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STUDER REVOX

Circle 10 on Reader Service Card

MARCH/APRIL 1986 VOLUME 20 NO. 2

the sound contracting engineer

the broadcast engineer

2 to 8 the smaller recording studio

columns

• Once again, we visit Boston's WGBH, this time to learn about a live Boston to Tokyo video and stereo audio transmission. Our cover shows a Studer 169 mix console and the picture monitor. FM Tokyo engineer Kenichi Matsuda is at the controls. The photo is by the article's co-author Jim Voci.

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Editorial

THIS IS THE ISSUE that some of you will be seeing for the first time, based on a copy you picked up at the NAB Convention.

Since *db, The Sound Engineering Magazine* is a magazine for professional audio engineers, we, of course, consider the broadcast audio engineer our reader as well. And many are indeed just that. If you are new, we hope you'll be joining us too.

In this issue, read how Boston's WGBH, working closely with FM Tokyo, beamed the Boston Symphony performing live to Japan whose population also saw it live in living color and in high quality stereo sound.

Wade Bray's article shows how an architecturally bad broadcast room can be made good sounding using modern electronics to do the trick at reasonable cost. This article was originally scheduled for our Jan/Feb issue, which was guest-edited by columnist Jesse Klapholz, but was unfortunately delayed.

In this issue, Jesse has submitted an article that details some of the engineering facts about MIDI. Since MIDI and SMPTE interfacing are so much a part of the broacast audio scene today, and will be more so tomorrow, read on.

In his column, Bruce Bartlett explores SMPTE synchronizing and what it does. And there's more.

In a major article, author Larry Oppenheimer tells all about the MIDI interface, what it is, what it does, and what it cannot do. If MIDI is something new to you, by the time you have finished this issue, you'll be a knowledgeable expert.

To tie things together, Larry Oppenheimer returns with a second article that explores the MIDI and SMPTE interfacing that noted sound designer Frank Serafine must go through when making sounds for the movies and commercials.

In *db, The Sound Engineering Magazine* you can depend on finding out the most about the latest, first. **LZ**

Barry Blesser

The Next Brave New World

• Once upon a time, when life was simple, there lived a group of engineers who liked to listen to music. These engineers were nice people and they liked to build things in their basement to improve the quality of their hi fi set. Some of these engineers both loved music and were good engineers; their designs were useful enough to start a simple company to make equipment for other hi fi enthusiasts. Life was simple. The engineer had a few boxes of electronic parts, a slide rule, and a pad of paper. He would sit in his laboratory playing with the parts, trying this and that. At times the results were impressive; often they were not. Out of this process came the professional audio engineer.

I can remember my apprentice experience. As an undergraduate at a technical university, I was given the task of replacing all of the old electrolytic capacitors in an old Ampex tape recorder. A year later, I was allowed to rebuild the master console. My first design was a discrete six transistor pre-amplifier. The evolution of my professional training went step by step. I personally did not experience the impact of the dramatic transition from a vacuum tube mentality to a semiconductor mentality because I was born into the semiconductor age.

Technology has a tendency to make lots of little changes which are easy to follow; however, at times there are dramatic changes which completely change the nature of the activity. Tubes to transistors was such an example. Transistors to large integrated circuits was another example. Today we are in the beginning stages of yet another revolution: computer-based design with custom integrated circuits. This revolution will be so dramatic that a large percentage of engineers will not survive the transition. Users will experience the revolution as a mixed blessing.

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uum tubes was, by definition, limited in complexity. A vacuum tube was a very large element, required a large socket, a source of power, and it dominated the cost. Equipment could only have a few stages of active elements. The passive elements played a minor role and they were hand soldered. In the next stage, transistors in a printed circuit board allowed for much larger and more complex designs, since the density of stages was much higher. Integrated circuits continued the trend with the passive components becoming a larger issue. It cost more to install six resistors than an amplifier. The above progression can be seen along the dimension of increased packaging density allowing for more complex systems. The most recent generation of equipment has reached a limit in these terms and we must seek a more revolutionary solution if the density is to get still greater.

Welcome to the world of custom integrated circuits. This world is not new in 1986; it has been around some years but it has been the exclusive domain of very large companies making products in very large volumes. The year 1986 is, however, the year that the cost dropped to the point where audio engineering companies will be finding it an attractive methodology. Hence, the revolution is starting. You may ask, "Why does a custom integrated circuit produce a revolution?"

CUSTOM IC

Many manufacturers are offering the facilities to design one's own integrated circuits using one of several technologies. These fall into the new buzz word catagory, "Application Specific Integrated Circuits" (or ASIC for short). The simplest and most accessible is gate-arrays. In this technology, the manufacturer makes an integrated circuit with a very large number of CMOS transistors in a rectangular array. None of the transistors have wires connected; it is simply a grid of active devices. The smallest is on the order of 3000 transistors and the largest is on the order of 100,000 transistors. Such circuits do nothing because there is no wiring to connect the transistors. The chip is a standard part.

To customize the chip, the user supplies his own wiring. The process of wiring the transistors is dirctly analogous to the wiring of a printed circuit board with hoizontal and vertical wires. If the process of adding wires can be made cheap enough, then everybody can make their own integrated circuit. Unlike a printed circuit board, the wiring of an integrated circuit gate array requires very sophisticated technology. Many of the semiconductor companies have built the facilities to allow their customers to supply the wiring diagram. In simple terms, the user supplies the schematic diagram of the circuit and the semiconductor company does the wiring for you.

The result is that you can then buy your own integrated circuit. The cost of this is extremely low. Let us explore some of the implications. A gate array with 100,000 transistors is equivalent to about 20,000 NAND gates or about 4,000 flip-flops. Such a device has the ability to do a full digital audio electronic reverberation system on one IC! The cost of such a part may only be on the order of $50. In 1975, such an equivalent performance would have cost on the order of $5,000. That is a 100 fold price reduction in just one decade. Systems made from this kind of technology will become very inexpensive. We have already seen this in the consumer area with the CD disk players. Consider that you can buy such a player for on the order of $180 and the retailer probably pays only $120. That means that the manufacturer was able to make the product for about $60. The ASIC technology first appeared in this area because of the large volumes of production. This is also why the consumer technology had been leading the professional technology. The reversal of order will become more typical as time goes on.

In the next few years, we will be seeing more of the professional equipment using this kind of technology. Unfortunately, the sophistication, both technically and economically, will separate the men from the boys (or the women from the girls).

DESIGN EXPERIENCE

Anybody who has ever designed a piece of equipment or a piece of software is familiar with the process of debugging. Anybody who has purchased equipment from the beginning of a production run also understands the idea of design "bugs." These are generally fixed on the fly with "patches." Hardware equipment typically has a few extra wires hung into the system to correct such bugs. Every engineer understands the use of the laboratory to find and fix such bugs. The ASIC techology, however, is different. You cannot fix the bugs inside the integrated circuit. It must be perfect or you start again. There is no fixing phase of the design. Similarly, you cannot repair such a circuit. Think about what that means. Think about a system with the equivalent complexity of 4,000 flip-flops; think about a system with the equivalent of 20,000 wires; now think about getting every one right! That frizzles the mind.

Nobody could ever expect such perfection from humans. To solve that problem, the semiconductor manufacturers have spent a very large amount of effort to build special computer design tools which allow the designer to fully simulate in software the functions of the ASIC circuit. This means that the ASIC device is first constructed in a simulation format where a computer duplicates the entire IC. It duplicates all of the characteristics of each transistor including delay, voltage sensitivity, power-supply voltage, capacitance loading, etc. The simulation is so good that the semiconductor manufacturer will guarantee that the simulator results will be identical to the actual IC. IF the simulator runs, so will the IC. Such simulators mean that the entire IC can be tested without being built.

One enters the schematic diagram and then creates sample input signals which are observed on output pins. This is the same kind of activity which a logic designer had done in the laboratory with a logic analyzer and special hardware test equipment. The simulator is one of a new class of CAD tools (computer-aided design). This is a software world in which the actual hardware is the end result. CAD tools are not just a question of a convenient way of design; there is no other way to deal with such a complex process. Having had the experience, I can tell you that there is a large amount of stress in attempting to test something and to prove to oneself that it is bug free. An error may cost $20,000 just for a simple defect in the design.

The organizational skills are very different from that of discrete design where the proto-type is tested in the laboratory. ASIC design is for men and women, not boys and girls.

One of the direct implications of these facts is that the initial cost for the proto-type is very high. This means that one cannot play with the design in the laboratory. Only big companies have the economic basis to create products in this technology. Many of the smaller companies will not be able to compete in this area.

We are recently seeing this process in the market area of electronic reverberations. Companies making products out of standard logic families generally sell products for $2,000 to $10,000 whereas a new entry into this market using the ASIC technology is able to sell at $800. This is a non-trivial difference and the market is likely to buy a lot of the less expensive units. It is necessary for that to happen because the developmental cost of the ASIC system may be as much as $1,000,000. Because the relationship between volume and cost is so non-linear, the lower priced units open up new markets. At $800 every studio, no matter how small, can afford several such reverberation systems. One can even begin to think in terms of hi fi system including their own professional-quality reverberation. This is kind of a democratization of the profession.

The user club will get larger as the designer club gets smaller. We have seen this pattern in other industries. I am not sure that I will like the brave new world but then again, nobody asked me if I liked it. Perhaps one reason that it makes me uncomforatable is that most American manufacturers will not play the game and that moves the technology to foreign countries. We are left to consumers. Also, our dependency becomes greater since we cannot repair the equipment nor modify it to our taste.

THE FUTURE

What we have just described will continue but at a faster rate. The number of transistors on an ASIC device will get larger as the size of each transistor gets smaller. The real cost is based on the area of the chip not the number of devices. the measure of size is given in microns (one millionth of a meter). The old CD4000 family of CMOS had been six micron technology. Several years ago, three micron was introduced but soon replaced with two micron. Several companies have recently introduced 1.5 micron and they are working at still smaller geometries. Crudely speaking, the density goes as the square of the lateral dimensions. From 6 to 1.5 represents grossly a factor of sixteen density change.

The reason that the costs are related to area is that the probability of a crystal defect is proportional to area, e.g., one defect per mil squared etc. When the chip get smaller, there is a corresponding reduction in the probability of a defect. The pattern is thus clear, the cost of an active device will get lower and lower.

The main cost will be controlled only by the area and the number of pins on the package. Internal connections cost nothing. Another positive gain from higher densities is that the capacitance becomes very small and the speed become correspondingly higher. The 2 micron technology has speeds corresponding to the ECL family, 1 nanosecond delay in a NAND gate. This means that the signal processing power increases rapidly as the density gets larger. With enough processsing power in one IC, we can think in terms of the digital audio mixing console becoming "cheaper" than the analog counterpart. The idea for such a console had been considered, but the cost of signal processing on each channel made the result very expensive. With an ASIC technology, it would be cheaper to use the ASIC digital signal processing than to have an 8-position switch for an analog change in filtering! It is hard to belive that this will be the world of the future.

While the initial motivation for digital audio had been the increase in quality; the final result will be based on the cost advantage. The tail will wag the dog. ■

Oliver Masciarotte

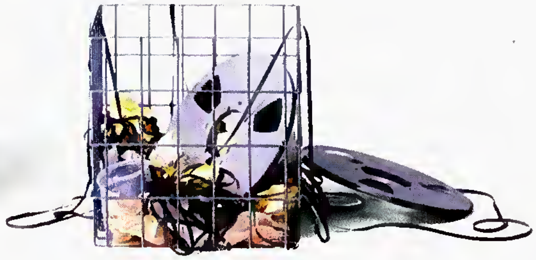
Delayed Reaction

• I was sitting with a friend the other day listening to music from his mini ghetto blaster when I noticed a normal/wide switch on the box. Placing my head between the speakers with the box in "wide" mode, I received the impression that the reproduce system somehow was much bigger than it really was. By jove, it works! Well, the "somehow" is psychoacoustics and this installment continues with the discussion of brain/ear processing and perception of auditory stimuli and how this stuff can be used purposefully.

Before I launch into this tirade I would like to review briefly. I have mentioned four phenomena contributing to perception of a sound source's direction. They are: 1) changing pitch with changing intensity, 2) the Equal Loudness curves, 3) changing timbre with different lateral angles, and 4) different perceived elevation at different frequencies. Item numbers three and four provide strong directional cues while item numbers one and two provide less directional information, yet serve to color our overall perception of sound.

You might be wondering how the aforementioned information fits in the scheme of things. Well, assuming that intensity varies as the inverse square of the distance from a source, then according to numbers one and two above, the pitch and timbre of a source will vary subtly as you move nearer or farther. Let's also assume that you are not situated in an anechoic environment. This means that the ratio of direct sound to reflected sound from any surface or object nearby will vary as you or the source move about (see Chowning). Don't forget that high frequencies (hf) are attenuated in their passage through air which causes distant sources to sound "wooly" or lacking in hf content. Taken together these factors provide a good indication of distance from a sound source. The qualifier about not listening in a reflection-free environment is very important as the time and amplitude distribution of reflections is a strong cue to listener-to-source placement in any environment. Also, the degree of "definition" (how focused or how closely does the source appear to be a point source) and the perceived spaciousness (how live or dead the environment is and what are its apparent physical dimensions) are determined to a large extent by the intensity and direction of reflections.

As for item numbers three and four above, there are folks currently working on digital signal processing that utilizes these psychoacoustic cues as well as other garden variety stuff men-



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TRUTH: A lot of monitors "color" their sound. They don't deliver truly flat response. Their technology is full of compromises. Their components are from a variety of sources, and not designed to precisely integrate with each other.
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TRUTH: JBL eliminates these consequences by achieving a new "truth" in sound: JBL's remarkable new 4400 Series. The design, size, and materials have been specifically tailored to each monitor's function. For example, the 2-way 4406 6″ Monitor is ideally designed for console or close-in listening. While the 2-way 8″ 4408 is ideal for broadcast applications. The 3-way 10″ 4410 Monitor captures maximum spatial detail at greater listening distances. And the 3-way 12″ 4412 Monitor is mounted with a tight-cluster arrangement for close-in monitoring.
CONSEQUENCES: "Universal" monitors, those not specifically designed for a precise application or environment, invariably compromise technology, with inferior sound the result.

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

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

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

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

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

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tioned in the last paragraph to synthesize complex reverberation (Kendall & Martens). Hopefully, engineers will soon have access to powerful signal processors that will provide simultaneous control of many individual parameters for the creation of unambiguous placement of a phantom image in an artificial acoustic space.

Back in the real world, we all know that if a sound source is located to someone's left or right, one ear will receive higher intensity information than the other. This is the basis for a panoramic potentiometer, the most basic tool for control of apparent direction. Any stereo mixing console has pan pots in the monitor section to proportionally route a monaural signal to the stereo mix buss. There is something analogous to a pan pot as part of your stereo. While playing back a mono source (those old records you haven't played in years), try twisting the "balance" control on your hi-fi to see what I mean.

Another way to fool the brain into perceiving direction is to simulate the differing transit times to each ear for a source located to one side of the head. There is a small but significant difference in the distance a sound must travel to reach each ear because our eardrums are physically separated by several inches. For a source arriving at our ears at different times, our brain is capable of perceiving an interaural delay of less than a tenth of a millisecond! Also, I have read a figure of .007 milliseconds which is equivalent to a source moving one degree to either side. I can hear a displacement of about two degrees so this makes sense to me. Still, we are talking about very small time delays affecting our perception of direction.

It is easy to "place" a monaural source in a stereo sound field with this approach by utilizing an audio delay line, preferably with a fine gradation of delay time adjustment. Simply pan a direct (unprocessed) signal hard to one side and route that same signal through a delay device set for around three to six milliseconds. Then take the delayed output and hard pan it to the opposite extreme from the direct signal. You can set the level of the direct and delayed signals to be equal and the listener will swear that the sound comes from the speaker that's playing back the direct signal. Researchers have given the name "precedence effect" to this phenomenon. An interesting thing about this is that it allows you to set equal amplitudes in the stereo mix buss so that the left/right levels are not unequal as *would* be the case with simply panning, yet the phantom image would be "panned" to one side. A word of caution in that this technique is not ideal for monaural reproduction. When you sum the stereo pair into mono, the delayed signal will cause phase cancellation and alter the timbre of the dry sound. While this method works with delay times of a fraction of a second on up, very short delay times will cause cancellation in the critical mid-frequency region where our amplitude perception is most acute. So, set your delay time to create the degree of "panning" you want and then check it in mono, altering the delay time a wee bit if necessary. Also, don't be afraid to go with a delay time of up to thirty-five milliseconds for some interesting variation, though you should reduce the amplitude of the delayed signal (8 to 10 dB) to compensate for the longer delay time.

The warning just mentioned brings me to some final words about psychoacoustics. The brain treats delayed versions of a sound as part of the sound if they are heard less than about thirty-five milliseconds after the initial direct sound. This is true only if the delayed versions are of lesser amplitude than the direct sound, the amplitude lessening logarithmically to about 12 dB below the direct signal at thirty-five milliseconds. If a sound lies outside of this log curve of decaying amplitude, the brain will not "fuse" it with the direct sound and perceive it as a discrete echo. This integration of direct and delayed information by the brain is called the "fusion" effect. Fusion and precedence are usually lumped into one entity often bandied about in audio circles, the Haas effect (Haas).

Psychoacoustics is a complex and slightly imprecise science, but a general understanding of the main precepts will go a long way towards improving one's ability to predictably control sound placement and perception.

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MIDI can communicate various types of data between devices properly interfaced that include commands for note-on/note-off, select a new voice, pitch change, velocity, pitch bend, modulation change, sustain control, after-touch, clocking information, various sequencer commands, plus a variety of system-exclusive commands that are up to the individual manufacturer to define. The system-exclusive commands allow manufacturers of devices that have special features to be cotrolled via MIDI, i.e., delay line settings. There are sixteen different channels in the MIDI standard in which each can transmit up to sixteen notes simultaneously or a total of up to 256 notes. In practice, several of the sixteen channels are used for sequencers, drum machines, and other automation tasks.

Devices that adhere to the MIDI standard are intended to work together properly in their least common level. MIDI can't add features to a device that does not have them. For example, if a synthesizer is not velocity sensitive then it will ignore velocity information from a velocity sensitive master keyboard.

How does MIDI work? Normally, MIDI devices have three different MIDI jacks (5-pole DIN): MIDI OUT: here the MIDI data are sent. MIDI IN: here the MIDI data are received. MIDI THRU: here the received MIDI data are looped through.

Devices that are designed for passive (Receive) mode of operation have only the IN and THRU jacks and thus, cannot send MIDI signals themselves. Hardware-wise, the MIDI circuitry is an opto-isolated TTL-compatible current loop configured for the 31.25 kilobaud asyncrhronous serial data transmission.

The opto-isolation may seem like a panacea at first since there is complete isolation from the audio ground and one can almost completely forget about ground loops. However, each opto-isolator requires a finite amount of time in which to respond to data on its input. Since the serial data pulses transmitted by MIDI have rise times that are faster than the response times of the opto-isolators themselves, each opto-isolator introduces slew-rate-limiting distortion to the MIDI data. Thus, if for example, three devices are communicating via MIDI daisy-chaining, then the MIDI data has passed through four opto-isolator circuits and the data distortion has been compounded three times. After only three or four times, the pulse symmetry becomes so degraded that the MIDI circuitry downline can no longer read the MIDI data reliably. In practice, three receiving devices is the limit in daisy-chaining.

The solution to this most common problem in MIDI applications is to use a MIDI THRU "box" as the heart of a star network (see *Figure 1*). An ideal MIDI THRU box would incorporate a single MIDI IN connected to a fast opto-isolator to drive up to twenty MIDI THRUs via digital logic. Thus, a single MIDI source could drive each receiving device with a virtually distortionless replica of the transmitted data. Currently, both Yamaha and Roland manufacture MIDI THRU boxes, but these devices utilize standard (slower) opto-isolators, and have only four MIDI THRUs each. In context, they represent a small-scale solution to the problem.

MIDI cables transfer high-speed digital data and as such must be constructed accordingly. The cables should be properly shielded to prevent MIDI data errors caused by electromagnetic and electrostatic interference. They must be low-capacitance cables to minimize data distortion due to high-frequency attenuation. MIDI cables should be as short as possible. At no time should a MIDI cable exceed fifty feet in length.

Getting our MIDI signals in and out of computers is another matter that requires a MIDI port. Yes, that's right—even if you have a serial, parallel, Cenntronics printer, joystick, mouse, RS-232, and IEEE ports—that's not enough. With the exception of the Atari 520ST, the Commodore Amiga, and the Yamaha CX5M computers (MIDI port is built-in), a peripheral-card that has the appropriate opto-isolators, digital-data buss interface, and DIN plugs is what it will take. These cards range from about $100 for a dumb card to about $300 for a smart card as exemplified by the Roland MPU-401. A dumb interface does just the electrical conversion to and from MIDI, while a smart interface takes some of the processing burden off of the computer.

Computers that can take advantage of MIDI include the Commodore 64/128, the Apple II series, the IBM PC series, and most recently the Commodore Amiga. The computer becomes the digital heart of the MIDI system and techniques are under constant development that are expanding its horizons. Sync codes that include pulse-per-quarter-note and MIDI clocking, FSK (Frequency Shift Key), and SMPTE allow just about anything electronic to be operated synchronously.

Technology from the computer and communications sector is providing us as musicians and engineers with new tools at an unprecidented rate. High-speed digital data transmission technology, for example, will allow us to transmit MIDI information across town or across the world. The need and resources of the information technology sectors greatly exceeds thsoe of the music industry. Fortunately, we are benefiting from the technology that trickles down.

MIDI has the potential of linking music, physics, and engineering if we all open our minds. MIDI has been around for just over two years. In this short time it has demonstrated the direction of our industry—even given us a preview of the future. When companies like Boesendorfer demonstrate digital/mechanical record/playback on a nine foot grand piano, it clearly shows the narrowing of the gap between the acoustical world and the electronic world.

There was never a greater contemporary visionary in music than Leopold Stokowski—the conductor of the Philadelphia Orchestra from 1912-1938. Under the direction of Stokowski the Philadelphia Orchestra achieved a reputation of being the first with historical events that included first to make orthophonic recordings (electrical as opposed to the earlier acoustical recordings), first to broadcast on radio, and first to participate in Hollywood movies—notably *Fantasia.* His interest in electronic music which started with the Teleharmonium in 1906, led him to precede a 1931 concert with a demonstration of the noises that could be produced by several oscillators. "Watch the birdie!" was his announcement.

During the period of the classics the acoustic state-of-the-art instruments were included in new compositions and performances. Today synthesizers and MIDI are providing us with a new generation of instruments creating a new period in music and communications. With Stokowski's fascination for the technology of music, were he to be alive today, there would be synthesizers, electronic architecture, and MIDI, in the Academy of Music.

For more on MIDI, see Larry Openheimer's in-depth article, "Making Sense Out Of MIDI." ■


Create a room
YAMAHA REMOTE CONTROL UNIT RCR-1
FOR DIGITAL REVERBERATOR REV-1
REVERB WAVEFORM
dB 0 -20 -40 -60
D2
DISPLAY
W/F F/C R/T RATIO
E/R REV P/S M
E/R MODE
REV. MODE
PRESET
PANEL
P EDIT AUTO
MEMORY
67
M STR RCL
FUNCTION
R/T D1 D2 M
LEVEL METER
IN
OUT
L R
FULL
ROOM SIZE
HIGH
E/R NUMBER
MID-HI
LIVENESS
LOW
HPF
LPF
400 10K
200 8K
100 6K
50 4K
4K 2K 1K
500 250 125
REV. TIME (R/T)
2.6 sec
MID-LOW
AUTO
E/R DELAY 1 (D1)
40 ms
REV DELAY 2 (D2)
58 ms
7 8 9
4 5 6
1 2 3
0 . CLR
UP DWN ENT
ON
DIRECT
EARLY REFLECTION
REVERBERATION
MASTER

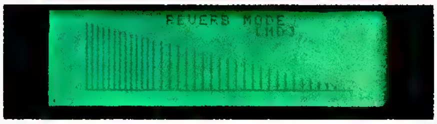
with a view.

