to initiate a sysex dump from the front panel. For existing owners, the revision can be purchased from local dealers. For new owners, the suggested retail price has been reduced.

Circle (198) on Rapid Facts Card

New software for Opcode Timecode Machine

Called Timecode Machine Desk Accessory, the new software is a rewrite of the Timecode Panel software and allows users to stripe a tape with any format of SMPTE, test an existing track of time code, display the current DIP switch sittings on the Timecode Machine, and display the version number of the ROM software. On-line help instructions for all procedures are also included. The software is free to all Timecode Machine owners.

Circle (199) on Rapid Facts Card

ART RS-232 interface option

The RS-232 option allows the company's ⅓- and ⅔-octave IEQ Intelligent Equalizer system to be connected to external computers. The interface is available as retrofit package for user installation or can be factory fitted.

Circle (193) on Rapid Facts Card

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