We'd like to open your eyes to the incredible REV-1 digital reverb. Because it gives you unheard-of control over virtually all reverb parameters. And something that has never been seen in any type of reverb: the capability to "look" at the sound as well as hear it.

The remote unit that controls the nineteen-inch rack-mountable unit has a lighted high-resolution LCD display that graphically depicts the results of the adjustments you make.

So getting just the right reverb sound is no longer a question of trial and error.

"EARLY REFLECTION" display mode showing room size and relative level and time of discrete reflections.



"REVERB DENSITY" display mode showing level and relative time of subsequent reverberation.


"REVERB TIME" display mode showing difference in reverb time in each of four frequency bands.

"MEMORY TITLE" display showing the titles of internal ROM memories.

The logical grouping of the parameter controls on the remote also makes it easy to create any effect you like. Then store it in any of 60 memories for instant recall.

The remote also contains 9 additional RAMs so you can store programs and carry them with you to use anywhere there's an REV-1.

And there are 30 additional ROMs with factory preset sounds. Many of which can be completely edited (as can the user-programmed sounds) by using the LEDs to tell you the set value or indicate in which direction to move the control so you can easily and precisely match the value of the originally programmed sound.

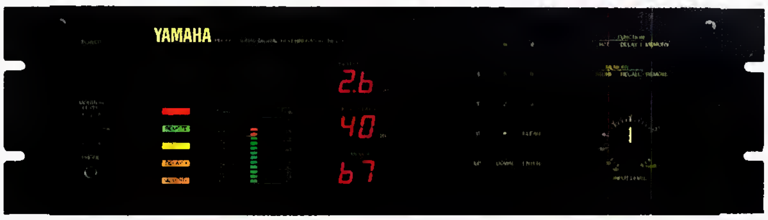
And the sound itself is far superior to any other digital reverb. The REV-1 uses specially developed Yamaha LSIs to create up to 40 early reflections and up to 99.9 seconds of subsequent reverberation. So the effect can be as natural (or unnatural) as you want it to be.

We could go on about the REV-1. Tell you about its 44.1 kHz sampling rate that provides a full 18 kHz bandwidth to prevent the natural frequency content of the input signal from being degraded.

How it has a dynamic range of more than 90 dB for the delay circuitry and more than 85 dB for the reverb circuitry.

But why not take a closer look at the REV-1 at your authorized Yamaha Professional Audio Products dealer. Or for a complete brochure, write: Yamaha International Corporation, Professional Products Division, P.O. Box 6600, Buena Park, CA 90622. In Canada, Yamaha Canada Music Ltd., 135 Milner Ave., Scarborough, Ont. M1S 3R1.

YAMAHA



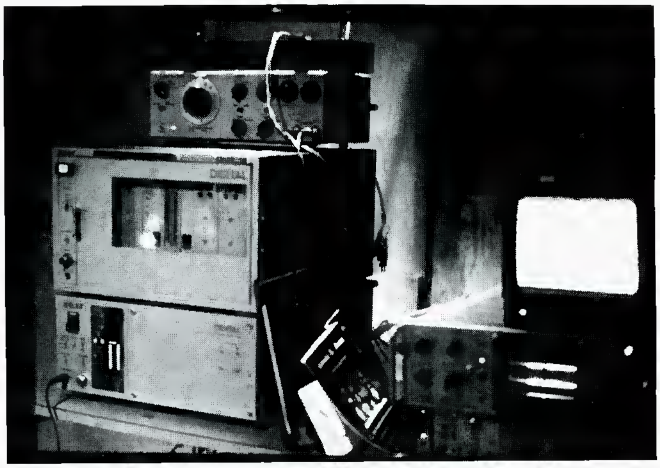
Circle 23 on Reader Service Card

Co-author William Spurlin is seen with the WGBH transmission equipment that was used.

Initially scheduled for November, the Boston to Tokyo transmission was postponed until January when an illness sidelined BSO artistic director Seiji Ozawa. The delay proved fortuitous in that it enabled WGBH and FM Tokyo engineers to utilize the circuits that had been reserved for the November broadcast to experiment with the international link. Originating from Symphony Hall, the program was slated to be fed on a 2 gHz ENG microwave to a relay station atop Boston's Prudential Tower, upconverted to 13 gHz and retransmitted to WGBH's studios. At WGBH, the signal was fed into another Sony 1630 for error correction and retransmitted ten miles via 13 gHz microwave to WGBH-TV's uplink facilities in Needham, Massachusetts. From Needham, the video format digital audio signal was uplinked to Westar IV, transponder 10D. KQED-TV, San Francisco was the domestic downlink that relayed the signal on AT&T circuits to the COMSAT uplink in Jamesburg, California. The program was beamed to an INTELSAT satellite, downlinked in Ibaraki, Japan and relayed to FM Tokyo for decoding and rebroadcast.

The three video-format digital audio (VFDA) systems used by WGBH for intercontinental transmission, Sony 1600, Sony PCM F-1 and dbx 700, employ different modifications to the standard NTSC television signal in order to encode digital audio. Both the dbx 700 and the Sony F-1 are closer to the NTSC or RS170 standard in that neither format begins encoding audio information until horizontal line twenty-one of fields one and two of the vertical interval. In contrast, the Sony-1600-format processors (i.e., the 1610 and 1630) begin encoding at line eighteen of field one and line seventeen of field two of the vertical blanking interval.

Although the post-vertical sync lines of the vertical interval are not used to convey picture information, they are used to transmit other kinds of information. Television test signals (Vertical Interval Reference Signals and Vertical Interval Test Signals) are routinely inserted on lines seventeen, eighteen, and nineteen for both transmission test purposes (VITS) and the automatic alignment of receivers (VIRS). Teletext and the CBS Labs-Thompson CSF Videotext system, used to convey large amounts of screen-oriented textual information, also use the vertical interval. in addition, DACS systems, employed by radio and TV networks to convey programming information to tele-



In this view of other transmission equipment we can make out the Sony 160 and 1630 units. The picture monitor and wave-form monitor is also seen.

type printers, also encode information in the vertical interval of satellite-TV network signals. In some cases, the vertical interval is wholly or partially reconstituted or "stripped" by various pieces of television equipment, notably frame synchronizers and certain processing amplifiers. Because of its format, the Sony 1600 is vulnerable to any equipment that removes or alters information in the vertical interval while both the dbx and the Sony F-1 signals are immune to "stripping."

Since WGBH engineers were aware of the peculiarities of the Sony 1600-format, it was not surprising that initial reports from Tokyo indicated that the test transmission was indecipherable because lines seventeen and eighteen were missing. Once a DACS had been removed from WGBH's satellite receiver, the transmitted signal was viewed from the WGBH downlink as intact, yet, Tokyo continued to be missing lines. WGBH engineers traced the problem to the KQED downlink. Equipment in KQED's signal path was stripping lines seventeen and eighteen. Quickly corrected by KQED's removal of the offending equipment, the transmission worked successfully for the balance of the test period.

Unfortunately, for the January broadcast, the original transmission path was unavailable. All of the PBS transponders on Westar IV, the satellite accessible by the WGBH uplink, were booked, thereby forcing an alternate route for the domestic leg of the international link. Western Union, the broker for the international relay, routed the signal from the New England Telephone's Central termination office in Boston to the RCA uplink in New Jersey via AT&T circuits. RCA uplinked the program on SATCOM 1R, downlinked it at their Point Reyes, CA facility and routed it via AT&T lines to the COMSAT facility in Jamesburg.

Assured that the common carriers in the transmission path would not process the vertical interval, WGBH engineers felt confident that problems experienced during the November test period would not reoccur. However, because of AT&T's extensive involvement in the new route, another potential source of difficulty presented itself—noise. The Sony 1600-system specifies a weighted (i.e., peak-to-peak signal divided by rms noise) signal-to-noise ratio of 49 dB. Such a s/n figure is standard for satellite transmission practice. However, AT&T specifies that the weighted s/n ratio of its video carriers shall not be less than 43 dB, a level of noise which can introduce an unacceptably high rate of errors in the 1600-system decoding process.

The FM Tokyo production team. Right, Yoshie Sakamoto, producer, and left, Huruhiko Hagimoto, announcer.

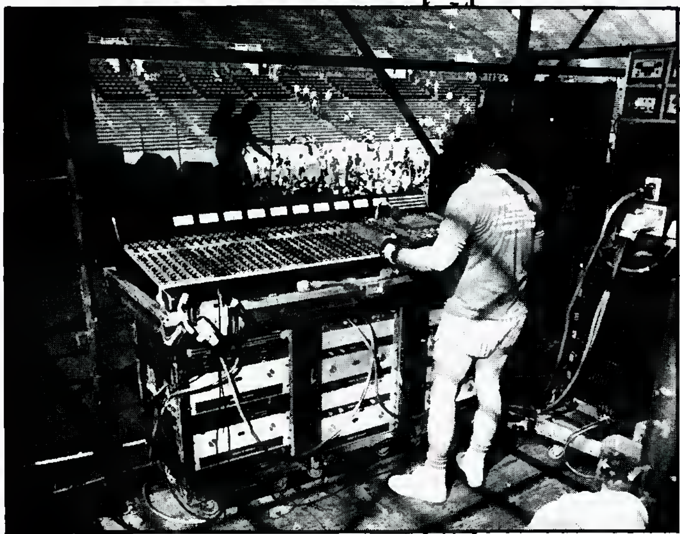
At left, Takeshi Yazawa of Sony. At right, Kenichi Matsuda of FM Tokyo, and in the rear, William Spurlin, co-author.

As the test period before the January broadcast commenced, complaints of decoding errors related to a s/n ratio of 43 dB were received by William Spurlin, WGBH Engineer for Maintenance and Transmission, and Takeshi Yazawa, Assistant GM of Sony's Professional Audio Division, the transmission engineers for the BSO broadcast. At this juncture a number of possibilities presented themselves for dealing with the noise. During the November test a scheme, (relying on the fact that the luminance level in the 1600 system is only sixty IRE units while standard television practice allows the transmission of one hundred units of luminance or picture-time information), had been devised and partially implemented. A processing amplifier (one that did not reconstitute the vertical interval) was placed in the transmission path at WGBH, allowing the separate level control of both the picture-time and the sync components of the video signal. A non-standard 1600-system signal was thus generated and transmitted, one that had one hundred units of luminance and forty units of sync. In Tokyo, an inverse procedure could have been used to attenuate the luminance portion to the 1600-system's standard of sixty units, at the same time attenuating any noise components by a factor of 100/60 or 4.4 dB.

Yazawa and Spurlin began to boost the level of the transmitted signal while advising Tokyo that filtering, another noise-reduction process which also relies on the special characteristics of the VFDA signal, might be tried. Since the 1600-system requires a bandwidth of 3.5 MHz and the standard television path has a bandwidth of 4.5 MHz, a s/n improvement of the square root of 4.5/3.5 or 1.1 dB can be expected by inserting a 3.5 MHz low-pass filter at the receiving end. Since part of the noise can be chroma leakage from adjacent-channel television satellite interference, a low-pass filter with a good 3.57 MHz rejection characteristic can improve the s/n further. A filter was available in Tokyo and when inserted into the line errors dropped to an acceptable level, obviating the necessity for any further noise-reduction treatment. The broadcast proceeded without further technical difficulty and was a success for all parties.

For WGBH and FM Tokyo, the digital transmission of live music enables the broadcaster a true high fidelity medium which can transmit the dynamics of live music and deliver a signal which retains true concert hall ambience. WGBH now hopes that funding can be secured to make 1986 also the year of the first live digital relay from Tokyo to Boston. ■


WABC77



Len Feldman

BGW Systems 8000 Proline II Power Amplifier

GENERAL INFORMATION

● The BGW Model 8000 is, first and foremost, one of the most beautifully built and designed amplifiers that it has been my pleasure to test and audition. Brian Gary Wachner (whose initials form the corporate name) was so proud of this design that he took the trouble to personally bring it to my lab in Long Island, all the way from California.

He spent a good deal of time at my lab telling me how hethought an amplifier should be designed and built, and on what basis it should be judged. As he correctly pointed out, there would be time enough after he left for me to measure the amp and to listen to it. Equally important, he felt (and I have to agree) are the designs criteria which the end user will appreciate, such as mechanical design integrity, thermal engineering integrity, serviceability, resale value, options, features and, of course, initial purchase cost. So strongly does Wachner feel about some of these points that he actually presented a paper at the 1985 AES Convention held in Anaheim last spring, in which he discussed the guidelines for evaluating an amplifier.

It was clear from the outset of my experiences with the Model 8000 that Wachner practiced what he preached. This is one of the most advanced solid-state, quasi-complementary, high-power bridgeable stereo power amps currently available for professional use. Among its many outstanding features are a huge toroidal power transformer, forced air cooling, a rugged power switch and circuit breaker, smooth-acting input level controls, totally modular construction, display meter assembly, dual looping 1/4-inch phone jacks, and electronic DC speaker protection.

A few of the options available for the Model 8000 include active electronic balanced line inputs, with or without a subsonic filter, (normally, the basic amplifier is supplied with unbalanced, 1/4-inch phone jack inputs), single or dual channel electronic crossovers, transformer balanced line inputs with looping XLR connectors and unbalanced line inputs with looping XLRs.

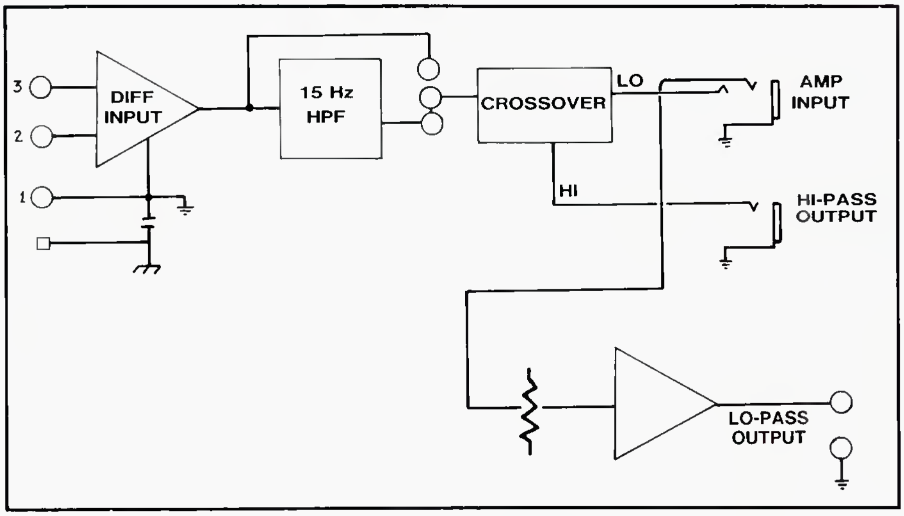
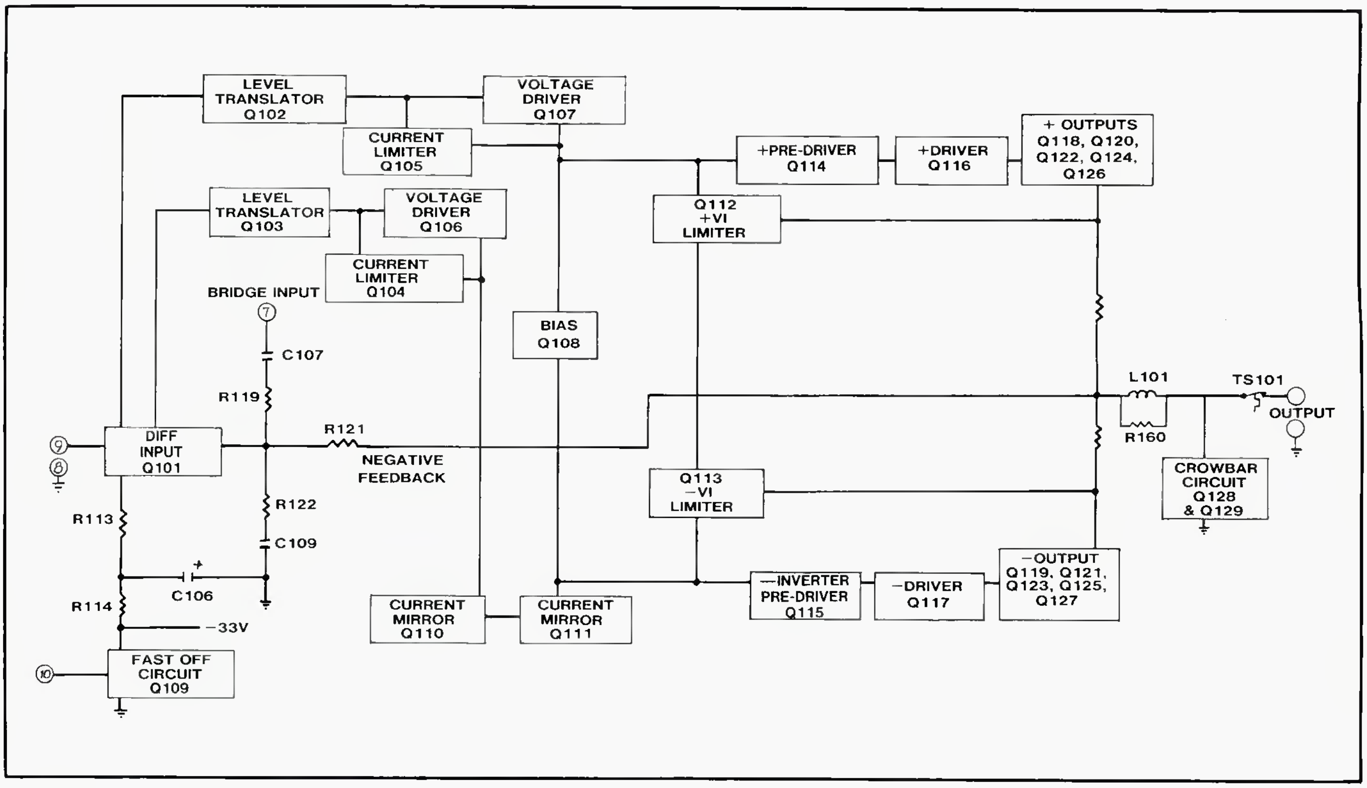


Figure 1. Block diagram Model 8000 with Differential Input and Crossover.



Block diagram for amplifier Model 8000.

Figure 1 illustrates how the differential input and electronic crossover option is integrated into the overall amplifier.

Physically, the amplifier features an all steel welded chassis and covers, large geometry output devices (a total of twenty-four of them, each with a 200 watt power dissipation capability for a total of 4800 watts), massive aluminum heat sinks and a huge toroidally wound power transformer that can be wired for any one of five domestic or international voltages (including 100 volts should you find yourself in Japan with a Model 8000).

CIRCUIT HIGHLIGHTS

A block diagram of the Model 8000 as shown in *Figure 2* is a low-noise matched dual transistor; Q101 is connected as a differential input stage. The output of Q101 is push-pull and the signal drives the emitters of Q102 and Q103 which are connected as common base amplifiers. Their collectors drive Q106 and Q107 which are connected as common emitter voltage amplifier stages. The output of Q107 drives the two driver stages Q114 and Q115, multiplier Q108 and current mirror stage Q110/Q111. Q104 and Q105 are current limiters for Q106 and Q107. Q108 provides bias voltage for the output stages and keeps idling current at a constant level with changes of operating temperatures. The output circuit consists of transistors Q114 through Q127 in a quasi-complementary output circuit. Q114 and Q116 are configured as a Darlington circuit to provide the needed current gain to drive output transistors Q118, Q120, Q122, Q124 and Q126 (the positive half of the output circuit). Q115 is connected as a common emitter and drives Q117 which is in turn connected as a common collector to provide the current needed to drive output transistors Q119, Q121, Q123, Q125 and Q127.

To maintain overall amplifier stability, linearity and low distortion, degenerative feedback is used throughout the amplifier. Except for the input and feedback loop the amplifier uses direct coupling throughout. A "Fast Off" circuit mutes the amplifier channel when a.c. power is turned off. A speaker protection d.c. circuit shunts the loudspeaker load at frequencies of 10 Hz or lower.

When the mono/stereo switch of the amplifier is set to mono, the second channel or "B" channel is converted to a unity gain inverting power amplifier.

CONTROL LAYOUT

The front panel includes two input level controls, a rocker-type a.c. power on/off switch, and dual three-color LED displays. Green LEDs indicate that power is on. Yellow LEDs denote the presence of a signal, and red LEDs serve as overload indicators.

The rear panel contains a circuit breaker, two sets of looping 1/4-inch phone jack input connectors, red and black 5-way binding posts for connecting speaker cables, and a mono/stereo switch to convert the amplifier to a fully bridged mono amplifier. Provisions are also available on the rear panel for an accessory input module. This circuit board, when installed, provides balanced electronic inputs, subsonic filter, and electronic crossover. The input module can be added at any time after purchase of the amplifier.

LAB MEASUREMENTS

Rated at 225 watts minimum continuous average power per channel with both channels driving 8 ohm loads, at any frequency from 20 Hz to 20 kHz, the Model 8000 did much better than that at mid frequencies. Using a 1 kHz test signal, we were able to drive the amplifier to an output of 250 watts per channel for its rated THD of 0.05%. BGW quotes rated distortion at 0.05% from 20 Hz to 10 kHz, and 0.1% at 20 kHz. If we backed off power output to the rated level of 225 watts per channel, THD decreased to 0.03%.

Switching to 4 ohm loads, the amplifier delivered 362 watts per channel for its rated distortion of 0.15%. At the manufacturer's power output rating (for 4 ohm loads) of 350 watts per channel, THD decreased to 0.09%. It's interesting to note that BGW's published specifications are very carefully and honestly worded. For 4 ohm loads they specify that the power band extends only from 40 Hz to 20 kHz atthe rated THD, and that's exactly what we found to be true for our sample. Although we don't show it in the VITAL STATISTICS chart which appears at the end of this report, we operated the amplifier in the mono (bridged) mode as well. As a mono amplifier the Model 8000 delivered 710 watts of power into 8 ohm loads for the rated THD level of 0.15%. That was just a bit higher than the 700 watts total claimed by BGW.

Damping factor, at 50 Hz, referred to 8 ohm loads measured 250 watts as against 200 watts claimed by BGW. All of the remaining measurements were equal to or better than BGW's published specifications. Input sensitivity measured 1.2 V. Signal-to-noise ratio, measured in accordance with EIA standards (referred to 1 watt output, with an input of 0.5 V) was 82 dB, while if measured with respect to rated output (with input level controls wide open) it measured 114 dB, or 4 dB better than claimed by the manufacturer. Dynamic headroom was rather high for an amplifier of this type. It measured 1.9 dB, which means that under conditions of short-term musical peak signals, the amplifier can deliver as much as 348 watts per channel while driving 8 ohm loads.

COMMENTS

Almost from the moment we removed the Model 8000 amplifier from its shipping carton we sensed that here was an amplifier designed to take the kind of punishment that hours and hours of use at or near maximum peak power levels is likely to impart to a professional sound-reinforcement product. Removal of the metal cabinet further confirmed our initial impression of the amplifier. Inside we found a huge, centrally mounted and totally shielded toroidal power transformer, flanked by the two amplifier modules. Eleven-pin connectors attach from the main chassis assembly wiring to each amplifier module so that the latter can be removed easily should servicing ever be required. As we explored the insides of this amplifier we concluded that here was a "textbook case" of good amplifier layout and design.

The excellence of design of the 8000 goes beyond the physical layout. Our own listening tests, conducted over a period of several days, convinced us that just because an amplifier is designed for professional sound reinforcement applications doesn't mean that it can't be a "high fidelity" power amplifier as well. Reproduced sound of the Model 8000, whether at highest power levels or at fractions of 1 watt, was always clean, smooth and as wide-band as the most demanding audiophile might expect. At the same time, continued operation at high power levels did not increase outer case temperatures anywhere near as much as is usually the case. The ventilating fan was one of the quietest we have encountered, even though it does run continuously.

In short, Brian Wachner and his staff practiced what they preached. They have designed and produced a rugged, high powered amplifier that, judged by the criteria enumerated in Wachner's own AES paper, scores as close to "100" as any amplifier I have tested.

VITAL STATISTICS POWER AMPLIFIER

MAKE & MODEL #: BGW 8000 Proline II

SPECIFICATION	MFR'S CLAIM	db MEASURED
Power Output for rated THD (8-ohms, 1 kHz) (Watts)	225	250
Power Output for rated THD (4-ohms, 1 kHz) (Watts)	350	362
THD @ Rated Output, 1 kHz, 8 ohms	0.05%	0.03%
THD@ Rated Output, 1 kHz, 4 ohms	0.15%	0.09%
THD @ Rated Output, 20 Hz, 8 ohms	0.1%	0.1%
THD @ Rated Output, 20kHz, 8 ohms	0.1%	0.1%
SMPTE IM Distortion, Rated Power	0.05%	0.045%
CCIF IM Distortion @ Rated Power	N/A	0.008%
EIA IM Distortion @ Rated Power	N/A	0.1%
Frequency response @ 1 Watt		
Hz to kHz, for −1 dB	20-20(±0.25dB)	7 to 105
Hz to kHz, for −3 dB	1 to 100	1 to 150
S/N Ratio, re: 1 W, A-weighted	N/A	82 dB
S/N Ratio, re: Rated Output	110 dB	114 dB
Dynamic Headroom	N/A	1.9 dB
Damping Factor @ 50 Hz, 8-ohms	200	250
EIA Input Sensitivity	N/A	80 mV
Input Sensitivity re: Rated Output	1.23 V.	1.20 V
Power Consumption		
Idling	N/A	100 Watts
Full Power, 8 ohms	1600 Watts	1400 Watts
Dimensions (W″ × H″ × D″)	19 × 5-1/4 × 13	Confirmed
Net Weight	43-1/2 lbs.	Confirmed
Suggested Retail Price		
Basic Amplifier	$1149.00	
w/Active Balanced Inputs	$1329.00	
w/Dual Input Transformers	$1329.00	

■

Wade Bray

Electronic Architecture In The Broadcast And Recording Studio

The following gives insight into attacking musical acoustics problems with electronic architecture rather than "fixing it in the mix."

TODAY'S BROADCAST and recording studio enjoys a highly developed technology not only for mic'ing, mixing, and recording, but also for adding acoustic ambience to the recorded product.

Signal delay devices, reverberators, and other processors create for the listener the sense of acoustical space and "feeling" which composer, performer, and producer intended. The studio is indeed the ultimate "multi-use hall," as its elaborate post-processing means attest. In contrast to this, the acoustic "deadness" and isolation in the studio itself often removes all room acoustic cues for the performers, who become uncomfortable, "force" their tone, and must concentrate unduly on "making the sound" to the detriment of giving an uninspired performance.

While the listener will experience the intended ambience, the performers making the music experience none of it. They must envision the result and perform accordingly, realizing their success at one remove as they listen to playbacks and work for the final mix. It is odd that the same technological development which has given us such powerful "output" processing tools has not been more widely applied to processing the "source room" itself.

The technology exists, as has been proven in many performance spaces and in very few broadcast studios. Design experience with "electronic architecture" enables today's broadcast or recording studio to make its performance space into any desired virtual space, duplicating the critical characteristics of real rooms of any size and type, and producing physically impossible ones. Parameters may be under independent control, with computer interfacing if desired for "scene presets" and dynamic changes of acoustics. The character of a space may be varied not only for different types of performance and mood, but "on the fly" as an artistic element of composition, expression, or production. Musicians perform "within the virtual space," their sound reflecting, developing, and reverberating from "surfaces" and within "volumes." The separation of artistic concept and result is removed, and the inspiration of creative involvement maximized.

Studios are aware that their business depends not only on the technical quality of products they are able to release, but also on the amenities of environment for clients and performers, to encourage relaxed creativity. In addition to innovative architecture, fine furniture, special instruments, and unusual locales, studios could add an extremely powerful incentive: full control of the "performing environment" as a function of artistic concept.

AN EARLY EXAMPLE—STUDIO 4, BBC TELEVISION CENTER

Television studios are made very absorbent and "dead," in order to supress background noise from equipment and incidental activity, focus the attention of the viewers on individual talkers, and maximize the ratio of direct sound to reflected and diffuse sound for boom mic'ing. As a result, musicians receive no "room cues" and the sense of ensemble is entirely absent.

In the early 1960s, BBC Television realized that the lack of acoustic cues made orchestral performance impossible in its 357,000 cubic foot Studio 4, so it installed a Philips EL6911 "ambiophony" system, consisting of a tape-loop machine, multiple amplifier channels, and sixty-two loudspeakers

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Wade Bray is an acoustical consultant with Jaffe Acoustics, an accomplished organist, and is a recording engineer and producer. He is currently digitally recording concert pipe organ performances.

positioned twenty-five feet off the floor, around the periphery of the studio, and in the ceiling.

One of the microphones, independent of the broadcast pickup, received the signal for the record head. Eight reproduce heads fed speakers in different areas of the studio, so that in general sound was re-emitted at the same time as the arrival of the direct sound at that location. A small increase in reverberation time was noted from regeneration (the pickup microphones receiving sound from the loudspeakers), and more reverberation time could be had by recirculating a portion of the last reproduce head signal back to the input, or by adding a reverberator in the input circuit. An initial delay of thirty milliseconds was maintained before the onset of the ambiophonic enhancement. The system was adjusted to simulate the effect of a concert hall at the position of the orchestra, and it permitted symphonic broadcasts to be aired from this studio.

THE NBC-TV "LIVE FROM STUDIO 8H" PROJECT

From 1929 through 1936 the famous Arturo Toscanini directed the New York Philharmonic Orchestra. During this time the RCA Radio City complex in Rockefeller Center was under construction, including the extravagant Studio 8H, then the world's largest broadcast studio.

Studio 8H was the home of the big variety shows of Ed Wynne, Eddie Cantor, Fred Allen, and Rudy Vallee. In 1937, NBC lured Arturo Toscanini away from the New York Philharmonic, giving him carte blanche to establish the NBC Symphony Orchestra with some of the world's top musicians of his choice. Some feel that the NBC Symphony, which continued at 8H until 1954, was among the finest orchestras ever assembled.

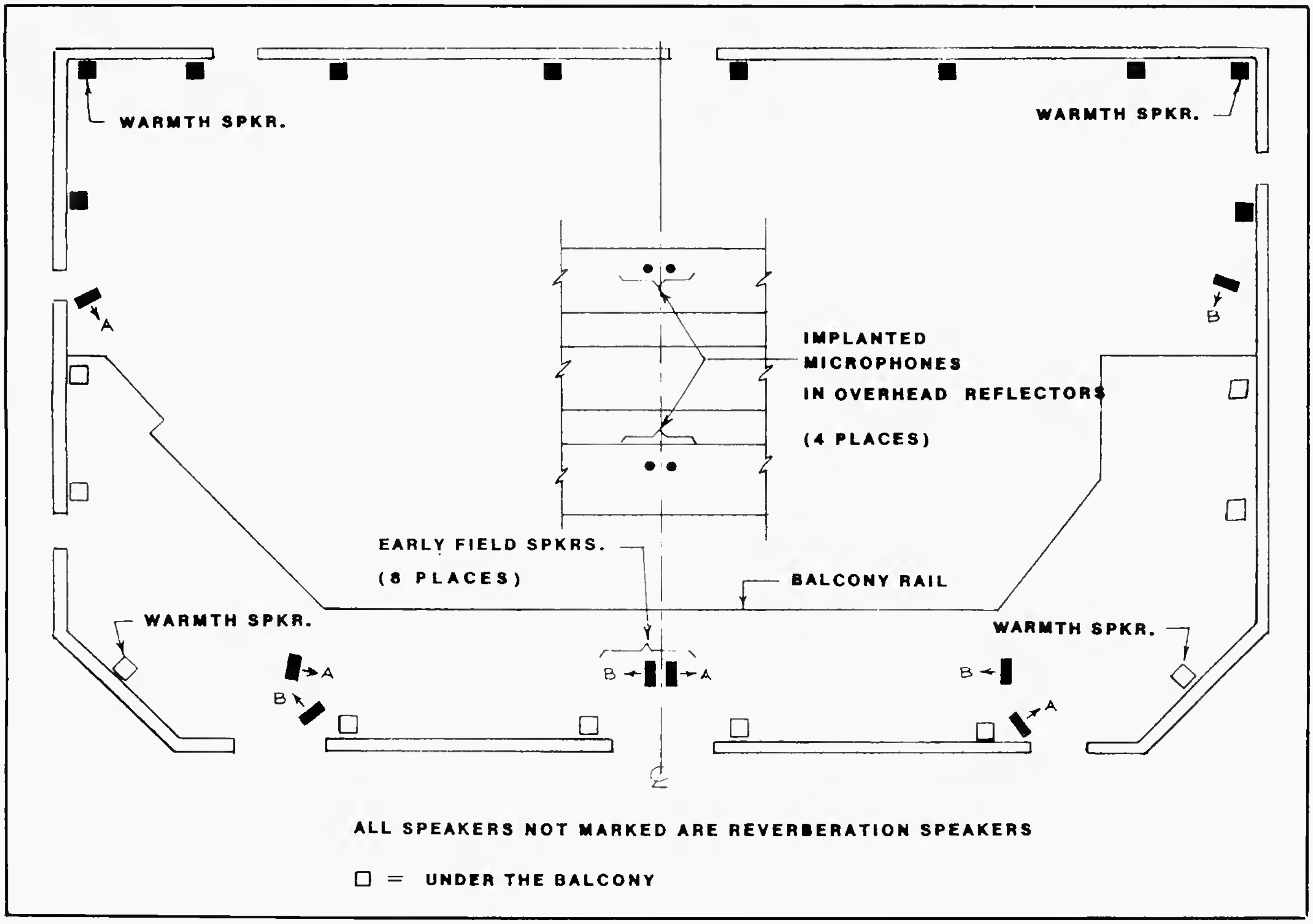
The original acoustics at 8H were rather "dead." Analysis of a full brass stop chord from a 1942 RCA recording of the NBC Symphony shows a mid-frequency reverberation time of 1.1 seconds, compared to typical symphony hall reverberation times of about 2.0 seconds. As television production began, the studio was made even more absorbent in order to facilitate boom mic'ing. This made symphonic performance impossible because of the lack of necessary cues. (Today, Studio 8H is best known as the originating point of Saturday Night Live.)

Late in 1979, NBC Television planned a pair of live concerts by the New York Philharmonic Orchestra under Zubin Mehta, to originate from Studio 8H. The first was to be a tribute to Toscanini and the NBC Symphony, the second a tribute to Enrico Caruso.

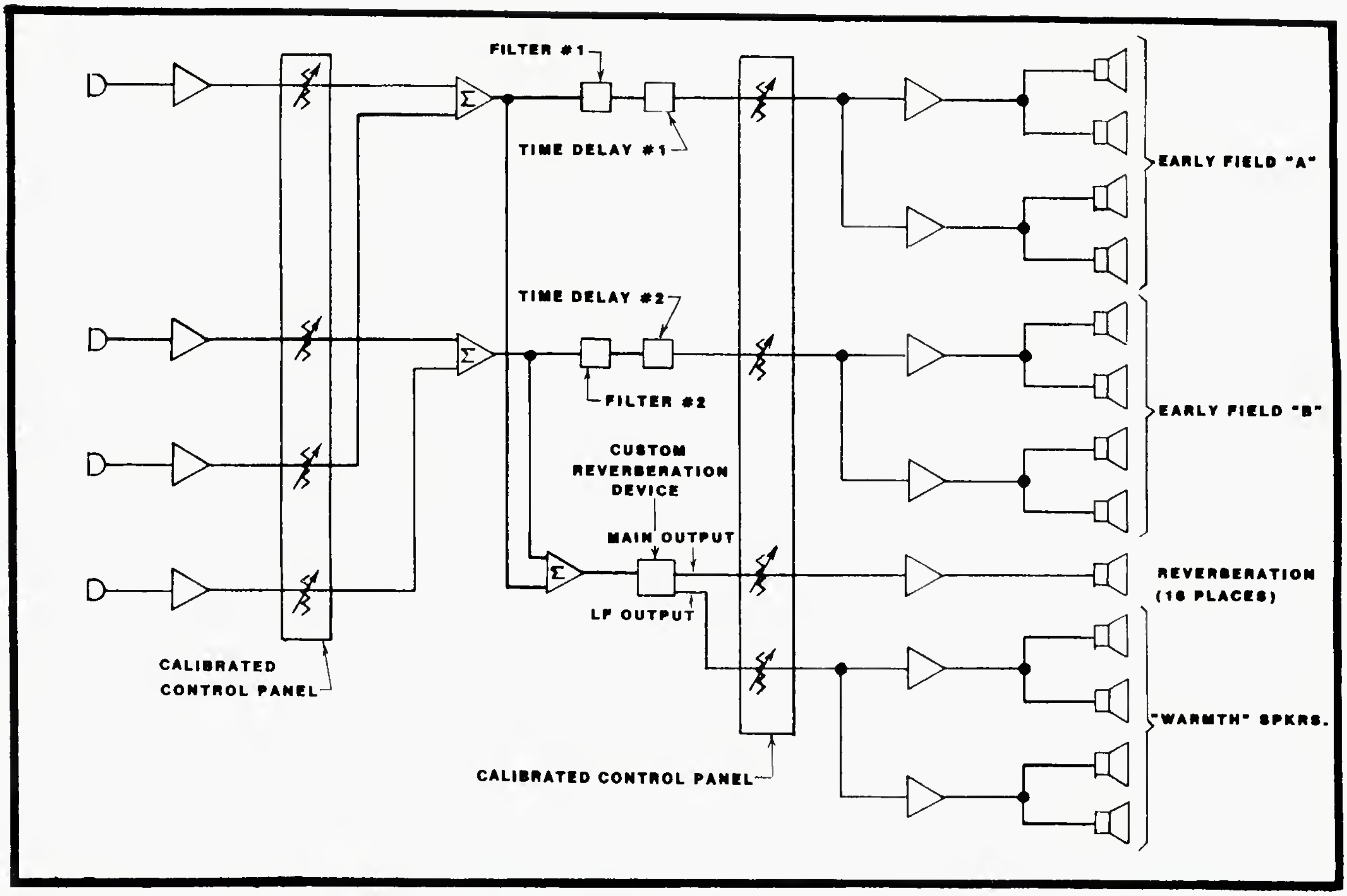
Conductor Mehta and the Symphony board, uneasy about performing in the "dry" studio, approached Jaffe Acoustics, Inc., to develop symphony-hall ambience in which the orchestra could perform.

Because of the room's low volume and primary use, few physical changes could be made. Even so, the small volume could not develop the necessary reverberance for symphonic performance. The only solution was to derive the acoustics through "electronic architecture."

Jaffe Acoustics designed the Studio 8H Electronic Reflected Energy System (ERES) and a concert enclosure (shell) to provide the needed acoustics for symphonic



NBC Studio 8H ERES System



NBC STUDIO 8H ERES SYSTEM

performance and balance. Maestro Mehta required that the electronic architecture system be in operation for all rehearsals as well as the broadcasts.

The live broadcasts, from 9:30 pm to 11:00 pm January 9, 1980, and January 16, 1982, were produced by Alvin Cooperman and Judith DePaul and staged for television by British director Rodney Greenberg. Audio producer was Andrew Kazdin; broadcast sound consultant was Robert Liftin; broadcast mix engineer was Scott Schachter; and host was Martin Bookspan.

For the 1980 performance, the orchestra was joined by Metropolitan Opera soloist Leontyne Price and violinist Itzak Perlman. Guest soloist for the 1982 performance was tenor Placido Domingo.

The ERES installation was by Filmways Audio Services, under Jaffe direction. It provided early reflections, "warmth," and reverberation. Four miniature Knowles condenser microphones flush-mounted in the canopy over the orchestra fed the system, which uses Spectra-Sonics, UREI, Industrial Research Products components, and Bozak loudspeakers. Electronic equipment is contained in two rack cabinets, one for low-level and line-level signals, the other for power amplification.

Two independent early-reflection processing channels each fed four small column speakers giving the correct lateral arriving directions for the audience in the small balcony. The frequency range for these channels was 250 Hz to 8000 Hz and timing was adjusted for a typical concert hall initial-time-delay-gap of about twenty milliseconds. Early reflections for the orchestra were primarily from the surfaces of the shell above and around them.

A "warmth" processing channel fed signals from 20 Hz to 250 Hz to four large dual 12-inch sealed bass speaker systems located in the corners of the studio, timed to coincide with the early portion of the reverberant field.

Sixteen reverberation systems, each using a single 12-inch driver in a sealed cabinet, were arrayed around the studio perimeter, some floor-mounted, and some hung from the walls. These handled a bandwidth of 20 Hz to 2500 Hz and by permitting a higher sound pressure level in the reverberant field provided a greater perception of reverberant energy than the time measurements alone would suggest. An average increase in reverberation time of thirty-four percent was achieved over the octave bands from 63 Hz to 4 KHz, with as much as forty-nine percent increase in the 63 Hz octave band:

OCTAVE CENTER	SYSTEM OFF	SYSTEM ON
63 Hz	0.86 sec.	1.28 sec.
125	1.10	1.33
250	1.00	1.20
500	0.91	1.28
1 kHz	0.83	0.94
2	0.71	0.82
4	0.41	0.72

This was the first use of a special reverberation device jointly developed by Industrial Research Products, Inc., and Jaffe Acoustics, Inc. It stemmed from the findings that electrical recirculation produced an unnatural character in a room, as did high echo density within the reverberator. It proved necessary for the reverberator to pass signal just once, with no recirculation (requiring a large digital memory), and to have a density always less than that of the early portion of the natural reverberant field of a room. Microphone placement is arranged so that an integration of the direct sound and reflections is fed to the reverberator. The natural density takes precedence over the reverberator's density and prolongs reverberation with the room's natural character. There is no change in tonality during the "tail." More recent ERES reverberation development includes the ability to generate up to several seconds of perceptibly natural reverberance, very high diffusion, incoherence, and "animation" of reverberant decay with a limited number of loudspeakers. Use of a Reverberation On Demand System (RODS computer, a development of Peter W. Barnett of Acoustic Management Systems, Ltd., London, UK) permits any amount of reverberant energy without the possibility of feedback.

The use of electronic architecture at Studio 8H allowed the New York Philharmonic Orchestra and its soloists to perform in an otherwise unsuitable space, and simplified mic'ing for broadcast. Some additional reverberation was added to the broadcast feed to compensate for close mic'ing. An FM stereo simulcast accompanied the television broadcasts.

ERES is the only electronic architecture system which develops, in addition to reverberation, three-dimensional early reflections and "warmth." It is manufactured in modular card-frame-based form by Technical Acoustics, Inc., South Norwalk, Connecticut.

OTHER STUDIOS WITH ELECTRONIC ARCHITECTURE

Philips MCR (Multi-Channel Reverberation) Systems are installed in two broadcast studios: the Hans Rosbaud Studio of Sudwesfunk (Southwest German Broadcasting) in Baden-Baden, and in Studio No. 1 at the new Limehouse Television Production Centre, West India Docks, London.

The Rosbaud Studio was built in 1950 as a music broadcast, rehearsal and concert facility with a small audience (330 seats). It corresponds somewhat to a university recital hall. Poor acoustics and inadequate reverberation led to a renovation in 1981 with the addition of the 70-channel MCR installation. Microphones are in a regular rectangular array on the ceiling, and loudspeakers are on the side walls. Two sensing microphones are suspended lower than the seventy system mics; their purpose is to sense an isolated fortissimo exceeding 112 dB and compress the seventy channels, avoiding overload.

Gain of the seventy channels is controlled by a single "reverberation time" knob at the mixing console, giving ten steps from no augmentation to approximately fifty percent augmentation of reverberation. The new flexibility, and the enhanced acoustics, have improved broadcast quality and given players greater satisfaction in their performances. The reverberance can be set for any type of performance, and can be changed instantly.

The newest studio electronic architecture is at Limehouse Studio 1 in London, which opened in late 1983. The concept behind the Limehouse complex was to produce the very best in terms of studio space, technical services, equipment and staff, and to provide the most supportive environment for creativity in television production.

While the purpose of Limehouse's MCR installation was to provide musical performance acoustics, the staff has made some unexpected discoveries. It turns out that by augmenting the reverberance slightly, studio audiences for panel games, situation comedies and the like, laugh and applaud more vigorously and spontaneously.

Acoustics can be varied instantly according to the scene created on the set, and actors are also stimulated by performing within the appropriate acoustic environment for each scene. Acoustic conditions can be set for outdoor, indoor, intimate, or large-scale portrayals. Choral, operatic, and symphonic performances all benefit from the electro-acoustic enhancement.

The Limehouse MCR installation has seventy-nine channels. The master control is at the mixing console, which can override a secondary control in the studio itself. The studio has dimensions of 24.8 meters by 12 meters high, and its natural acoustic is absorbent.

APPLICATIONS AND BENEFITS

European broadcast organizations have expressed considerable interest in variable electroacoustics, and have discovered unexpected benefits. Limehouse specified it at the start of construction in order to permit a wider range of events to be originated.

Studios have found not only the right acoustics for various musical performances, with musicians better able to perform, they have also found mic'ing and on-air sound improved. The familiar studio techniques of adding reverberation are still valuable, to compensate for close-mic'ing and to create additional effects.

The "Live from Studio 8H" project with NBC Television demonstrated the benefit of electronic architecture in transforming an American teleproduction studio into a good space for music origination. The simplicity, cost-effectiveness, modularity, and controllability of ERES and other electronic architecture equipment make it a valuable adjunct for any major recording or broadcast studio.

With such systems, performers can interact creatively with an ambience they themselves have composed for and visualized, rather than trying to create it "after the fact." "Walls" can be positioned closer or farther away than the actual studio walls; "reflectors" of various sizes and shapes can be floated overhead; "ceilings" can be raised or lowered; tonal balances in the reverberant field can change perceptions; and "impossible" spaces can be created—all electronically. The elements of acoustics can become performance instruments or composition elements in themselves.

One interesting option is applying a "skeleton" ERES to a location recording venue which has poor acoustics. By positioning a few loudspeakers appropriately near the microphones, appropriate directional reflections and ambience can be created in their vicinity which is heard on the recording as if the entire hall were architecturally transformed.

The recording studio is the ideal place for the fullest flowering of "electronic architecture," a technology which has already proven itself in concert halls, pavilions, churches, and studios. The studio that installs such systems will have a distinct competitive edge.

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db Buyer's Guide

Power Amplifiers

THIS MONTH'S buying guide is focused on that extremely critical component—the power amplifier. Manufacturers were asked to respond to specifications that were chosen to assist in the understanding and value of the equipment. The price would normally vary according to the quality of the circuitry, hence initiating the classic dollar vs. value dilemma.

As with any piece of audio equipment, there are always exceptions to the high quality/high price rule. Also, in some cases, sacrifices in certain specs might be made in order for the buyer to attain certain other desired features. In order to be unbiased, all manufactureres of which we are aware are included. (If a manufacturers seems to be absent it's because they didn't send back the information forms.)

Some manufacturers seem to have more specs than others. Not everyone who filled out the forms may have had all the information at their disposal, and it certainly wasn't a cover up by the manufacturer. It may simply indicate that the response was not verified at certain ratings (ie: IM, 1/4 watt, %.)

We provided the manufacturer with charts to cover specified things. We then re-wrote them to a paragraph form to make for an easier read. However, these are the specs asked for: 1/4 watt and full rated power in both intermodulation and total harmonic distortion. The 1/4 watt IM and THD numbers refer to the percentage of distortion while operating at low to moderate listening levels such as would be encountered in small studios with monitors and/or when listening with headphones. Distortion should be very low at these levels, since your ear is most sensitive to non-linearities (dissonance) at such levels. The rated power IM and THD on the other hand, tells you what the amplifier is doing when it is doing all it can; that is, at maximum power such as you would only encounter in multi-speaker rigs or at peak musical levels in studios.

Continuous power/channel is all channels of an amplifier with a common power supply driven to just before distortion/clipping—in other words, the maximum undistorted power the amplifier can provide. It's important to know that there is a relationship between amplifier power and rated handling capability of speaker systems. In a home studio system this is relatively unimportant. Speakers rated at 150 watts could be used with 300 watt amplifiers under these conditions. However, in a multi-speaker system such as is used in large areas (indoor and outdoor) careful matching of amplifier power and speaker load capabilities is required.

We also asked for High Level Sensitivity information. This is given in volts and is important that this voltage must come from your console or preamp if you are to be able to drive the amplifier to full power. Power bandwidth is the frequency range over which a power amplifier produces at least one half of its rated output power. This is most critical at bass and high end extremes, where at least half power (-3 dB) is needed. The spec for frequency response describes what the frequency response is when the signal varies in amplitude, generally less than 3 dB. Obviously, the flatter the response (lower dB) the better. This is for the most part one of the most significant specifications when comparing amplifiers. The circuitry involved in producing a "flatter" response takes more sophistication, therefore necessitating a higher price. The benefit of having amplification with a flatter response is more accurate reproduction with amplification and this superior circuitry results in better distortion specifications.

The other items that we asked the manufacturers had to do with physical charactistics such as the number of channels, dimension, weight and, of course, price. Note that virtually all amplifiers fit standard 19-inch racks, so if this is important to you, note it particularly.

Finally, we asked them all to give us any comments that they felt made their product unique or special in some way, and we have included their particular approach to the practical compromises necessary beween the various features and price.

All power ratings are minimum continous power per channel, at 8 ohms, with all channels driven. Frequency response is rated at 1/4 watt and is ±3 dB unless specified differently. All dimensions are H x W x D, in inches.

AMR

The PMA-200 is a two-channel, 100 watt amplifier with a power bandwidth of 100 kHz. THD is 0.005% at full rated power. Frequency response is 10 Hz-50 kHz, +0/-1 dB; s/n is 100 dB (below full power), and sensitivity is 1V. Features peak reading front panel LED meters, switchable DDT compression, thermal and short circuit protection. Dimensions are 5.25 x 19 x 12.125; weight is 29 lbs.
Price: $399.50.

ALTEC LANSING CORP.

The 1268 is a two-channel, 60 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.03% at 1/4 watt and full rated power. THD is 0.01% at full rated power. Frequency response is 30 Hz-20 kHz, ±0.25 dB; s/n is 95 dB, and high level sensitivity is 0.775V. Features Peak/Error computer, stepped input attenuator, and load protection from transients. Dimensions are 3.5 x 19 x 10; weight is 31 lbs.
Price: $1,072.00.

The 1269 is a two-channel, 120 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM and THD are the same as the 1268, above. Frequency response is 20 Hz-20 kHz, ±0.25 dB; s/n is 95 dB, and high level sensitivity is 0.775V. Features and dimensions are the same as the 1268, above. Weight is 45 lbs.
Price: '$1,456.00.

The 1590E is a single-channel, 200 watt amplifier with a power bandwidth of 30 Hz-15 kHz. THD at full rated power is less than 1%. Frequency response is 30 Hz-15 kHz, ±1 dB; s/n is 85 dB, and high level sensitivity is 0.8V. Features 70 and 200V output, optional input transformer, and battery backup. Dimensions are 10.5 x 19 x 8.25; weight is 41 lbs.
Price: $1,536.00.

The 1593C is a single-channel, 50 watt amplifier with a power bandwidth of 50 Hz-15 kHz and a THD of less than 1%, at full rated power. Frequency response is 30 Hz-15 kHz, ±1 dB; s/n ratio is 85 dB, and high level sensitivity is 0.8V. Features are similar to the 1590E, above, but it has 4, 8, 16 ohm and 70V outputs. Dimensions are 5.25 x 19 x 7.38; weight is 23 lbs.
Price: $1,004.00.

The 1594C is a single-channel, 100 watt amplifier with a power bandwidth of 40 Hz-15 kHz and a THD of less than 1%, at full rated power. Frequency response is 30 Hz-15 kHz, ±1 dB; s/n ratio is 85 dB, and high level sensitivity is 0.8V. Features are the same as the 1593C, above. Dimensions are 7 x 19 x 17.6; weight is 35 lbs.
Price: $1,176.00.

The 2200A is an 8-channel (max.), 75 watt amplifier with a power bandwidth of 20 Hz-20 kHz. THD is 0.03% at 1/4 watt and 0.25% at full rated power. Frequency response is 10 Hz-20 kHz, ±0.5 dB; s/n ratio is 96 dB, and high level sensitivity is 0.6V. Unit can be configured in many different variations with a wide choice of output power and channels. It will drive from 2 ohms to 70V lines. Dimensions are 7 x 19 x 17.6; weight is 70 lbs.
Price: $4,836.00.

The 2280A is an 8-channel (max.), 78 watt amplifier with a power bandwidth of 10 Hz-20 kHz. IM is 0.1% at 1/4 watt and full rated power. THD is 0.02% at 1/4 watt and 0.1% at full rated power. Frequency response is 10 Hz-20 kHz, ±0.5 dB; s/n ratio is 92 dB, and high level sensitivity is 0.775V. Features, dimensions and weight are the same as the 2200A, above, but it also has LED indication of signal or failure mode.
Price: $4,908.00.

The 1270B is a two-channel, 220 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.03% at 1/4 watt and full rated power. THD is 0.01% at full rated power. Frequency response is 20 Hz-20 kHz, s/n ratio is 95 dB, and high level sensitivity is 0.775V. Features are the same as the 1269, above. Dimensions are 5.25 x 19 x 15; weight is 51.5 lbs.
Price: $2,040.00.

ASHLY AUDIO

The FET-200 is a two-channel, 123 watt amplifier with a power bandwidth of 10 Hz-100 kHz. IM is 0.007% at 1/4 watt and full rated power and THD is 0.004% at 1/4 watt and full rated power. Frequency response is 10 Hz-50 kHz, ±0.5 dB; s/n ratio is 110 dB, and high level sensitivity is 1.4V. Features MOSFET output circuitry, fan cooling, meters, balanced inputs, and mono and bridging switch. Dimensions are 3.5 x 19 x 16; weight is 33 lbs.
Price: $799.00.

The FET-500 is a two-channel, 300 watt amplifier with the same specs and features as the FET-200, above, except the high level sensitivity is 1.7V. Dimensions are 5.25 x 19 x 16; weight is 62 lbs.
Price: $1,150.00.

BGW

The 2125 is a single-channel, 110 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.05% at 1/4 watt and full rated power. THD is 0.1% at 1/4 watt and full rated power. Frequency response is 1 Hz-50 kHz, s/n ratio is 100 dB, and high level sensitivity is 0.77V. Features 70 volt output and high pass filter. Dimensions are 3.5 x 19 x 12; weight is 31 lbs.
Price: $589.00.

The 320B is a two-channel, 100 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.05% at 1/4 watt and full rated power and the THD is 0.2% at 1/4 watt and full rated power. All other specs are the same as the 2125, above. Features 70V output, stepped attenuators and magnetic breaker. Dimensions are 5.25 x 19 x 11.75; weight is 39 lbs.
Price: $939.00.

The 620B is a two-channel, 200 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.06% at 1/4 watt and full rated power and THD is 0.25% at 1/4 watt and full rated power. Frequency response is 1 Hz-70 kHz, s/n ratio is 100 dB, and high level sensitivity is 0.7V. Features are the same as the 320B, above. Dimensions are 8.75 x 19 x 11.75; weight is 58 lbs.
Price: $1,239.00.

The 250D is a two-channel, 100 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.02% at 1/4 watt and full rated power and THD is 0.1% at 1/4 watt and full rated power. Frequency response is 1 Hz-90 kHz, s/n ratio is 110 dB, and high level sensitivity is 1.41V. Features stepped attenuator, magnetic circuit breaker, and 2 ohm operation. Dimensions are 5.25 x 19 x 11.75; weight is 33 lbs.
Price: $869.00.

The 150 is a two-channel, 50 watt amplifier with the same specs as the 250D, above, except the s/n ratio is 109 dB, and high level sensitivity is 1V. The dimensions are 1.75 x 19 x 11.5; weight is 17 lbs.
Price: $699.00.

The 6500 is a two-channel, 100 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.02% at 1/4 watt and full rated power and THD is 0.1% at 1/4 wat and full rated power. Frequency response is 1 Hz-100 kHz, s/n ratio is 102 dB, and high level sensitivity is 1.23V. Dimensions are 3.5 x 19 x 15.5.
Price: $749.00.

The 7500 is a two-channel, 200 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.002% at 1/4 watt and full rated power and THD is 0.1% at 1/4 watt and full rated power. Frequency response is 1 Hz-100 kHz, s/n ratio is 110 dB, and high level sensitivity is 1.23V. Dimensions are 5.25 x 19 x 10.5.
Price: $899.00.

BIAMP SYSTEMS

The 1200 is a two-channel, 150 watt amplifier with a power bandwidth of 10 Hz-62 kHz. IM is 0.02% at 1/4 watt and 0.05% at full rated power. THD is 0.06% at 1/4 watt and 0.05% at full rated power. Frequency response is 10 Hz-35 kHz, ±1 dB; s/n ratio is 103 dB, and high level sensitivity is 1.4V. Features fully balanced symetrical design and Auto-Limit circuit. Dimensions are 5.25 x 19 x 10; weight is 44 lbs.
Price: $849.00.

The 2400 is a two-channel, 310 watt amplifier with the same specs as the 1200, above, except it has a 108 dB s/n ratio and a high level sensitivity of 2V. Features are the same as the 1200, above, but it can also provide 1320 watts in mono bridged mode. Dimensions are 7 x 19 x 14; weight is 56 lbs.
Price: $1,199.00.

BRYSTON

The 2B-LP is a two-channel, 50 watt amplifier with a power bandwidth of 0.5 Hz-100 kHz. IM is 0.01% at 1/4 watt and full rated power. THD is 0.01% at 1/4 watt and full rated power. Frequency response is 0.5 Hz-50 kHz, ±0 dB; s/n ratio is 100 dB, and high level sensitivity is 0.75V. Features bridgeability for mono operation, dual power supplies, and 60V/usec slew rate. Dimensions are 1.75 x 19 x 10; weight is 18 lbs.
Price: $600.00; available in balanced version for $50.00 more.

The 3B is a two-channel, 100 watt amplifier with the same specs and features as the 2B-LP, above, except the high sensitivity level is 1V. Dimensions are 5.25 x 19 x 9; weight is 35 lbs.
Price: $1,075.00.

The 4B is a two-channel, 200 watt amplifier with the same specs and features as the 2B-LP, above, except the high sensitivity level is 1.25V. Dimensions are 5.25 x 19 x 9; weight is 50 lbs.
Price: $1,600.00.

The 6B is a single-channel, 500 watt amplifier with the same specs, dimensions and weight as the 4B, above. It is also switchable to 100 watt, high current operation.
Price: $1,700.00.

CARVER

The PM-200 is a two-channel, 100 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM and THD is 0.15% at 1.4 watt and full rated power. Frequency response is 3 Hz-60 kHz, s/n ratio is 103 dB, and high level sensitivity is 1.4V. Dimensions are 2.9 x 19 x 9.7; weight is 14 lbs.
Price: $499.00.

The PM-1.5 is a two-channel, 450 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM and THD is 0.5% at 1/4 watt and full rated power. Frequency response is 3 Hz-80 kHz, s/n ratio is 115 dB, and high level sensitivity is 3V. Features electronic multifunction protection. Dimensions are 3.5 x 19 x 9.7; weight is 21 lbs.
Price: $995.00.

The PM-1.5 Mono Block is a single-channel, 1,200 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM and THD is 0.5% at 1/4 watt and full power rating. Frequency response is 3 Hz-80 kHz, s/n ratio is 115 dB, and high level sensitivity is 3V. Features, dimensions and weight are the same as the PM-1.5, above.
Price: $1,195.00.

The PM-1.5L is a two-channel, 450 watt (at 2 ohms) amplifier with a power bandwidth of 20 Hz-20 kHz. IM and THD is 0.5% at 1/4 watt and full rated power. Frequency response is 3 Hz-80 kHz, s/n ratio is 115 db, and high level sensitivity is 3V. Features, dimensions and weight are the same as the PM-1.5, above.
Price: $1,095.00.

CARVIN

The DCA-800 is a two-channel, 200 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM and THD is 0.01% at 1/4 watt and 0.05% at full rated power. Frequency response is 5 Hz-60 kHz, +0/-3 dB; s/n ratio is 100 db, and high level sensitivity is 2V. Features modular construction, thermostatic protection, bridging, short-circuit protection and automatic forced air cooling. Dimensions are 5.25 x 19 x 12; weight is 44 lbs.
Price: $579.00.

The DCA-300 is a two-channel, 100 watt amplifier with the same specs, dimensions and features as the DCA-800, above. They also both feature damping greater than 200 and a slew rate of over 35V/usec. Weight is 37 lbs.
Price: $419.00.

CERWIN-VEGA

The LPA-600 is a two-channel, 350 watt amplifier with a power bandwidth of 7 Hz-60 kHz. IM and THD is 0.03% at 1/4 watt and full rated power. Frequency response is 7 Hz-46 kHz, s/n ratio is 115 dB, and high level sensitivity is 1.4V. Features dual secondary power transformer, protection circuitry, 2-speed forced air cooling, and switchable bridging. Dimensions are 5.25 x 19 x 18; weight is 54 lbs.
Price: $1,700.00.

CREST AUDIO

The 4000/01 is a two-channel, 325 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.001% at 1/4 watt and less than 0.01% at full rated power. THD is 0.003% at 1/4 watt and 0.006% at full rated power. Frequency response is 1 Hz-50 kHz, ±0.1 dB; s/n ratio is 106 dB. Available with or without LED bargraph meters. Dimensions are 5.25 x 19 x 13; weight is 58 lbs.
Price: $1,960.00.

The 3000/01 is a two-channel, 240 watt amplifier with the same specs and features as the 4000/01, above. Dimensions are 5.5 x 19 x 11.5; weight is 46 lbs.
Price: $1,520.00.

The 2501A is a two-channel, 200 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.001% at 1/4 watt and less than 0.01% at full rated power. THD is 0.004% at 1/4 watt and 0.01% at full rated power. Frequency response is 20 Hz-20 kHz, ±0.2 dB; s/n ratio is 96 dB, and high level sensitivity is 1V. Will deliver 1,100 watts mono into 4 ohms. Dimensions are 3.5 x 19 x 13; weight is 38 lbs.
Price: $1,199.00.

The 2001A is a two-channel, 125 watt amplifier with the same specs and dimensions as the 2501A, above, except the high level sensitivity is 0.79V. Weight is 32 lbs.
Price: $999.00.

The 1501A is a two-channel, 80 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.001% at 1/4 watt and less than 0.01% at full rated power. THD is 0.004% at 1/4 watt and 0.01% at full rated power. Frequency response is 1 Hz- 50 kHz, ±0.1 dB; s/n ratio 95 dB, and a high level sensitivity of 0.63V. Dimensions are 1.75 x 19 x 10.5; weight is 19 lbs.
Price: $799.00.

The 1001A is a two-channel, 335 watt amplifier with the same specs and dimensions as the 1501A, above, except the high level sensitivity is 0.47V. Weight is 16 lbs.
Price: $649.00.

The POWERLINE 400 is a two-channel, 290 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.001% at 1/4 watt and less than 0.01% at full rated power. THD is 0.01% at 1/4 watt and 0.05% at full rated power. Frequency response is 20 Hz-20 kHz, ±0.2 dB; s/n ratio is 100 dB, and high level sensitivity is 1V. Dimensions are 3.5 x 19 x 13; weight is 38 lbs.
Price: $1,160.00.

The POWERLINE 300 is a two-channel, 220 watt amplifier with the same specs and dimensions as the Powerline 400, above, except the high level sensitivity is 1.17V. Weight is 32 lbs.
Price: $879.00.

CROWN

The PS-200 is a two-channel, 90 watt amplifier with a power bandwidth of 1 Hz-20 kHz. IM is 0.05% at 1/4 watt and full rated power. THD is 0.001% at 1/4 watt and full rated power. Frequency response is DC-20 kHz, ±0.1 dB; s/n ratio is 112 dB, and high level sensitivity is 1.3V. Features distortion and signal presence indicators, low frequency protection, adjustable turn-on delay, and massive heat sinks. Dimensions are 5.25 x 19 x 10.13; weight is 25 lbs.
Price: $769.00.

The DELTA-OMEGA 2000 is a single-channel, 730 watt amplifier with a power bandwidth of DC-45 kHz. IM and THD is 0.05% at 1/4 watt and full rated power. Frequency response is DC-45 kHz, ± 0.1 dB; s/n ratio is 120 dB, and high level sensitivity is 1.76V. Features compensation for load impedance to improve transient response, indicators, and 70V option. Dimensions are 8.75 x 19 x 16.5; weight is 92 lbs.
Price: $2,995.00.

The D-75 is a two-channel, 40 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.05% at 1/4 watt and full rated power and THD is 0.001% at 1/4 watt and full rated power. Frequency response is 20 Hz-20 kHz, ±0.1 dB; s/n ratio is 110 dB, and high level sensitivity is 0.812V. Features full protection circuitry. Dimensions are 1.75 x 19 x 9; weight is 10 lbs.
Price: $499.00.

The D-150A is a two-channel, 80 watt amplifier with a power bandwidth of 1 Hz-20 kHz. IM is 0.05% at 1/4 watt and full rated power and THD is 0.001% at 1/4 watt and full rated power. Frequency response is DC-20 kHz, ±0.1 dB; s/n ratio is 110 dB, and high level sensitivity is 1.19V. Features are the same as the D-75, above, and it has distortion indicator. Dimensions are 5.25 x 19 x 8.75; weight is 24 lbs.
Price: $729.00.

The DC-300A is a two-channel, 155 watt amplifier with the same specs and features as the D-150A, above, except the high level sensitivity is 1.75V. Dimensions are 7 x 19 x 9.75; weight is 45 lbs.
Price: $1,149.00.

The MICRO-TECH 600 is a two-channel, 220 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.05% a t 1/4 watt and full rated power. THD is 0.05 at full rated power. Frequency response is 20 Hz-20 kHz, ±0.1 dB; s/n ratio is 105 dB, and high level sensitivity is 0.775V. Features reversable air flow, computer controlled protection and grounded bridge output. Dimensions are 3.5 x 19 x 16; weight is 39 lbs.
Price: $899.00.

The MICRO-TECH 600LX is the same as the Micro-Tech 600, above, plus it has front panel level controls and status indicators, and XLR inputs.
Price: $1,169.00.

The MICRO-TECH 1200 is a two-channel, 320 watt amplifier with same specs, dimensions and features as the Micro-Tech 600, above. Weight is 44 lbs. Price: $1,169.00. (Also available as the 1200LX, as above.)
Price: $1,369.00.

FOSTEX

The A300 is a two-channel, 100 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM and THD is 0.05% at full rated power. Frequency response is 10 Hz-50 kHz, ±0.25 dB; s/n ratio is 97 dB, and high level sensitivity is 0.8V. Features ultra fast rise time, full load protection, and input attenuators. Dimensions are 3.5 x 19 x 15; weight is 28 lbs.
Price: $699.00.

The A600 is a two-channel, 200 watt amplifier with same specs and features as the A300, above, except the s/n ratio is 95 dB. Dimensions are 5.25 x 19 x 15; weight is 36 lbs.
Price: $995.00.

DAVID HAFLER

The P500 is a two-channel, 255 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.007% at 1/4 watt and full power rating and THD is 0.025% at 1/4 watt and full power rating. Frequency response is 8 Hz-85 kHz, s/n ratio is 95 dB and, high level sensitivity is 1.55V. Features mono switchability, 3-speed fan, clip and signal present lights. Dimensions are 7.75 x 19 x 14; weight is 53 lbs.
Price: $995.00.

The P225 is a two-channel, 115 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.005% at 1/4 watt and full rated power and THD is 0.02% at 1/4 watt and full rated power. Frequency response is 2 Hz-160 kHz, s/n ratio is 100 dB, and high level sensitivity is 1.55V. Dimensions are 5.5 x 19 x 11.75; weight is 32 lbs.
Price: $525.00.

The P505 is a two-channel, 255 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.007% at 1/4 watt and full rated power, and THD is 0.025% at 1/4 watt and full rated power. Frequency response is 2 Hz-120 kHz, s/n ratio is 100 dB, and high level sensitivity is 2.3V. Dimensions are 7.25 x 19 x 14; weight is 50 lbs.
Price: $850.00.

The P125 is a two-channel, 62 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.005% at 1/4 watt and full rated power, and THD is 0.009% at 1/4 watt and full rated power. Frequency response is 4 Hz-200 kHz, s/n ratio is 100 dB, and high level sensitivity is 1.1V. Dimensions are 3.5 x 19 x 9.
Price: $395.00.

HILL AUDIO

The DX1000 is a two-channel, 375 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.002% at 1/4 watt and full rated power. THD is 0.003% at 1/4 watt and 0.05% at full rated power. Frequency response is 20 Hz-20 kHz, +0/-0.5 dB; s/n ratio is 100 dB, and high level sensitivity is 1.5V. Features dual power supplies, transformer coupled driver stage and short circuit protection. Dimensions are 5.25 x 19 x 16; weight is 57 lbs.
Price: $1,765.00.

The DX1000A is a two-channel, 500 watt amplifier with the same specs, features and dimensions as the DX1000, above. Weight is 58 lbs.
Price: $1,999.00.

The DX2000 is a two-channel, 400 watt amplifier with the same specs and features as the DX1000, above. Dimensions are 5.25 x 19 x 19; weight is 78 lbs.
Price: $2,275.00.

The DX3000 is a two-channel, 550 watt amplifier with the same specs, features and dimensions as the DX2000, above. Weight is 80 lbs.
Price: $2,650.00.

The DX500 is a two-channel, 280 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.003% at 1/4 watt and 0.012% at full rated power. THD is 0.005% at 1/4 watt and 0.01% at full rated power. Frequency response is 20 Hz-20 kHz, +0/-0.2 dB; s/n ratio is 100 dB, and high level sensitivity is 1.5V. Dimensions are 3.75 x 19 x 13; weight is 37 lbs.
Price: $1,149.00.

JBL PROFESSIONAL

The 6215 is a two-channel, 35 watt amplifier with mono/stereo bridge switch, 5-way output binding posts and front panel headphone jack. Dimensions are 1.75 x 19 x 10; weight is 18 lbs.
Price: $576.00.

The 6230 is a two-channel, 110 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.03% at 1/4 watt and full rated power. THD is 0.1% at 1/4 watt and 0.2% at full rated power. Frequency response is 20 Hz-20 kHz, +0/-1 dB; s/n ratio is 100 dB, and high level sensitivity is 1.1V. Features are the same as the 6215, above, except without headphone jack. Dimensions are 5.25 x 19 x 10.5; weight is 26 lbs.
Price: $618.00.

The 6260 is a two-channel, 190 watt amplifier with same specs and features as the 6230, above. Dimensions are 7 x 19 x 10.5; weight is 45 lbs.
Price: $870.00.

The 6290 is a two-channel, 340 watt amplifier with same specs and features as the 6230, above. Other features are fan- cooled dual mono amplifiers. Dimensions are 7 x 19 x 16; weight is 85 lbs.
Price: $1,299.00.

PANASONIC (RAMSA)

The WP-9210 is a two-channel, 200 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.01% at 1/4 watt and full rated power, and THD is 0.05% at 1/4 watt and full rated power. Frequency response is 20 Hz-20 kHz, ±0.5 dB; s/n ratio is 105 dB, and high level sensitivity is 1.95V. Features load sensing and thermal protection. Dimensions are 5.75 x 16.5 x 14.5; weight is 43 lbs.
Price: $995.00.

PEAVEY

The M-3000 is a single-channel, 130 watt amplifier with a power bandwidth of 10 Hz-30 kHz. THD is 0.1% at 1/4 watt and full rated power. Frequency response is 10 Hz-30 kHz, +0/-1 dB; s/n ratio is 95 dB, and high level sensitivity is 1V. Features fan cooling, DDT compression, level controls, and LED array. Dimensions are 5.25 x 19 x 12.5; weight is 30 lbs.
Price: $379.00.

The M-2600 is a two-channel, 90 watt amplifier with a power bandwidth of 20 Hz-30 kHz. THD is 0.1% at 1/4 watt and full rated power. Frequency response is 20 Hz-30 kHz, +0/-1 dB; s/n ratio is 95 dB, and high level sensitivity is 1V. Features DDT compression, level controls, and LED status indicators. Dimensions are 5.25 x 19 x 12; weight is 28 lbs.
Price: $379.50.

The CS-400 is a two-channel, 120 watt amplifier with a power bandwidth of 5 Hz-40 kHz. IM is 0.1% at full rated power, and THD is 0.1% at 1/4 watt and full rated power. Frequency response is 5 Hz-40 kHz, +0/-1 dB; s/n ratio is 95 dB, and high level sensitivity is 1V. Features massive heat sinks, 2-speed fan, load protection, level controls, and clip indicators. Dimensions are 5.25 x 19 x 14; weight is 50 lbs.
Price: $599.00.

The CS-800 is a two-channel, 230 watt amplifier with the same specs and features as the CS-440, above, except the high level sensitivity is 1.3V. Dimensions are 7 x 19 x 13.5; weight is 59 lbs.
Price: $799.00.

The CS-1200 is a two-channel, 350 watt amplifier with a power bandwidth of 5 Hz-50 kHz. IM is 0.01% at full rated power, and THD is 0.003% at 1/4 watt and full rated power. Frequency response is 5 Hz-60 kHz, +0/-1 dB; s/n ratio is 103 dB, and high level sensitivity is 1.4V. Features 2-speed fan, DDT compression, level controls, LED arrays and fault indicators, and mono/stereo bridging. Dimensions are 7 x 19 x 17.75; weight is 70 lbs.
Price: $1,199.50.

The DECA-700 is a two-channel, 200 watt amplifier with a power bandwidth of 5 Hz-20 kHz. THD is 0.1% at 1/4 watt and full rated power. Frequency response is 5 Hz-20 kHz, +0/-1 dB; s/n ratio is 95 dB. Features mono/stereo bridging, level controls, LED arrays and fault indicators, and DDT compression. Dimensions are 3.5 x 19 x 13.5; weight is 30 lbs.
Price: $849.00.

The DECA-1200 is a two-channel, 300 watt amplifier with the same specs and features as the Deca 700, above, except s/n ratio is 105 dB. Dimensions are 3.5 x 19 x 15; weight is 26 lbs.

QSC

The 1080 is a two-channel, 35 watt amplifier with a power bandwidth of 12 Hz-60 kHz. IM is 0.01% at full rated power, and THD is 0.1% at full rated power. Frequency response is 20 Hz-20 kHz, +0/-0.5 dB; s/n ratio is 100 dB, and high level sensitivity is 0.83V. Features dual power supplies, DC and sub-audio speaker protection, and active balanced inputs. Dimensions are 1.75 x 19 x 8.7; weight is 12 lbs.
Price: $488.00.

The 1200 is a two-channel, 100 watt amplifier with the same specs and features as the 1080, above, except high level sensitivity is 1V, and it has optional 70V output transformer. Dimensions are 5.25 x 19 x 8.5; weight is 28 lbs.
Price: $548.00.

The 1400 is a two-channel, 200 watt amplifier with the same specs and features as the 1200, above. Dimensions are 7 x 19 x 8.5; weight is 37 lbs.
Price: $768.00.

The 1700 is a two-channel, 325 watt amplifier with the same specs and features as the 1200, above. Dimensions are 7 x 19 x 10.8; weight is 57 lbs.
Price: $1,0998.00.

The 3200 is a two-channel, 110 watt amplifier with a power bandwidth of 8 Hz-60 kHz. IM is 0.02% at 1/4 watt and full rated power. THD is 0.1% at full rated power. Frequency response is 8 Hz-300 kHz, +0/-3 dB; s/n ratio is 100 dB, and high level sensitivity is 1V. Features dual mono configuration, front removable channels, muting, and the same features as the 1200, above. Dimensions are 1.75 x 19 x 14.6; weight is 26 lbs.
Price: $958.00.

The 3350 is a two-channel, 200 watt amplifier with the same specs and features as the 3200, above. Dimensions are 3.5 x 19 x 15.9; weight is 41 lbs.
Price: $1,248.00.

The 3500 is a two-channel, 300 watt amplifier with same specs, features and dimensions as the 3350, above. Weight is 50 lbs.
Price: $1,488.00.

The 3800 is a two-channel, 375 watt amplifier with the same specs and features as the 3200, above. Other features are 850 watt rating at 2 ohms. Dimensions are 5.25 x 19 x 15.9; weight is 75 lbs.
Price: $1,958.00.

RANE

The 19A6 is a six-channel, 100 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.1% at 1/4 watt and full rated power, and THD is 0.2% at 1/4 watt and full rated power. Frequency response is 5 Hz-80 kHz, +0/-3 dB; s/n ratio is 90 dB, and high level sensitivity is 0.775V. Features a 15 dB limiter for each channel and auto-bridging. Dimensions are 5.25 x 19 x 11.5; weight is 44 lbs.
Price: $1,299.00.

SOUNDCRAFT

The SA 150 is a two-channel, 85 watt amplifier with a power bandwidth of 20 Hz-50 kHz. IM is 0.02% at 1/4 watt and full rated power. THD is 0.05% at 1/4 watt. s/n ratio is 100 dB. Features electronic protection from over voltage or excessive current and thermal protection. Dimensions are 1.75 x 19 x 17; weight is 25 lbs.

The SA 600 is a two-channel, 150 watt amplifier with the same specs and features as the SA 150, above. Dimensions are 3.5 x 19 x 17; weight is 40 lbs.

The SA 1000 is a two-channel, 300 watt amplifier with the same specs and features as the SA 150, above. Dimensions are 5.25 x 19 x 17; weight is 60 lbs.
Prices furnished upon request for all models. Please contact manufacturer.

SOUNDCRAFTSMEN

The PM860 is a two-channel, 205 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM and THD is 0.05% at 1/4 watt and full rated power. Frequency response is 20 Hz-20 kHz, ±0.1 dB; s/n ratio is 105 dB, and high level sensitivity is 1.5V. Features 450 watts at 2 ohms, 2-speed fan, clip indicators for each channel, and optional rack mount/cabinet panel. Dimensions are 5 x 8.5 x 14; weight is 22 lbs.
Price: $549.00.

The PR1800 is a two-channel, 375 watt amplifier with the same specs as the PM860, above, except the high level sensitivity is 1.22V. Features are MOSFET design, dual relay DC protection, compressor/limiter circuit and 2-speed fan. Dimensions are 5.25 x 19 x 17; weight is 60 lbs.
Price: $1,499.00.

The RA7501 is a two-channel, 250 watt amplifier with the same specs as the PR1800, above, except the s/n ratio is 110 dB and the THD is 0.09% at 1/4 watt and full rated power. Features Class H signal tracking power supply, no current limiting, overload protection, and clip indicators. Dimensions are 7 x 19 x 15; weight is 55 lbs.
Price: $899.00.

The RA6501 is a two-channel, 250 watt amplifier with the same specs, features and dimensions as the RA7501, above. Weight is 53 lbs.
Price: $799.00.

The RA7502 is a two-channel, 250 watt amplifier with the same specs, features and dimensions as the RA7501, above. Other features are metering for each channel. Weight is 55 lbs.
Price: $999.00.

The RA5502 is a two-channel, 125 watt amplifier with the same specs as the PR1800, above, except the high level sensitivity is 0.95V. Features MOSFET circuitry, LED power meters, and clip indicators. Dimensions are 5.25 x 19 x 15; weight is 30 lbs.
Price: $649.00.

SUNN/FENDER

The Sunn SA-20 is a two-channel, 300 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM and THD is 0.05% at 1/4 watt and full rated power. Frequency response is 2 Hz-200 kHz, s/n ratio is 108 dB, and high level sensitivity is 1V. Dimensions are 5.25 x 19 x 12; weight is 36 lbs.
Price: $649.00.

The Sunn SGA-310 is a single-channel, 120 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.04% at 1/4 watt and full rated power. THD is 0.08% at 1/4 watt and full rated power. Frequency response is 20 Hz-20 kHz, +0/-2 dB; s/n ratio is 95 dB, and high level sensitivity is 1V. Features 15 dB, 10-band graphic equalizer. Dimensions are 5.25 x 19 x 10.5; weight is 25 lbs.

The Sunn SPL-7000 is a two channel, 225 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM and THD is 0.05% at 1/4 watt and full rated power. Frequency response is 20 Hz-20 kHz, +0/-1 dB; s/n ratio is 100 dB, and high level sensitivity is 1.2V. Features bridge switch, active balanced inputs, and level controls. Dimensions are 5.25 x 19 x 13; weight is 40 lbs.
The Fender 2235 is identical to the Sunn SPL-7000, above, and it has XLR input connectors.

UREI

The 6150 is a two-channel, 100 watt amplifier with a power bandwidth of 20 Hz-20 kHz. IM is 0.05% at 1/4 watt and full rated power. THD is 0.1% at 1/4 watt and 0.2% at full rated power. Frequency response is 20 Hz-20 kHz, +0/-1 dB; s/n ratio is 100 dB, and high level sensitivity is 1.1V. Features bridging, active balanced inputs, full complementary circuitry. Dimensions are 1.75 x 19 x 14; weight is 22 lbs.
Price: $746.00.

The 6250 is a two-channel, 175 watt amplifier with the same specs and features as the 6150, above. Dimensions are 3.5 x 19 x 14; weight is 36 lbs.
Price: $896.00.

The 6300 is a two-channel, 310 watt amplifier with the same specs and features as the 6250, above. Also has fan cooling. Dimensions are 5.25 x 19 x 14; weight is 52 lbs.
Price: $1,346.00.

The 6500 is a two-channel, 360 watt amplifier with the same specs and features as the 6300, above. It also has dual mono amplifiers and plug in modules. Dimensions are 7 x 19 x 16; weight is 84 lbs.
Price: $2,396.00.

YAMAHA

The PC1002 is a two-channel, 100 watt amplifier with a power bandwidth of 10 Hz-100 kHz. IM is 0.01% at 1/4 watt and full rated power, and THD is 0.01 % at full rated power. Frequency response is 10 Hz-50 kHz, +0/-1 dB; s/n ratio is 105 dB, and high level sensitivity is 0.775V. Features input attenuators, multiple protection circuits and balanced XLR inputs. Dimensions are 5.5 x 19 x 13.25; weight is 34 lbs.
Price: $810.00.

The PC2002 is a two-channel, 240 watt amplifier with same specs and features as the PC1002, above, except THD is 0.003% at full rated power and s/n ratio is 110 dB. Dimensions are 7 x 19 x 16.25; weight is 44 lbs.
Price: $1,290.00.

The PC5002M is a two-channel, 500 watt amplifier with the same specs as the PC2002, above. Features dual mono amps, thermal and power protection, input attenuators, and illuminated peak meters. Dimensions are 10.4 x 19 x 19; weight is 134 lbs.
Price: $4,290.00.

The P2075 is a two-channel, 50 watt amplifier with a power bandwidth of 10 Hz-50 kHz. IM is 0.005% at 1/4 watt and full rated power. THD is 0.01% at 1/4 watt and 0.05% at full rated power. Frequency response is 10 Hz-30 kHz, +0/-1 dB; s/n ratio is 100 dB, and high level sensitivity is 1.23V. Dimensions are 3.9 x 19 x 14.5; weight is 20 lbs.
Price: $395.00.

The P1150 is a single-channel, 100 watt amplifier with a power bandwidth of 10 Hz-100 kHz. IM is 0.005% at 1/4 watt and full rated power. THD is 0.003% at 1/4 watt and 0.007% at full rated power. Frequency response is 10 Hz-50 kHz. +0/-1 dB; s/n ratio is 110 dB, and high level sensitivity is 1.23V. Dimensions are 5.25 x 19 x 16.6; weight is 28.5 lbs.
Price: $395.00.

The P1250 is a single-channel, 170 watt amplifier with the same specs and dimensions as the P1150, above. Weight is 33 lbs.
Price: $495.00.

The P2150 is a two-channel, 100 watt amplifier with a power bandwidth of 10 Hz-50 kHz. IM is 0.005% at 1/4 watt and full rated power and THD is 0.007% at 1/4 watt and full rated power. Frequency response is 10 Hz-50 kHz, +0/-1 dB; s/n ratio is 110 dB, and high level sensitivity is 1.23V. Dimensions are 5.25 x 19 x 16.6; weight is 37.5 lbs.
Price: $545.00.

The P2250 is a two-channel, 170 watt amplifier with the same specs and dimensions as the P2150, above. Weight is 42 lbs. Price: $695.00.

ALTEC CORPORATION
PO Box 26105
Oklahoma City, OK 73126

AMR
Rt. 2, Highway 503
DeCatur, MS 39327

ASHLY AUDIO
100 Fernwood Ave.
Rochester, NY 14621

BGW SYSTEMS, INC.
13130 S. Yukon Ave.
Hawthorne, CA 90250

BIAMP SYSTEMS
PO Box 2160
Portland, OR 97208

BRYSTON VERMONT, LTD.
RFD 4
Berlin, Montpelier, VT 05602

CARVER CORPORATION
PO Box 1237
Lynwood, WA 98046

CARVIN MFG. CORP.
1155 Industrial Ave.
Escondido, CA 92025

CERWIN-VEGA!
12250 Montague St.
Arleta, CA 91331

CREST AUDIO
150 Florence Ave.
Hawthorne, NJ 07506

CROWN INTERNATIONAL
1718 W. Mishawaka Rd.
Elkhart, IN 46517

FOSTEX CORP. OF AMERICA
15431 Blackburn Ave.
Norwalk, CA 90650

THE DAVID HAFLER CO.
5910 Crescent Blvd.
Pennsauken, NJ 08107

HILL AUDIO, INC.
5002B N. Royal Atlanta Dr.
Tucker, GA 30084

JBL/UREI, INC.
8500 Balboa Blvd.
Northridge, CA 91329

PANASONIC INDUSTRIAL CO.
Pro Audio Systems
1 Panasonic Way
Secaucus, NJ 07094

PEAVEY ELECTRONICS
711 A Street
Meridian, MS 39301

QSC AUDIO PRODUCTS, INC.
1926 Placentia Ave.
Costa Mesa, CA 92627

RAMSA
see Panasonic

RANE CORPORATION
6510 216th SW
Mountlake Terrace, WA 98043

SOUNDCRAFT, INC.
1517 20th St.
Santa Monica, CA 90414

SOUNDCRAFTSMEN
2200 So. Ritchey
Santa Ana, CA 92705

SUNN/FENDER
1130 Columbia St.
Brea, CA 92621

UREI
see JBL

YAMAHA INTERNATIONAL CORP.
Combo Division
6600 Orangethorpe Ave.
Buena Park, CA 90622 ■

2 to 8 trk

Larry Oppenheimer

Serafine FX: MIDI in the Movies

MIDI in the movies. Frank Serafine is in the forefront of sound design using MIDI linked synthesizers. Read on for how he uses MIDI and SMPTE.

FRANK SERAFINE is out in space, and he loves it. From robots to photon torpedoes, Serafine has filled science fiction movies of the 80s with his sounds. I'm sure you've heard the credits: *Star Trek 1 and 3, Tron, The Day After, Brainstorm,* and so on. The music and effects that Serafine contributed to these films were created at Serafine FX, his relatively earthbound Santa Monica, CA, studio, by a virtual arsenal of musical instruments and recording tools that epitomize space age technology. One might expect that a man this involved with electronic music might find MIDI a boon to his work. And so he does, employing it extensively in all of his projects.

MIDI's most evident uses to this point have been layering sounds and sequencing. Serafine does a great deal of both of these, often combining his instruments into a synchronized network that produces a thickly-layered texture.

Although Serafine uses sequencing in his work, it is not always the most effecient method. One recently added aid to performing non-sequenced cues is the IVL Technologies Pitchrider. "If we're channelizing synthesizers for a particular sound effect," explains Serafine, "what we've been doing is taking sounds off tape, running them through the IVL pitch-to-MIDI converter, and then having those sounds trigger sounds on the synthesizers." Serafine also uses the Pitchrider to control synthesizers from his flute. (A flute is an acoustic instrument. Remember them?)

Considering the nature of his work, it comes as no surprise that sampling is one of his main tools, and the E-mu Systems Emulator II is one of his main axes. The E II's ability to synchronize its sequencer to tape, MIDI, or SMPTE time code makes it invaluable in film work. Serafine makes field recording trips to capture his own source material onto a Sony PCM-F1-based digital recording system, then dumps them into the E II upon return to his studio. Getting the sounds from the E II to the appropriate place in the soundtrack may be done with synchronized sequencing, or, again, performed.

When sequencing effects or sound for film, synchronization to the visual is a primary consideration. Before MIDI, such synchronization was approximated by laying down a track of some sort of sync clock onto tape, then recovering it and clocking a sequencer with it. This did not achieve ideal synchronization because it provided only a clock signal at a given rate for the synthesizers, with no relative reference or demarcation of elapsed time. This characteristic dictated that sync-to-tape tracks must always be started from the beginning; it is impossible to pick up in the middle of a sequence.

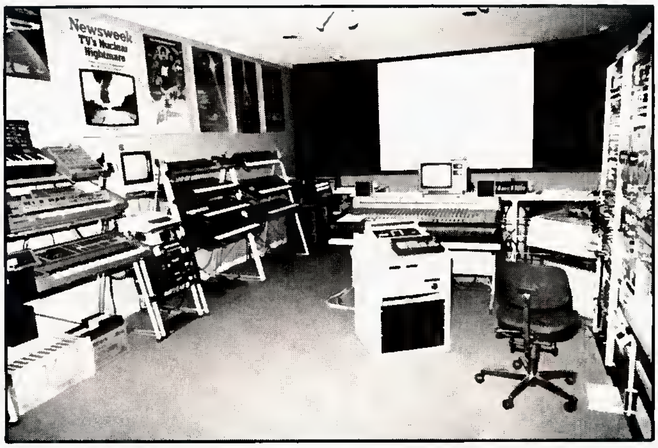

Figure 1. An overview of Serafine FX. Synths on the left, projection screen behind the console, and tape recorders on the right.

In the last few years, SMPTE time code has come into increasing usage as amaster time reference for synchronization in film, video, and audio studios. As MIDI's ability to synchronize a number of instruments became visible it became clear that linking MIDI and SMPTE time code was a powerful new production technique for synchronizing synthesized music and effects to pictures.

Typically, Serafine will compose from a work print of the film on a videocassette. One of the cassette's audio tracks contains SMPTE time code, which is used in some critical situations, while performing effects directly is chosen as a quicker and easier way with simple cues. "I'm pretty good at performing stuff right to the picture," says Serafine. "Let's say it's a sound effect we're doing, instead of locking up all the machines and trying to do electronic edits. I'll just go over to the Emulator and say, 'OK, where's that sound supposed to go?" 'Then I'll just play it and drop it in, either directly to the multitrack, or into the Emulator's sequencer while it is locked up to SMPTE time code.

This latter technique is often used when Serafine is trying to place a sound, such as a ray gun blast, that must correspond very tightly to the picture. In this circumstance, the tape will be slowed down in playback so that it is easier to find the precise time code at the cue point. This time code is then entered into the E II as the start time for the sequencer, which plays the desired sound. When played back at full speed, the effect will be perfectly placed within the resolution of the time code.

Larry Oppenheimer is a consultant and musician whose San Francisco firm, Toys In The Attic, specializes in computer music and digital audio.

The Emulator II is only one instrument of many at Serafine FX, however. For applications that call for more than one instrument Serafine employs a software sequencer running on his Apple IIe. Although he uses several software packages, at this time he tends to lean towards the newest program from Passport Designs, Master Tracks. Unfortunately, Serafine has not found any software with onboard SMPTE capability, which makes it more difficult for him to SMPTE sync these programs in any straightforward fashion. He tried SMPTE-to-MIDI interface boxes, such as Roland's SBX-80 for a while, but Serafine found these devices "not as responsive as we'd really like them to be."

As a result, a more roundabout method is currently used. In this scenario, Serafine records a sync clock onto the other audio track of the video tape (dumping the dialogue, which is unnecesary at this point) and then uses sync-to-tape to drive the software sequencer. Although this technique sacrifices all the advantages of time code, particularly the ability to pick up in the middle of the track, the use of a software sequencer simplifies matters when there are a great number of cues and edits to be performed.

The older synchronization methods still finds use, even in a MIDI studio, but that does not mean that using older equipment in this Age of MIDI is not a pain. A number of Serafine's instruments did not come with MIDI, but it was added later when the demand became great. Unfortunately, since these instruments were not designed with MIDI in mind, the implementation and performance of MIDI retrofits is typically more limited than on new equipment. "You have to keep adding little things like the Roland MPU-103 for filtering channels and channelizing," sighs Serafine. "For example, the Emulator I receives MIDI data in Omni mode only, so it has to have a channel filter (to simulate Poly mode operation). The Drumulator retrofit, though, can be channelized."

Despite this strength, the Drumulator retrofit is hindered by other implementation shortcomings. It can only receive MIDI note information, not drive its sequencer from the MIDI timing clock. (It should be noted that the Emulator I and Drumulator are no longer made, and their replacements, the Emulator II and Emulator SP-12, have excellent MIDI implementations.) Serafine moves around this block by playing the Drumulator from the E II's keyboard. This turns out to have several benefits. One is that the Drumulator retrofit does respond to MIDI velocity information, so the dynamic playing on the E II keyboard will impart dynamics to the drums. More importantly, since the E II's sequencer can be synced to MIDI, drum parts can be played into the sequencer from the keyboard and then transmitted to the Drumulator for playback.

But being forced to deal with MIDI's limitations during a major production can be vexing. "It's been a headache just trying to figure it all out," Serafine admits. Overall, however, the benefits have far outweighed the drawbacks. In fact, he has been fortunate enough to have no encounter with some of the other problems that have been attributed to MIDI. Perhaps the complaint most often heard about MIDI concerns speed problems that arise when using a number of instruments at once in a MIDI system, but this problem has yet to plague Serafine.

"I've always had all my synths going at the same time. I may have two synthesizers playing strings and two synthesizers doubling on a bass sound. I may only use eight tracks, but all the synthesizers are going at the same time. Throughput delay problems, usually caused by daisy-chaining, MIDI instruments, have been very simply solved by not daisy-chaining at all, using MIDI thru boxes instead to provide the necessary multiple outputs.

Even with the considerations discussed here, Serafine never really had any doubt about committing to MIDI, even when it was new. "You can control so much more with MIDI than with control voltage systems. Control voltage never really turned me on except for what you could do with sequencing. It's amazing that you can control all these keyboards so much easier than you could with control voltage systems. You had to have pretty complex patching to do control voltage stuff. It was a pain because you didn't have just one cable, you had four cables—control voltage in and out, and trigger in and out—so it was a lot more complex figuring things out when you wanted to patch. Also, you couldn't go into a computer very easily. I saw things coming up over the horizon, so I just said, 'Well, MIDI'll be the way to go, so we'll just wait until we can move into that.'"

Figure 2. In this view across the console, you can see the tape recorders and equipment racks. Note the DEC computer in the right rack.

Although bothered at first with difficulties like trying to use multipleperformers with MIDI, solutions like MIDI merge boxes are appearing to give relief and open new possibilities. And as new possibilities keep coming, Frank Serafine is ready to explore them. "I want to look into controlling my signal processors through MIDI. I make extensive use of the Eventide SP-2016; if a MIDI retrofit becomes available for it, I'll be getting into that." (A MIDI version of the SP-2016, retrofitable, is promised from Eventide.)

Serafine is using his MIDI setup for a number of current projects, including more sci-fi fare such as the feature films *Navigator*, *Short Circuit*, and *Poltergeist 2*. Perhaps his most interesting application of MIDI technology will come in another collaboration with Disney Productions, for whom he worked on *Tron*. A multimedia laser show is being produced for the lake at Disney's Epcot Center in Florida, and Serafine FX is providing music with MIDI. All music will be sequenced and performed at Serafine FX with a MIDI synthesizer system, then laid onto a 24-track tape, which will then be synchronized to the laser performance.

Certainly Frank Serafine is in the thick of it. For better or worse, he is committed to MIDI as a primary tool in his everyday work. There are always detractors who point out the weak spots in the MIDI specification, but Serafine feels that some of these criticisms may be exaggerated. His view of MIDI is one of a creative panorama stretching to the horizon, where the next incredible advance lurks just out of sight. "There aren't a whole lot of limitations," he concludes about his current direction. "It's pretty good stuff."

Serafine FX: Instrument Inventory

SYNTHESIZER AND SAMPLING INSTRUMENTS

E-mu Systems
Drumulator (with MIDI retrofit)
Emulator II
Emulator I (with MIDI retrofit)

Minimoog (with MIDI retrofit)

Yamaha
(2) DX-7
(2) DX-7

Sequential
Prophet 5, Rev. 2./(with MIDI retrofit)

Korg
DW-8000

Casio
CZ-5000

Alpha Syntauri
computer music system

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IVL Technologies Pitchrider

Software

Passport Designs
Master Tracks
Polywriter

Yamaha
DX Pro

Cherry Lane
Texture
DX Heaven
DX Librarian

Syntech
Music Digital

■

Larry Oppenheimer

Making Sense Out of MIDI

Here you'll find out all you will ever need to know about MIDI.

IS THERE anybody out there who has not heard about MIDI? I didn't think so. But events have moved so fast since MIDI appeared on the horizon that it has become easy to lose perspective and be overwhelmed by it all. Perhaps it's time to take a step back and consider this amazing phenomenon.

PRE-MIDI INTERFACING

Before the rise of MIDI, making two electronic musical instruments talk to each other was a painful and chancy process. Well, it still can be, but nonetheless, things have progressed far since some unfortunate synthesist decided for the first time that it would be fun to hook his Moog modular system up to his ARP or Buchla system. In the final analysis, though, the nature of the problem and the range of possibilities are all that have really changed.

In the old days, connecting two synthesizers together meant a separate patch cord for each signal to be transmitted from one to the other, and often some circuitry (usually homebrew or custom-built) to transform a signal output from one instrument into a form usable by the input of the other instrument. This presumes that the desired output even existed, and that the formats used by the two instruments were known to the user. Until the proliferation of ARP instruments (which used a system of 1V/octave control voltage response, ± 10V signal amplitude, and gate-and-trigger timing signals) nudged the market towards that format as a pseudo-standard, there existed no rules for synthesizer manufacturers to follow when designing the characteristics of their instruments.

As a consequence, musical parts involving more than one synthesizer were most often done by simply performing the parts on each instrument individually; in live performance this meant playing a different instrument with each hand, in the studio it meant massive overdubs.

MICROPROCESSORS ENTER THE PICTURE

In 1977, the Prophet 5 synthesizer was released by a small

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San Jose,California-based company called Sequential Circuits. The instrument had several things that made it noteworthy. First, it was polyphonic. Polyphonic syntheisizers had only entered the scene in 1975, with the introduction of the Polymoog. Second, it had a great sound: fat, thick, full, whatever you want to call it, but certainly distinctive. Third, it was programmable. Programmability meant that the hours spent working up the "perfect" sound were preserved, even if not laboriously transcribed to paper. It also meant that a number of "perfect" sounds were instantly obtainable at the touch of a button. In those days, this was really hot stuff. (It still is, but we often take such things for granted.)

The programmability was achieved by designing the Prophet 5 around one of the first of the exciting new generation of electronics: microprocessor chips. Microprocessors contained the guts of the most critical part of a computer, the Central Processing Unit (CPU), on one tiny silicon wafer enclosed in a plastic package about the size of a New York City cockroach (that's about two inches long by one inch wide, for those of you not familiar with urban fauna). The important point here was that the microprocessor did not deal at all with the sound itself, but with information about the sound: oscillator tuning, VCF and VCA cutoffs, keyboard note depressions, etc.

Although the Prophet 5 was not actually the first synthesizer with a microprocessor (that honor having gone to a couple of guys in Santa Cruz,California, who called themselves E-mu Systems, and their 4060 microprocessor-based keyboard), it proved to be the first portable, polyphonic, programmable synthesis instrument to be widely available. Soon, Prophets were everywhere (in those Biblical days of music synthesis), and other programmable machines quickly appeared. Commercial computer music was underway.

In spite of the power of the microcomputers inside these new machines to process information, no one concerned themselves very much at that time with directing any of that power towards communication with the "outside world." Hence, these instruments were not much easier to interface than the older ones.

As programmable synthesizers became more sophisticated, some designers began to realize the possibilities of intimate communication between different computer-controlled instruments. Computer interfaces began to appear on the back of several manufacturer's instruments (most notably Oberheim, Roland, and, to a lesser degree, Yamaha), to allow them to talk with other instruments made by the same manufacturer; and sync clock outputs aided interconnection of sequencers and drum machines. Still, different brands couldn't talk directly to each other.

Not only that, but the prospect of trying to transform one kind of computer interface to another was so much worse than transforming control voltages, gates and triggers or even sync clocks as to be impractical. In fact, the information on these interfaces was not even available from most manufacturers that used them. (Indeed, the appearance of this data in my upcoming book, *Connections: The Handbook to Interfacing Musical Instruments* will probably be the first time this now-obsolete information has ever been published).

PRESENT AT THE CREATION

At Sequential, David Smith was pondering the absurdity of hundreds ofinstruments with such large information-processing capabilities sitting next to each other, yet talking only to themselves or a small group of friends. As he analyzed the problem, ideas began to form. He consulted with others in the industry, particularly several Japanese and American manufacturers, and presented his ideas for a "Universal Synthesizer Interface" to the Audio Engineering Society at a convention in 1981. Two years and many conversations and debates later, MIDI was announced jointly by Sequential, Roland, Yamaha, Korg and Kawai. Some manufacturers, such as Oberheim, dissented on the grounds that the proposed "standard" was flawed, particularly in terms of its speed. But others jumped in, and, when it became obvious that the tide was indeed turning towards MIDI, Oberheim and the others threw themselves into it wholeheartedly.

WHAT IS MIDI ANYWAY?

Just in case you haven't seen any of the slew of literature explaining MIDI, let's give a short definition of it. If you need more detail, there are many excellent articles and books, such as Craig Anderton's "MIDI for Musicians" to refer to.

MIDI stands for Musical Instrument Digital Interface, and it is a specification that describes a set of codes that comprise a sort of computer language for transmitting information about musical performances; plus a simple hardware circuit (consisting primarily of an optoisolator, a current loop, and a 5-pin DIN connector) which is designed to help prevent ground loops and other electrical problems from interfering with transmissions. There is even a paragraph specifying the construction and maximum allowable length of MIDI cables. That said, "Let's look a little closer at what is defined in the spec."

The MIDI specification (currently in version 1.0) dictates that information will be transmitted serially (one bit at a time over a single wire) in 8-bit bytes with one start bit and one stop bit (which indicates the beginning and end, respectively, of a data byte) at a rate of 31.25 kilobaud (baud equals "bits per second"), which means that one byte of MIDI data takes about 320 microseconds to transmit. MIDI data is transmitted, with only a few exceptions, in groups of two or three bytes that combine to make a "message." This means that a MIDI message will often take a millisecond to transmit. The spec also states that there will be sixteen "channels" over which MIDI data can be sent.

There are two basic kinds of messages in MIDI: Channel and System. Thesefunction just as their names indicate. Channel messages apply only to thespecific MIDI channel named in the message, while System messages address all channels. Channel messages are the most common, and these give information on whether an instrument should send or receive on one channel or all, beginnings and endings of note events, and control information like velocity, program change, aftertouch, pitch wheel, etc.

System messages give timing information, like MIDI's timing clock, the current bar of the song, and start/stop, plus song select information and several "housekeeping" messages. One of the more interesting provisions of the MIDI specification is a "trap door" feature called "System Exclusive," which allows manufacturers to transmit any information not otherwise defined in the spec. System Exclusive information is now used to transmit everything from a synthesizer's patch parameters to a sampler's actual

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sound data. (*Editor's Note: See Jesse Klapholz' "Computer Audio" column in this issue for more MIDI information.*)

WHAT TO DO WITH MIDI

Oh boy, here we go. Although it has been nearly three years since MIDI'sinception, the range of applications is just now beginning to become clear, I think! Maybe next year someone will come up with things we haven't thought of at all. Let me explain.

When MIDI was originally conceived, it was intended to provide an easy way to use multiple pieces of equipment, from different manufacturers or the same one, in simultaneity. In other words, play several synthesizers from one keyboard, or use a drum machine of one made with a synthesizer of another. This has more or less happened, although there are still a few kinks, as we'll see shortly. More significantly, creative dreamers began to come up with more sophisticated and grandiose applications.

Arguably, the most important of these was the quite logical idea of hooking a personal computer into MIDI through the use of a hardware interface. Although stand-alone MIDI sequencers are quite capable of driving a number of instruments, a personal computer can offer more flexibility and a better display, in addition to being useful for other musical and non-musical tasks. Using a personal computer as a master for a number of slave instruments, an entire orchestra can literally be put under the control of one person. The ever-expanding plethora of available instruments yields an almost overwhelming pallette of sounds with which to work and orchestrate. Best of all, using sophisticated sequencing software compositions can be quickly and easily built, and edited with the results being instantly available for auditioning. For centuries, composers of large-scale works have been lucky if they ever heard their music performed, because of the staggeringly high price of paying an orchestra full of musicians. This made detailed reworking problematic. Now they can hear their compositions whenever and as often as they like, making changes at any time.

Two other benefits of using personal computers with MIDI are voicing and library software, both achieved through the use of System Exclusive information. Voicing software, which allows programming of a synthesizer's parameters from the computer, began appearing first for the Yamaha DX series of FM synthesizers. A personal computer monitor can show the DX's parameters more completely and in easier-to-see graphic representations than the instrument's own small display. Library programs can store many user presets on a disk and allow them to be loaded back into that or any identical instrument.

Another important application of MIDI arose as an extension on the idea of playing several instruments from one controller. Rather than trying to makeseveral sounds at once by simply stacking a few instruments, each instrument is set up to make one component of a single desired sound, then combined with others to create a composite with more realism than any one can get alone. Yamaha's TX-816, a rack containing the equivalent of eight DX-7s, is perfectly suited to this: use one module to make the hammer attack of a piano, one for each string of a given note, one for the string transients, or one for the sustain, etc.

Soon after the appearance of MIDI, peripherals of various sorts started showing up: MIDI switchers and routers, sync boxes, even MIDI event processors. These have all created new applications. Now it is possible to score an entire soundtrack with MIDI control and perfect synchronization by simply receiving a video work print of the film which is striped with SMPTE time code. The time code is fed into a SMPTE-to-MIDI box, which then clocks a PC software sequencer that acts as master to a number of instruments. In fact, the rise of sampling instruments even makes it possible to use this same system for a great deal of sound effects and Foley work. Furthermore, there is the matter of MIDI-controllable signal processors. Not only is it possible to buy a number of digital reverbs and delays that can recall presets from MIDI Program Change commands, but Lexicon's new PCM-70 will actually let you vary parameters of reverb or effects programs in real-time, using MIDI controllers. Now your reverb and effects can be added to your automated production facility.

Recently, attention has been turned to MIDI controllers, such as pitch-to-MIDI converters (Fairlight Voicetracker and IVL Technologies Pitchrider) and new controllers like the Synthaxe string controller, Roland Octopads, and Kat Expandable Percussion Controller. MIDI data processors are starting to get interesting, too, with Yamaha's MEP-4 and Axxess Unlimited's Mapper leading the way. My feeling is that MIDI presents so many possibilities that it will become obsolete long before all the possible applications are exploited.

WHAT'S WRONG WITH MIDI?

MIDI is indeed a flawed specification. There are a number of reasons for this, a great deal of them related to the difficulty of reaching an agreement between several manufacturers on an issue. To keep things in perspective, however, let us remember that MIDI is the first time, to my knowledge, that musical instrument manufacturers have agreed on anything of any consequence. So, in any case, it already represents a mighty step forward. After all, there may be problems with MIDI, but we should give credit where credit is due.

What are a few of these problems? Most of them have something to do either with speed or incompatibility. The biggest speed problem arises directly from the MIDI spec. If it takes about one millisecond to send a MIDI message for one note, you can imagine how long it takes to transmit many notes to an array of polyphonic synthesizers hooked up to a personal computer, plus any timing or status information that needs to be sent. A ten or fifteen millisecond delay between instruments sounding notes is quite audible.

Another common source of speed problems is the response time of an instrument to MIDI information. Some of the early MIDI instruments were retrofitted versions of existing designs. Adding MIDI functions put more strain on the instrument's microprocessor, causing delays to occur between the time MIDI information was received and the time a note is actually sounded. Even a sufficiently powerful system might have delay if the system's operating software is not well written. In the worst case, this delay could be as long as twenty-four milliseconds! Obviously, a network of different instruments having different delays gets nightmarish. Furthermore, there are often delays (called "throughput time"), introduced between data appearing at an instrument's MIDI In port, and going out its MIDI Thru port, which is supposed to carry an unaltered copy of MIDI In. If a number of instruments are "daisy-chained" (connected in series, MIDI Thru to MIDI In), the result is a cumulative delay of an unpredictable length. Aarrgh!

The solutions have not been straightforward. In the case of MIDI's baud rate, there's not a whole lot than can be done.

The MIDI spec does have a provision, called "running status," which allows elimination of redundant status bytes from a string of MIDI data. This can make a significant difference, but sometimes even this is not enough. The result is that parts are still frequently multi-tracked instead of laid down with all the instruments in one pass. Since MIDI's earliest days there has been discussion of creating some "souped-up" version of MIDI at a faster baud rate, and Sequential has recently shown several products which have a "Turbo MIDI" mode that operates at twice the normal baud rate. As of this writing, though, there are no other manufacturers making instruments compatible with this scheme.

For processor delays there are several techniques in use. The most common solution to throughput delays is to use "MIDI Thru" boxes which act as "MIDI splitters" and provide a number of simultaneous outputs without delay. Thus, daisy-chaining is avoided. Note that delays are even more difficult to deal with. Since it is impossible at this time to move the offending unit's output forward in time (at least until someone announces a digital anticipation line that makes a sound before you do), the solution in the studio is to move everything else back in time in a fashion similar to that used to deal with sync clock processing delays. The slowest instrument's delay is found and used as the maximum value of delay for a master reference signal on tape, usually a clock track or SMPTE time code, and digital delays (or offsets in the case of SMPTE) are individually tuned for all the other instruments until everything sounds together. Manufacturers are working to eliminate processor delays in their products, and the advent of electronics like Yamaha's MIDI controller chip and inexpensive single-chip microcomputers indicate that soon MIDI communications will be tended by a separate processor, lifting the load from the instrument's main processor.

Still of concern, however, is incompatibility between different manufacturers' MIDI implementations. The MIDI spec leaves some points of implementation somewhat open to interpretation or choice, and this leads to problems, e.g., differences in "note off," message implementations could lead to notes being "stuck" on. Only the manufacturers are in a position to solve this problem, but careful documentation of experiences with MIDI instruments can help you identify and keep track of specific problems.

Other problems with MIDI have been more easily solved. For example, the problem of constant repatching of cables to try different configurations has been virtually eliminated by the appearance of various MIDI switching and routing boxes. These useful accessories can range from about sixty dollars on up, depending on how much flexibility you need.

THE IMPACT OF MIDI

MIDI has definitely revolutionized the audio and music industries. Many manufacturers have seen rejuvenated sales and new ideas for products since MIDI's inception. In fact, entire new sectors of the industry have appeared as MIDI software, interfacing devices, and peripherals have blossomed. A great deal of these new kinds of products are coming from small and "cottage industry" manufacturers.

Furthermore, MIDI has also altered the long-standing tradition of everysynthesizer having a keyboard (or occasionally some other controller) attached. In the early days of analog synthesizers, Robert Moog was hesitant about building an organ-tupe keyboard because he was afraid that people would think it was important to the instrument. Of course, Dr. Moog was right, and for years people have foolishly considered synthesizers to be keyboard instruments. MIDI is slowly taking us back to a more accurate view of synthesizers as sound producing instruments which can be controlled in any number of ways. New controllers are appearing all the time (a MIDI harmonica? Yes, that has been tried, too) and synthesizers are starting to appear in rack-mount boxes with no controller attached. The separate controller/synthesizer approach is finally leading musicians to some of the uninhibited and fabulously expressive contexts that have often been subjugated to the "super organ" viewpoint dominant in the past.

But MIDI has most certainly affected musicians as much as manufacturers. Live performance has changed with the ability to do extensive live sequencing and easy layering of several machines. Parts can be worked out at home on a PC-based sequencing program, then dumped into a less fragile stand-alone sequencer for roadwork. Conversely, music written on the road with the stand-alone sequencer can be uploaded to a PC at home for more detailed editing.

Studio recording has gone through even more extensive transformations. With MIDI, a musician can do virtually all pre-production for a demo or record at home where it is cheap, and spend a minimal amount of time in expensive studios. For example, a musician might have a stand-alone sequencer and several inexpensive MIDI synthesizers (such as the Casio CZ-101). All lines and parts can be worked out at the musician's leisure, whenever inspiration strikes. A few days before going into the studio, the musician rents the keyboards that are desired for the actual demo/record and programs the sounds to be used with the already composed parts. The sounds, of course, can be immediately heard playing their lines and tweaked to perfection in context without having to play and program at the same time.

When the sound programs are complete, they may also be dumped into a PC, running a librarian program. Finally, the musician goes into the studio with all his computer disks and software and simply dumps the sounds into the waiting instruments, then runs the sequence with the tape recorder going, and voila! The tracks are perfectly transferred onto tape. A rough mix can be laid back onto a tape supplied by the musician, and the process may then be continued at home again to work out overdubs. Notice that the musician need merely carry computer disks and have availability to the machines to do this. You can do some work in New York, some in Los Angeles, some in Punxsatawney, and some in Plainview, without ever transporting anything besides a few computer disks. The entire relationship between personal studios and professional studios has been altered. There are even new facilities springing up that are exclusively MIDI synthesizer studios. As with any new technology, there are also a host of MIDI consultants that have arisen to advise and solve problems.

CONCLUSIONS

MIDI has only been around for three years or so. In some ways, it really shows its immaturity in the problems that are still rampant. In other ways it is nothing less than astounding to see how many musical dreams have become realities in so short a time. There are no signs of this madness flagging in the next few years. Although something will eventually come along to render it obsolete, MIDI is something we have now and which is changing our whole musical universe.

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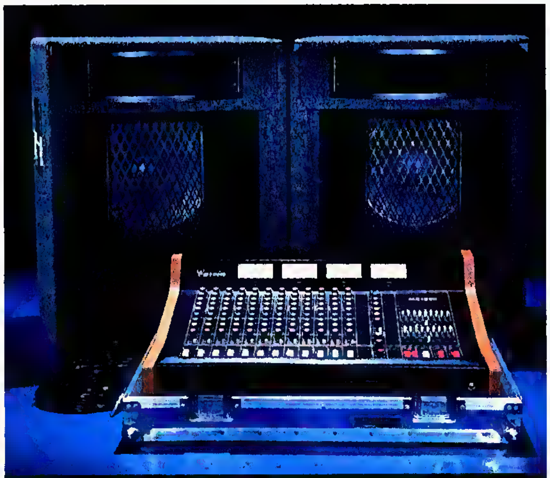

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Marshall King

The Modus Operandi Of David Kurtz–Part II

Tune in to learn about the new trend in film scoring as seen by composer David Kurtz.

THEY SAY if you want to get a difficult job done right, give it to a busy person. Why this is so is anybody's guess, but there was a time when busy composer David Kurtz never thought that would apply to him. Those not-too-distant weeks and months of taking odd jobs around Hollywood just to survive didn't point to much hope for a busy future composing music for television and films. But, with a generous amount of belt-tightening, a lot of whistling in the dark, and taking whatever jobs in music that popped up occasionally...gopher, proofreader, copyist...things began to open up. Today, working mainly from his studio at home, David understands about being busy. Having just come off a long stint with daytime drama on CBS, his current assignments include writing for the TV series *Great Scott!*. Other recent television credits are *A New Day In Eden* (Showtime) and *Fantasies* (PBS, New York).

Last month we asked David to give us a closer look at those "door-knocking" days when he was trying to get a foothold in the Hollywood music scene. "Protocol is a key word when you're coming in out of the rain, " David told us, as well as such considerations as budget, patience, persistance, and humor. He reminds us that, when it all starts to come together, any natural talent in music has to have a friendly relationship with technical know-how.

After his in-depth look at the usual way of scoring music to film, with particular emphasis on the composer's relationship with the film editor, David now turns his attention to a new trend in film scoring: working at home from a video cassette where the SMPTE code recorded on the edge of the tape takes the place of the sprocket holes on film, thus facilitating the same accurate timing a composer needs in fitting his music to the picture. This led me to other questions:

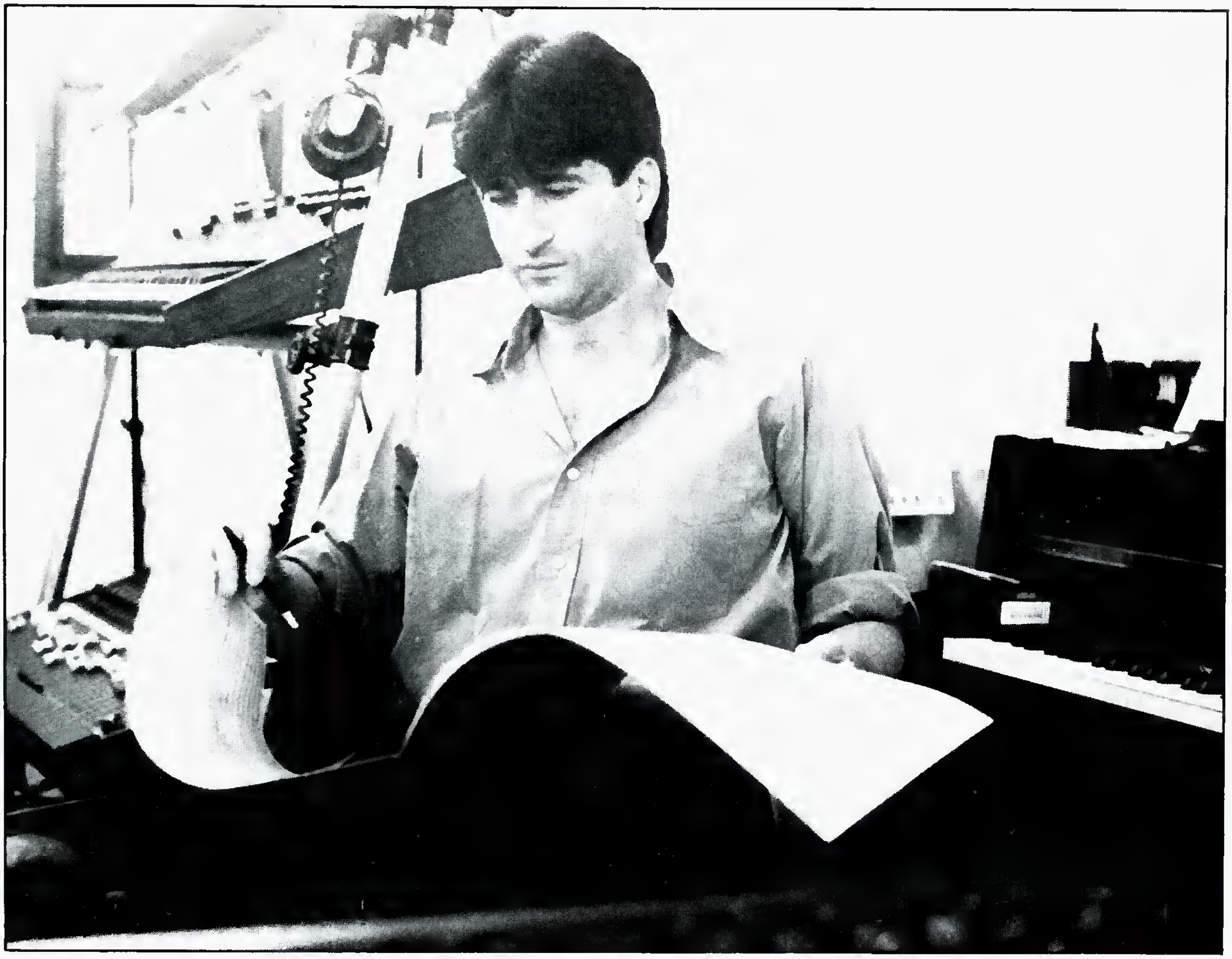
db: Is it because you can do more on your own equipment that you work mainly at home?

DK: Yes, that's the position I'm in right now. I finally got enough money together to start slowly building my own studio, for the excitement of electronic music never left me; it's a whole new world of sound. My goal is not to replace traditional music and traditional musicians with electronic sound, but to use it on its own musical terms for its own musical qualities. I'm not looking to do traditional music when I go into my studio; I'm looking to develop new sounds that excite me as an artist.

I mention these things because there is an understandable pressure on composers from labor groups who fear that electronic music will replace many or most traditional musicians. This is unfortunate, for it could conceivably drive serious conventionally performed musical experimenters to Europe or Japan or wherever, and the country certainly doesn't need that kind of loss...its prominence in contemporary music. Not that things will ever get that far, but I think it needs to be recognized that you can't legislate restrictions on new forms of artistic expression.

Traditional instruments and their players will always be needed for the human qualities that a synthesizer can never produce. If there are indeed composers out there using synthesizers to replace traditional instruments rather than augment them, this is unfortunate, but I think that movement will die of its own transparency, for who can't see through the difference between a synthesized violin and the real thing? It's musically unsatisfying, and for those who can't tell the difference...this may speak more for the listener's shortcomings than any composer's evil intentions.

This is not to say that there aren't electronic designers out there striving to emulate some of the parameters of traditional instruments, (thus, we have samplers and breath controllers), but whether these are to be applied in achieving a synthetic violin or a musical door slam has never been specified. I use my synthesizer studio purely for its own sonic capabilities, and more often than not these are incorporated with traditional instruments, for I get quickly bored with using only synthesized sounds. This is my own personal prejudice. They are inhuman, even with all the tricks applied, and they are hard to get what I call musical gestures out of. They cannot duplicate those imperfect



variables that create a bond between the listener and the performer; they lack the expressivity that you can so easily get with a traditional instrument in the hands of a fine player. Even the very attitude of a player comes through his instrument. You can never duplicate that in computerized music. My joy is not to use electronic equipment to do what traditional instruments can do, but to do what they can't.

I happen to believe that we are in a very "sound productive" society right now, a sound-influenced society, and when I listen to the radio and hear what's current, it occurs to me that it's not so much the music itself we listen to, but it's the production of it that captivates. I find that most of today's music is musically uninteresting but fascinating soundwise. The production tehniques are so incredible now that we've become a very quality-conscious society. For example, people are laying out three times what records cost for compact disks. The exciting thing for me is that that's where TV and film are heading. The Lucas film THX process where the sound comes out and surrounds you and grows and overwhelms you, that's the kind of thing that's going to save theaters. Stereo television is on its way now, too. This is why the time is right for me to take whatever sonic notes I can create, combine them with traditional instruments and come up with sound colors that are entirely new.

db: Perhaps because of our age difference I can remember when anything other than traditional sounds, such as saxophone or trumpets, strings or piano, was considered a novelty, an intruder not to be taken seriously. Today, however, the changes both in musical style and musical technology are coming faster and faster. Obsolescence is in the air. Does this bother you?

DK: No, and it's interesting what you said about age difference, because it points to an almost unspoken facet of our new musical attitude. I'm twenty-eight now, and I believe my generaton is the first to identify its music only as it sounds after having been electronically manipulated rather than "live." For example, I am very well versed in concert music, but I didn't learn it from live concerts, I learned it from recordings. The point of this is that, for me and probably most of my generation, going to live concerts is somewhat of a let-down sonically, particularly if acoustic instruments are involved. My musical beliefs were formed from hearing things from speakers within arm's length, controlled hearing, if you please. I suspect that people today would rather listen to a recording on a really good system, on a compact disk, than to hear it done live.

db: Then, what about the throngs that flock to live concerts?

DK: I suspect they go for the pageantry, not the music. They go in order to be part of an event. When the Beatles went to Shea Stadium or the Hollywood Bowl, fans didn't go there just to hear their music...because you couldn't hear it. Just listen to the tapes of those old concerts! All you heard was screaming and the kind of audience participation that comes from being part of an event. The audience already knew the music before they got there, and they learned it from a controlled, amplified presentation such as radio or records, so hearing this familiar music being done "live" was not so much a musical treat as it was a participation. In fact, since even the live concert is amplified, we might say that people today simply have not heard their music done "live" as they think they have; from the stage they are hearing an electronic reproduction of even the voices. The only thing "live" about it is that the performers are actually standing on stage, but they are delivering sound that are electronically created, shaped, and enhanced. We must remember that at one time "live" referred to acoustic sounds coming unaided either from the instrument or the throat, as used to be common in chambers, theaters, and concert halls.

So, I'm saying that times have changed to the point where we come to know our music not from live performances, but from the controlled electronic environment of radio and disc players. Thus, the production of the music has come to be its identifying factor rather than the more academic qualities of instrument identification, voice attributes, harmony and orchestration. The music I listen to and enjoy the most does not come from these "live" presentations, but from fine work in the studio and control room.

Why do you think the Beatles stopped touring? I think it's because they found that the interesting things they could do in the studio couldn't be done on the road with the results they wanted, so they withdrew further into the recording process. I think that some time after their *Rubber Soul* or *Revolver* albums, they were doing incredibly unique things; you never knew what you were going to get on a Beatles album. I mean, when you go out now and buy a heavy metal record you know what you're getting. Not so with the Beatles; you never knew! Things played backward, tape collages...you didn't know whether to expect Paul on the guitar by himself or a double chorus with string quartet. In short, the production of the music has come to be the strived-for element that is written into the work for the listener to hear, perhaps overshadowing the more traditional parameters of lyric, melody and rhythm.

db: When given a picture assignment, do you write only electronic music?

DK: Not at all. I write to the picture. I try never to view a picture with any preconceived ideas about the music. That's why I don't like to read scripts; I'd rather see the film and get my first impression from that, then apply whatever style or mood the story seems to demand. I've spent a lot of time working toward musical versatility, to the point where I feel comfortable working in any style, in any genre. It's like adding one more valuable tool in your workshop.

I won't lie about it though; my preferences, my pritorities when I have a choice, lie in what I can accomplish here in my own studio—music that I can combine with traditional instruments. Then there are a lot of colors to choose from.

db: Even though your studio is electronically complete, it is rather small. Can you accomodate live musicians here?

DK: No, I have to go outside for that, and I want to. Which brings me back to the point about hibernating. Earlier I said you have to go out there and be visible, and yet, I've worked hard to get my own studio where I can work in private. This seems to be a contradiction, but I solved the danger of hibernation by finding a good agent who works very hard at keeping me visible, keeping my work on the marketplace. I can lock myself in here for quite a while until a project is finished, but once it's over I like to get out. In fact, I must get out. I even like to take my tapes out of here and mix somewhere else, just to get into a different space, to be among other people, to hear what's going on, to learn about the latest equipment, and generally just to stay in touch.

db: Unlike other music such as records or opera or concerts, film music is traditionally not a thing in itself...it's part of a visual work. Do you think that's changing?

I think so. Traditionally, film music has always been in the background, thus, we have the term "underscore." We were taught that the viewer was supposed to hear the music but not be aware of it. Themes, of course, are different, just as are production numbers, but I'm talking about basic underscoring.

What's happening now though, is that film music is really coming to the forefront, it's being used more for the effective tool that it is. MTV is the exaggeration of this, of course, where the picture is the underscore for the music, but you can see what I mean. This trend is infiltrating both films and television. Perhaps the best current example of this is *Miami Vice*, where the music is almost a part of the dialogue. But here you have a producer who understands this and who apparently intends to make the most of music in dramatic film.

Here's another example of what I mean. Long before I saw the film *Chariots Of Fire* I heard people speaking of the score in the most glowing terms. I sensed right away that there must be something very different about it. Now, I had already heard the music from the album, but I had yet to see the picture, so I could only wonder what all the talk was about. But when I went to see the picture I could understand. First of all, to use an electronic score for a film like that is innovative, certainly not the obvious choice. Then I started seeing (and hearing) other ingenious touches. In one effect, a runner is moving in slow motion and the music is right out front; all other sound, both dialogue and effects, is suspended at this point and the music tells it all. What happened was that the director (or writer or producer) provided a unique circumstance where the underscore stepped forward to supply dramatic narration. True, you might not shoot a whole film this way, but a magic moment was created with a skillful marriage of music and picture. There are other pioneering examples, as in *The Graduate* where Dustin Hoffman is swimming in the pool, or in *Butch Cassidy* where Redford and Newman are riding bicycles to the tune of "Raindrops"...but of course these are songs whereas I'm talking about the new use of cues, or underscores.

No doubt MTV is responsible for this combustion of energy, for everyone now wants a *Miami Vice* type score. What you're seeing in this kind of film music is what I call "through" scoring whereby, although the music is appropriate, it goes through the scene on its own without attempting to hit "action cues" in the traditional way. Instead of being subservient to every other parameter of the film, such as dialogue or sound effect or change of dramatic mood, the music has an identity of its own which it does not abandon during the length of its existence on the screen. In movies wheich embrace this idea, such as the Tangerine Dream score to *Risky Business*, you hardly ever hear action cues in the music...it just goes right through the scene, not something apart from the film, but as an element with its own identity which belongs in the film.

db: What are your most often used electronic unstruments?

DK: I lean on the Fairlight a lot; that's my main axe. Once I learned its potential and how to use it for what I want to do, it seemed to be the right instrument for me. I can come up with about any sound I can visualize, either through sampling or through original waveform manipulation.

You must remember that this hardware, this computerized music equipment, does not have a sound of its own as does, say the electric organ in your living room. It is merely a very expensive piece of equipment just sitting there waiting to take instructions, for it hasn't the slightest idea of what to do. People have a tendency to be very wary of computers, thinking they are smarter than we are. On the contrary, they are very, very stupid, but extremely obedient. We take advantage of this by telling them exactly what we want. You can't say the same thing about a clarinet or a harp. The only catch is, we have to be very careful what we tell it, for it's going to do exactly that. The only limitation of the computer is our imagination. And our imagaination is given a terrific boost by software which can be purchased on plastic disks and inserted into the computer, program information which tells both the computer and us what is possible to be done. The people who write programs recorded on this software are called programmers; they may or may not be musicians, but they certainly know computers.

I'm finding that the software written for the Fairlight is very user-friendly, and even more so (I hear) in the newer version, the Series III. This is good news for people like myself who are not brilliant in either mathematics or electronic theory, but who are forced to approach all this purely from a musical level. With the programs available on disk, you can take total control, unlike that organ in your living room where you are presented with perhaps two dozen sounds to choose from. But on computerized equipment, there are unlimited ways to form the sounds, for you are essentially delving into the guts of the thing (through the disk programs) and changing the most basic parameters of the sound waves. You have complete control of your product, which in itself is the excitement for me. People who use such fine equipment merely to reproduce traditional instruments are wandering up a dead end. If you want the sound of trombones, you'll be far ahead of the game by hiring trombone players.

If I were to speculate on which company to watch I would not hesitate to say Yamaha. They're doing some noteworthy things with their FM synthesis, in very econimical ways. They are bringing some complicated digital manipulation, exciting new sounds, to users at a surprisingly low cost.

Another valuable innovation in our business is MIDI. With a MIDI hookup I can use the Fairlight to control all of the other instruments you see in my studio. In other words, I've got the sound resources of such instruments as the Yamaha, the Oberheim, the Roland, or whatever else I need, all of which are controlled through the MIDI cable. So, I'm writing music on the Fairlight and reproducing it through these other instruments via the MIDI connection. That's the direction in which electronic music is going.

db:I'm not interested in plugging brand names, but did you have to go through a lot of soul searching before deciding what to buy?

DK: Of course. I couldn't afford to plunge in without giving it a lot of thought. The instrument I bought seems to be the right one for me. Another popular instrument, the Synclavier, works on a different principle and is another true heavywieght in fine equipment...probably the most expensive. Therefore, it was a major decision for me, for I had to consider both performance and cost. Yes, there was some soul searching, for you don't just go out and blithely write a check for something like this without sweating a little. In the final analysis I decided on the setup you see here: the computer is used as a control source for many less expensive instruments. I feel this gives me many, if not most, of the finest features of more elaborate and costly equipment. In a way it's a pointless comparison, defending product A against B or C or D, for no one can tell you what to spend or what you should be able to accomplish. It's a very private matter, and one that's a bit dangerous, for your emotions can get in the way when you're out there kicking tires and looking at the velour upholstery.

db: But now that you've made your choice and spent your money, are you worried about obsolescence?

DK: I learned to grow with the knowledge that obsolescence exists no matter what you buy. But, do I worry about it? No, not anymore. You learn as you go along. I could have started out by buying several synthesizers, then a sampler, then a sequencer, then a drum machine, and then....What I began to see was a never ending spiral of inexpensive equipment, all leading up to a king's ransom, with none of it being the total answer. So, for me it was necessary to get a machine that would enable me to put it all together right now. As for obsolescence, I finally had to look at it this way: you can sit still and let everything pass you by, or you can jump on the bandwagon and start working.

Again, since I'm not of a technical nature, what's going to keep my hardware from becoming obsolete is the software, which can be updated on a regular basis at a reasonable cost. Since I had to jump into this field without a PhD in mathematics, what particularly impresses me about the fairlight is that you can approach everything on a musical level; you can go in there without knowing a lot about computers and deal with this machine in a no nonsense manner and produce music with it.

I wouldn't be surprised if within, say, two years, all the manufacturers that I mentioned earlier...Emulator, Synclavier, Yamaha, Roland, Apple and IBM...are going to standardize on a system similar to my Fairlight at a fraction of the cost.

db: Don't bet on it. Manufacturers are far too wary of each other to cooperate. I can tell you this from my experience with personal computers. The industry is shamefully paranoid.

DK: Perhaps you're right, but we can't wait for that problem to heal. We must settle for what we can afford for what we want. Some people can write brilliant music with an old Prophet Five and an 8-track tape deck. If you can realize your musical potential goals within these parameters, I say more power to you, and I really mean it, for you can get hopelessly caught up in this "equipment" game, and it can milk you dry. It's like a drug. And companies still keep coming out with more to tease your addiction. You just have to do what you have to do after setting your limits. It's no different now than when western music was being born. If the king came down and commissioned you to write a string quartet, you had to work with four instruments, even though you looked forward to the day when you'd be commissioned to write for a much larger ensemble. But then you'd just have bigger limits.

In the last analysis we're seeing a revolution. The hardware, the software, and the incredible MIDI connectors which join it all together, only serves to remind me that things are changing at an exponential rate, that we are being handed a most fantastic array of innovative equipment to choose from. But there's a warning here: we can't exchange our respect for music for an addiction to equipment. In the end it's only the music. ■

Mark E. Battersby

On Taxes

Insurance For The Studio Owner

• Virtually every major decision made contains some element of risk. Usually, business risks involve a possible profit, whereas insurable risks involve only a possible loss. Naturally, not every risk involving a possible loss is insurable, so it follows that the recording studio owner would not normally seek the same types or amounts of insurance coverage that a larger recording business would.

Today, the most common—and most necessary type of insurance is so-called "negligence" liability. With the increasingly successful efforts of plaintiffs' attorneys in the negligence field, it is difficult to feel that any activity in which the recording operation is engaged will be free of alleged negligent damage to the person or property of others.

There are countless risks of potential liability for damages because of personal injury or property damage from operating a recording activity. These can arise out of operations at or from occupied premises; on the premises of others; ownership of non-occupied premises; use, ownership or control of equipment or other property such as vehicles, aircraft or watercraft; or even resulting from the performance of services for others.

Statutory liability insurance, on the other hand, provides protection from accidents arising under various state and federal laws. Some states, for example, have "Safe Places To Work" statutes covering liability for bodily injuries arising out of failure to provide a safe place to work.

If, however, there are employees in the picture, the chances are that your recording studio or operation is a business. And businesses have even more insurance needs than the hobbyist. There are several types of insurance that no recording business would dream of ignoring. For example: general liability, fire insurance, workers' compensation, business Interruption insurance, vehicle insurance.

Plus, there are also a number of optional coverages that may be available, necessary, and affordable depending upon the recording operation owner's situation. These optional coverages include: burglary, robbery, life insurance, key-man life insurance, and fidelity bonds.

With the tremendous variety of insurance available —both necessary and desirable—there is one common factor that contributes generally to the premium cost—your history of losses. It should be obvious to every recording business owner that the better the loss experience, the better the owner's bargaining position with insurance underwriters for broader coverage, lower cost, and leverage in loss adjusting.

Whether business or hobby, there is one tool that can help reduce losses and improve the experience rating. Loss control, whether exercised by a small home-recording operation or the largest multi-national pays off. Loss control embraces more than prevention. It includes prompt and effective reporting procedures, minimizing the basic causes and efforts to prevent reoccurence. Alert recording business owners solicit help from their insurance carriers and listen carefully to all recommendations.

Most owners run into trouble with

so-called "third party" liability losses. These losses can be reduced by avoiding assuming liability of others through acceptance of contracts containing hold harmless provisions. Try to assume nothing except liability for sole negligence. Consider having contractors and suppliers assume all liability arising out of their supplying goods or services. A good attorney can explain how to legally accomplish this cost-cutting protective measure.

The subject of contractual liability is not widely understood and yet it provides potentially serious loss possibilities often overlooked and therefore frequently uninsured. Oral or written contractual responsibility can create a liability in the eyes of the law which is independent of any statutory or negligence liability. Obviously every contract should be reviewed for potential contractual liability.

A word of caution about the one contract most of us take for granted—the lease. Recording business operators should avoid accepting leases where the lessee agrees to return the property in the condition received, wear and tear excepted. This makes the lessee an "all risk" insurer but probably without insurance other than a properly written umbrella liability policy. Most fire legal liability policies exclude contractual liability. The point, however, is to avoid potential loss producing situations.

In most cases, losses or claims will develop even with the most elaborate prevention program. Still, an effective program to minimize the after effects will be highly productive. Fire losses cannot be totally prevented although a well designed fire protection system, proper fire fighting equipment, etc., can frequently minimize such incidents.

Dishonesty losses cannot be completely prevented either. Their incidence and severity can be reduced by an awareness of how embezzlers and thieves work.

Probably the most common opportunity to minimize loss effect comes in workmen's compensation and third-party accidents. Insurance costs on workmen's compensation, general, product, automobile and other liability is directly related to loss experience. Most loss dollars used in experience ratings, however, are the result of someone's guess. It's an educated guess, to be sure, but a guess nevertheless. Insurance companies have claim examiners who estimate the value of all workmen's compensation and third party liability claims not fully settled.

Prompt notice of any loss or claim is not only required but is in your recording operation's best interest. This is particularly true in liability claims where witnesses may disappear. A reporting procedure for all claims should be established with your insurer beforehand so that everyone knows what is expected. Be certain that if there is a possibility that a third party, e.g., a supplier, may be liable, the report calls attention to it. If his coverage has previously been certified, this should also be noted.

Loss control is an important program. It is nothing to take for granted. Nor should it ever be stagnant. It should expand and be refined as time goes on and show growth in the music or recording operation. The owner who seeks constantly to improve in all facets will probably develop a more effective loss control program resulting with broader insuring provisions and lower costs than one which ignores opportunities.

Self-insurance is often discussed but generally misunderstood. After all, why should any music business owner pay an insurer's overhead? Many music-related business owners routinely assume the entire property risk where there is no catastrophe involved; e.g., collision in private passenger cars, plate glass, parcel post, etc. This is sensible. Of course, the risks which can probably be safely assumed over the long pull will vary greatly between operations.

The most common form of self-assumption in property coverage is the deductible. Normally deductibles are not available in statutory liability coverages and when they are used in other third party liability insurance, it's usually at the insurer's insistence to avoid the expense and nuisance of small claims.

The so-called "disappearing deductible" is used only in property, principally fire insurance. Here, the net loss, after application of the deductible, is increased by a fixed percentage, thus, partially or totally offsetting the deductible. The larger the loss, the larger the dollar offset.

The distinction between self-insurance and other self-assumption is that self-assumptions generally are simply expressed or written off, whereas self-insurance usually involves setting aside (non-tax deductible) reserves. The trend is distinctly away from liability self-insurance. After all, modern liability rating plans allow sophisticated buyers an almost unlimited choice of combinations of self-assumption and insurance services. In other words, in most cases, self-insurance is no bargain.

Finally, some recorder's insurance is purchased directly from insurers through full-time salespeople. Most insurance, however, is handled through brokers and agents. The legal distinction between broker and agent is that the brokers represent the insured and theoretically are not limited to any particular insurance company, whereas the agent represents the insurer and can be offered to only those insurers represented.

As a practical matter, most brokers and agents are independent business people or organizations, many of whom deal with or represent enough insurance companies to make the distinction unimportant as it relates to the buyer. What is important to consider is that there are all levels of competence, ranging from part-time salesmen of little or no competence to large, professionally oriented organizations staffed with qualified experts in all phases of coverage and loss controls.

Theoretically, consultants provide independent advice. They vary from practitioners of little competence but great salesmanship, sometimes with elaborate picture and graph-laden reports, to highly competent effective organizations. Consultants charge a fee for services rendered. Some are really brokers or agents willing to waive the fee in return for being named broker or agent.

It should be evident that the insurance program of your recording operation is nothing to take for granted. Shortcomings can result in unnecessary and possibly crippling financial loss. Many bankrupt studio owners would now be enjoying their profits except for an inadequate insurance program.

Where there is any self-assumption, there should be an awareness of it. Unfortunately, self-assumption is not always intentional. It all too frequently exists because of ignorance of the availability of coverage. Every music or recording business owner and manager should know what he has or doesn't have. This requires considerable thought, as well as knowledge on someone's part. Whether that person is you or a professional—the best time for improving insurance coverage is now. ■

N. Pasquariello

Leslie Shatz
Sound Design For The Japanese

Get a glimpse of how sound designer Leslie Shatz went about creating the soundtrack for the Japanese feature film, Mishima.

How does an American sound designer, who has never been to Japan and does not understand Japanese, go about creating the soundtrack for a Japanese feature film about the life and death of that country's foremost post-war cultural hero?

Should he travel to Japan to learn something about Japanese culture before embarking on this project? Or should he first study the language and customs of this ancient, yet modern people?

And perhaps most importantly, what would be the best hardware to utilize in creating raw sound for this potentially exotic track?

These are just a few of the questions that sound designer Leslie Shatz faced after co-producer, Tom Luddy, hired him to design the soundtrack for the controversial Japanese/ American co-production, *Mishima.*

Initially, director Paul Schrader provided Shatz with a complete script of the biographical study of Yukio Mishima—Japan's acclaimed post-war novelist, playwright and author, who, in 1970 committed ritual suicide while attempting a military coup at the Department of Defense.

Leslie Shatz's work began at the same time as the picture editor's, which is a rarity for Hollywood films, but is fairly common for post-productions done in the San Francisco area. (*Mishima* was post-produced at Lucasfilm and Russian Hill Recording). One of the advantages of such an early entry into the project was that Shatz was able to screen the partially edited workprint of *Mishima* almost as soon as there was one.

In the sections based on Mishima's novels, Leslie Shatz discovered extraordinary visuals in the sets of famed Japanese production designer, Eiko Ishioka. "Her style is a super realistic one that is surreal. She uses extremely vivid colors and schematic props and archetypes to represent reality, ideas, and images. It was her work that inspired mine; this was my main preparation for my sound design," said Shatz. (He decided he had all the material he needed for his sound design at hand, and did not need to travel to Japan after all.)

The task was clear to this ten-year veteran of post-production film sound work: to create a textured, complex sound design analogous to Ishioka's sets.

What hardware would help him create these "archetypes of sound"?

On the advice of musician and synthesist friends, Leslie Shatz decided to explore the creation of sounds on a piece of equipment he had never used before—the DX-7 Synthesizer. The Yamaha DX-7 had been recommended for its unusual capability to imitate natural sounds, yet create sounds which were disjointed from reality, as Ishioka's sets were disjointed from the reality of Mishima's life in the film.

"So I got the DX-7, put it in my editing room and spent two agonizing weeks learning how to program this device. It was real agony. It was like going to school again. I started pushing buttons and for days I got nothing but disappointment. I kept playing the cricket sound (which he had successfully mastered) and thinking, 'We have this. There must be more possibilities.'"

Leslie Shatz's feature film sound credits include: *Apocalypse Now* (1979), *Dragonslayer* (1981), *One from the Heart* (1982), *Rumblefish* (1983), *The Black Stallion Returns* (1983), *Once Upon a time in America* (1984), *Dune* (1984) and *Mishima* (1985).

DB: Would you give me an example of how Ishioka's sets inspired or influenced you to do a specific sound effect, design, or sequence on *Mishima*?

LS: Yes. The entire concept for sound for her sequences was derived by the application of the same concept that she used; I wanted to find archetypes of naturalistic sounds and to apply them very specifically and achieve a heightened sense of realism because that would be the only proper companion to Ishioka's sets.

Maybe you're right; maybe I did get a little bit intuitive with it. I managed to program the DX-7 to make a nightingale that not only was realistic sounding, but one I could perform. I could play it in many different ways. Depending on how hard I pressed the key or which key I pressed, I would get a different nightingale.

DB: That's just one effect.

LS: That's just one effect and at that point things started to open up for me.

DB: Do you understand why?

LS: I don't know; I guess it's part of the creative process. All of the days that I felt were spent wasted and just pushing buttons and fooling around, I guess I did begin to understand, on a deeper level, what this machine was doing.

I think if you talk to anybody about the DX-7, they'll tell you programming it is a nightmare. There are guys whose specialty is just to do that. At many points I thought, "Oh, God! It's worth it for me to pay my own money to somebody to bail me out of this mess."

I guess I finally understood how it worked though. I started to get wind and wind chimes and different kinds of birds and frogs and ducks and owls and seagulls.

These were to be part of the sound design for the section of the film dealing with the first novel, which was *The Temple of the Golden Pavillion.* There was one line that always stood out in my mind: "The beauty of nature is sheer hell." It was a line that this one cynical acolyte said to a young boy as they were walking down this beautiful path.

And so I thought that line was a great banner to keep in my mind as I tried to program these sounds. I was making realistic sounds, but when I got them all together it sounded like some kind of a prison that these people were in. The beauty was so sweet and sickening that I felt it conveyed that concept.

DB: So how much of this was suggested by Schrader?

LS: Not a single thing. In fact, I hid what I was doing from everybody because I thought they would laugh and say, "Come on, get this stuff out of here. We've got to get down to real work."

One day Schrader came into the mixing room and was just standing there. He leaned against the synthesizer. He didn't know what it was. Then the sound of a car revving up came out of the speaker. Schrader started looking around and asked, "What's going on?"

And then he looked back and pressed a key and laughed. He realized this was what I'd been doing. I said: "Oh, yes. I got a bunch of other stuff here, too." He said:"Oh, wow! That's great, fantastic."

I really felt that after all this effort I wanted to sit down and play him every sound and say, "Gee, Paul, this one took me two weeks. This one took me a week. What do you think of this one?" But he wasn't that kind of guy. He wanted me to get the work done and that was it.

DB: How long did this preparatory phase last?

LS: A month and a half, maybe two months. All the while I was trying to lay the ground work for the rest of the work and we had scratch mixes.

Oh, God! Scratch mixes are becoming the bane of the sound designer's existence. First, they (producers and/or directors) want to show the movie in its bare form and then directors get an itch—and rightfully so. They want it to be as good as it possibly can when they project it for themselves and for other people. So they say, "Do you have the sound of a door opening and closing, a car going by...?"

Soon, it becomes more expanded and you start cutting actual sequences of sound and you realize, "Hey, I don't want to do crummy work that's going to be seen by everybody." In the middle of a terribly cut sequence you can't just say, "Oh well, this is just for the scratch mix."

So you start having to lavish large amounts of labor to do it. Then the scratch mix is obsolete the minute the film is projected because they go back to the cutting room and make changes.

And this was complicated because Philip Glass' music is fairly continuous so you can't cut all the elements. I was trying to juggle all of that while I was making these sounds with the synthesizer. I wasn't getting a lot of sleep because then I would be remixing the scratch mix at 10:00 pm.

But my big advantage was that rather than waiting until the very end to have Paul hear these sounds, I would slip one or two of them in various scratch mixes, so that people would hear them and would become comfortable with them. At first people were very uncomfortable and would say, "Well, what's that? What is that sound? I don't like that."

I guess I was becoming sort of woebegone because I didn't know if I was on the right track or if what I was pursuing was valid. And finally we had a screening for George Lucas and Francis Coppola (co-executive producers of *Mishima*). They had several comments about the film and one of them was that they thought these sounds were great and that they wanted more of them.

The sound was one of the elements that was going to differentiate the novel sections from the rest of the film or at least that's the way Tom Luddy related it to me. It made my spirits very buoyant. After that screening Paul came up to me and said, "Do as much of it as you can." Now he was embracing it fully, whereas before he was sort of standing on the sidelines.

DB: So that was some sort of a consensus?

LS: Yes, at that point it was. When Francis and George agree on something it becomes a consensus very quickly!

DB: Well, how does this compare to previous films you've worked on in terms of the way you went about creating sounds and experimenting and researching?

LS: Normally, I will sit with a director who will have very specific comments and concepts at various points in the reel. I want this here and that there. He'll probably want to hear very specific sounds and at that point you go to work in a pretty linear fashion. You gather the sounds you need. You get a crew of people together who can work fast and well. You record Foley and then you go to the mix.

But on this picture, (*Mishima*), the director never sat down with me to spot it. He never communicated his vision of the soundtrack of the movie, which I think was evolving all the way along. He wasn't familiar with the mechanical aspects of the soundtrack so I was on my own. Also, the budget on the film was low so I had to do things in cost cutting ways wherever I could. For instance, my assistant became the Foley walker and then he became the dialogue editor. I trained him in these various tasks. We normally record Foley on 10 or 12 tracks and it's a big chore to cut it and mix it.

We did the Foley at a place called Russian Hill Recording. They were very cooperative with us and it was quite an encouraging environment to work in. They recorded to 2-inch multi-track tape so we weren't limited by the number of tracks that we used. We then did Foley in the normal spread out fashion of up to 12 tracks per reel. We mixed 12 tracks down to 2 tracks; in some cases we have a third track, but not usually. Then the editor just had 8 tracks to cut. It was really great because if the synch did become really screwy we could go back to the 24-track and transfer those little bits that were wrong. We didn't have to do that though.

DB: So early on in the project you were given Glass' pre-recorded music track?

LS: He was able to do a temp version of his score for the entire movie within the third week of editing, which is pretty amazing. This wasn't just him playing the piano; it was with bells and strings—all of the instruments that represented what was going to be there in the final performance. And considering that the score runs so continuously throughout the movie this was such a boon; I don't think the film could have been edited or created in the way that it was without this music on hand in that early stage of the filmmaking.

I would hope that all composers would be able to move in that direction. So often in a film you work and edit the picture and cut the sound effects. Then, at the very last minute, they bring in a score that they've spent hundreds of thousands of dollars on, but have only heard once or twice. So there's a mad scramble to try to figure out how it fits with the movie.

DB: So you had a nearly completed score going into the mix?

LS: No, we had a complete rendition of the score three weeks into the editing process. Then we cut the score and the picture, one against the other. The picture would dictate cuts in the music and the music would dictate certain picture cuts. Then nearing the final cut of the film these notes and cassettes of the way the music was applied to the picture were sent back to Mr. Glass. Then he conducted and recorded the real instruments to make the real score onto multi-track.

Then we brought that back on multi-track to Lucasfilm where we mixed the music down to the picture, which is a technique that was used exclusively in the old days when music was recorded to 3- and 4-track and it wasn't a big deal. Now it's not done so much anymore.

Again, in this case, I don't see how we could have done it any other way. We used the cue lock and locked the film recorder at Lucasfilm and the projector in the big room up to the multi-track downstairs. It was kind of a nightmarish session because the cue lock had many problems as they often do. But what we achieved was a tailored mix and a tailored cut of the music because in many cases the music had to be trimmed or expanded.

It was done with the director, composer, editor, music producer, music conductor and myself, so there were no surprises.

Nowadays music is usually delivered on the mixing stage as a 3-track—left, center and right—which gives the director no option on how to play it. I think that's a mistake because the mixdown of the music is usually just as critical as the mixdown of the sound effects or the dialogue. There's so much flexibility in the way you mix music in terms of the instruments you choose to feature or even leave out.

On the movie I'm working on now (*Natty Gann* at Disney Studios), the director likes a particular music cue very well, but there's one instrument in it—a marimba—that he hates. The composer gave him a completely mixed down track so his only option was to drop the cue, which was foolish, so he's going to go back and remix it.

DB: On *Mishima,* how many tracks did you deliver to Schrader?

LS: It varied because certain reels were more complex, but we probably had 50 tracks of sound for any particular reel. There are so many elements involved: dialogue, ADR, and Foley. There's a whole set of real sound effects that had to support the real part of the film—the black and white and biographical sections. Then there were the stereo sound effects which were the ones that I was creating.

See, they had never planned mixing the movie in stereo. They had planned mixing it in mono because it was a low-budget film and they had the notion that it would be too expensive to mix it in stereo. All the while I felt it was ridiculous to make this movie with a strong element from music and strong element from sound in mono. I was planning on a stereo mix regardless of what they said, which is why I prepared my sound effects in stereo.

What we ended up doing was having the parts that dealt with Mishima's final day and of Mishima's biography in mono. We spread the sound image to stereo whenever we entered the world of novels, which was another way we distinguished the novel sections from the rest of the film.

DB: Was that your idea?

LS: Yes it was, and Paul took to it very well.

DB: From what I know about the Hollywood film industry, it's hard to imagine that anyone would be given the kind of freedom to research and explore original concepts for sound design like you did on *Mishima.* Could you imagine working that way down there?

LS: Every film is different.

DB: Have you ever worked on a film that way in Hollywood?

LS: This one (*Natty Gann* for Disney Productions) was sort of like that, but I snuck it in on them. I never told anybody, "Hey, I'm just going to go off and do sound design and all of this conceptual work...."

DB: But you snuck it in on *Mishima,* too.

LS: Well yes, it's true, I snuck it in on *Mishima* as far as the director was concerned, but the producer always knew that was what the film required. That's why he hired me. And it was just a matter of bringing the director up to speed.

Tom Luddy (co-producer of *Mishima*) played a very important role because he got Michael Chandler (editor), me, Philip Glass, and Eiko Ishioka. He got all of those people around Paul and all of those people had such an incredible contribution. But it's always that way. Filmmaking is collaborative and that's why I find it so exciting. I don't know how well I would do if I was just out on my own, doing my own little compositions and saying, "Here world, here is my stuff." It's so much more exciting to be in the midst of many other people who have great minds and great contributions.

Philip Glass' music inspired me so much. Many of my sounds were tailored to fit within his music. I could control the pitch of the sound on my keyboard to match the key of his music. That worked out very well in certain cases. That was another advantage of having the score so much in advance. I could do stuff that wouldn't be buried by it or try to overpower it.

Bruce Bartlett

Recording Techniques

Introduction To SMPTE Time Code

• Have you ever wished you had more tracks? Suppose you've filled up all the tracks of a 16-track recorder, and the band you're recording wants to overdub several more instruments. You don't have a 24-track machine.

There's an alternative: synchronize the 16-track machine with an 8-track machine by using SMPTE time code. This is a special signal recorded on tape that can sync together two tape recorders so that they operate as one. Time code can also synchronize an audio recorder with a video recorder.

Here's how it works: A time code generator creates the time code signal (a 1200 Hz modulated square wave). You record or "stripe" this signal onto one track of both recorders. A time code reader reads the code off the two tapes. Then a time code synchronizer compares the codes from the two transports and locks them together in time by varying the motor speed of one of the transports.

SMPTE stands for the Society of Motion Picture and Television Engineers. They standardized the time code signal for use in video production, and it can be used in audio recording as well.

SMPTE time code is something like a digital tape counter, where the counter time is recorded as a signal on tape. Let's explain.

Pictures on a video screen are updated approximately thirty frames per second, where a frame is a still picture made of 525 lines on the screen. SMPTE time code assigns a unique number (address) to each video frame: eight digits that specify hours, minutes, seconds, and frame numbers. Each video frame is identified with its own time code address.

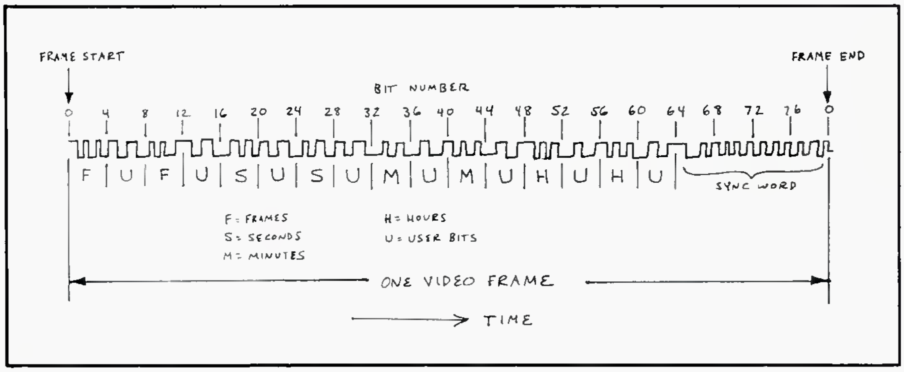


Figure 1. An 80-bit time code word.

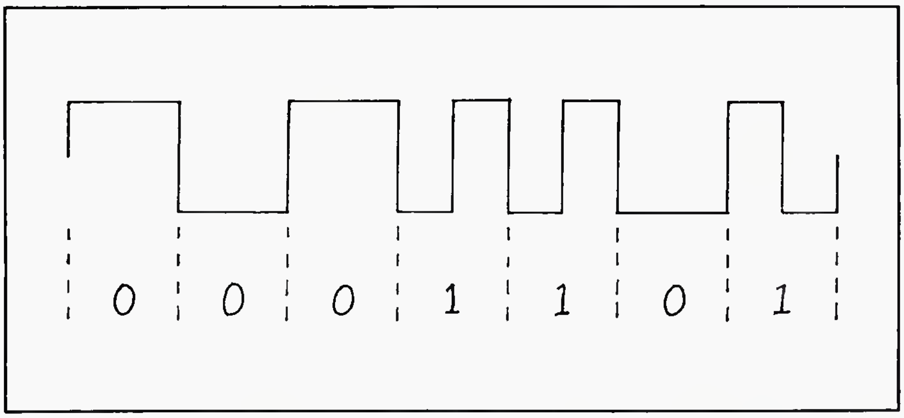


Figure 2. Bi-phase modulation used in SMPTE time code. A voltage change in the middle of a half-cycle (bit number) signifies a binary 1; no change signifies a binary 0.

These addresses are recorded sequentially: For each successive video frame, the time code number increases by one frame count. There are approximately thirty frames per second in the American TV system, so the time code counts frames from zero to twenty-nine each second.

SIGNAL DETAILS

SMPTE time code is a data stream divided into code words. Each code word includes eighty binary digits or bits that identify each video frame (*Figure 1*).

The 80-bit time code word is synched to the start of each video frame. The code uses binary ones and zeros. During each half-cycle of the square wave, the voltage may be constant (signifying a zero) or changing (signifying a one). That is, a voltage transition in the middle of a half-cycle of the square wave equals a one. No transition signifies a zero. This is called bi-phase modulation (*Figure 2*). It can be read forward or in reverse, at almost any tape speed.

A time code reader detects the binary ones and zeros, and converts them to decimal numbers to form the time code addresses.

SMPTE words also can include user information. There are thirty-two multi-purpose bits (eight digits or four characters) reserved for the user's data, such as the take number.

The last sixteen bits in the word are a fixed number of ones and zeros called sync bits. These bits indicate the end of the time code word, so that the time code reader can tell whether the code is being read forward or in reverse.

DROP-FRAME MODE

SMPTE code can run in various modes depending on the application. Let's explain drop-frame mode and why it's needed.

Black-and-white TV runs at 30 frames/sec. A time code signal also running at 30 frames/sec will agree with the clock on the wall.

Color TV, on the other hand, runs at 29.97 frames/sec. If a color program is clocked at 30 frames/sec for one hour, the actual show length will run 3.6 seconds (108 frames) longer than an hour.

Drop-frame causes the time code to count at a rate to match the clock on the wall. Once every minute, frame numbers 00 and 01 are dropped, except every tenth minute. (Instead of seeing frames ...27, 28, 29, 00 on the counter; you'll see frames ...27, 28, 29, 02.) This speeds up the time code counter to match the rate of the video frames.

The video frames still progress at 29.97 frames/sec, while the time code progresses at 30 frames/sec, but drops every few frames—so that the effective time-code frame rate is 29.97 frames/sec.

You program the time code generator to operate in Drop or Non-Drop mode. Non-Drop can be used for audio-only synchronizing, but Drop mode should be used if the audio will be synched to a video tape later on.

SETTING UP A TIME CODE SYSTEM

To use SMPTE time code, you need a time code generator, reader, and synchronizer. These may be all-in-one or separate. *Figure 3* shows a typical system hookup, in which the generator, reader, and synchronizer are combined in one unit.

Set the generator to Time-of-Day

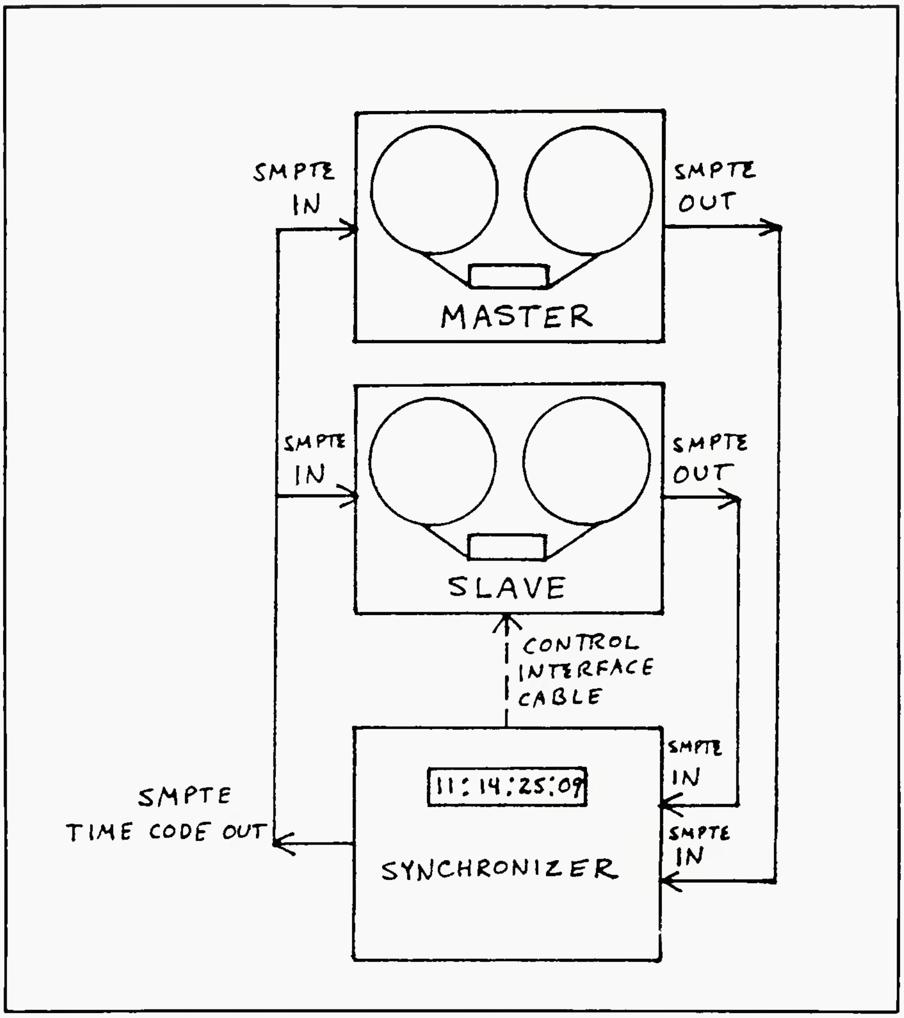


Figure 3. Typical hook-up for synchronizing two tape transports with SMPTE time code.

code, or any other convenient starting time.

If you are synching to video, feed the generator a sync signal from the video source being recorded. This will lock the generator together in time with the video source. For audio-only applications, use the internal crystal sync.

Select Drop-Frame or Non-Drop-Frame mode, and stay with it for the entire production. Use Drop-frame mode if you anticipate synching audio to video in the future.

Next, set the frame rate: 29.97, 30, 24, or 25 frames/sec. Color video productions require 29.97; black-and-white TV or audio-only productions use 30; film usually runs at 24; and European TV (using EBU time code) requires 25.

The time code signal appears at the generator output, which is a standard 3-pin audio connector. Signal level is +4 dBm. The signal is fed through a standard 2-conductor shielded audio cable. To avoid crosstalk of time code into audio channels, separate the time code cables from audio cables. Patch the time code signal into an outside track of the recorders you want to lock together.

Patch the outputs of those time-code tracks to the inputs of the time code reader. The reader decodes the information recorded on tape and, in some models, displays the time code data in an *HOURS: MINUTES: SECONDS:FRAMES* format. Some readers have an error bypass feature which corrects for missing data.

The time code synchronizer matches bits between two time-code signals to synchronize them. The synchronizer compares tape direction, address, and phase to synchronize two SMPTE tracks via servo control of the transport motors.

The two tape machines to be synched are called "master" and "slave." The synchronizer controls the slave by making its tape position and speed follow that of the master.

Connect the shielded, multipin interface cable between the synchronizer and slave machine to control the slave's tape transport and motors. This interface cable has channels for controlling the capstan motor, tape direction, shuttle modes, and tach (more on tach later).

Since the time-code signal becomes very high in frequency when the tape is shuttled rapidly, special playback amplifier cards with extended high-frequency reponse may be needed to reproduce the SMPTE signal accurately. These cards are available from the recorder manufacturer.

Unfortunately, when the tape is in shuttle mode (fast forward or rewind), the tape usually is lifted from the heads—losing the SMPTE signal. In this case, the recorders are synchronized with tach pulses from the recorders used as a replacement for SMPTE time code. Some synchronizers are fed tach pulses from the slave only.

If chase mode is available, the slave follows the shuttle motions of the master. Otherwise, the synchronizer notes the address of the master tape when it is stopped and cues the slave to match that location. Chase mode is useful for repetitive overdubs.

HOW TO USE SMPTE TIME CODE

Suppose you want to synchronize two multitrack recorders. First clean the heads and tape path. Record SMPTE time code on an outside track of both recorders at -5VU to -10 VU, leaving the adjacent track blank if possible to avoid time-code crosstalk. Start recording the code about twenty seconds before the music starts, nonstop with no breaks in the signal.

Stripe the two tapes simultaneously. If that is not possible, you'll need a time code editor to correct or insert an offset.

During playback, manually cue the slave to approximately the same point as the master tape, using time code address information as a reference. Then put both recorders in play mode. Finally, adjust the slave's tape speed to gradually reduce the error between transports to less than one time code frame.

With some synchronizers, this operation is automatic. You set the slave tape to approximately the same point as the master tape. Then put the master in play. When the synchronizer detects master time code, it will set the slave machine in play mode and, in a few seconds, will adjust the slave's speed to synchronize the two recorders.

When you're recording on two synchronized transports, try not to split stereo pairs between two tapes. The very slight time differences between machines can degrade stereo imaging. Keep all stereo pairs on the same tape, copying them onto the other tape if necessary.

RE-STRIPING DEFECTIVE CODE

You may encounter degraded or erased sections on a time code track. This lost code must be replaced with good code in proper sequence. If you need to re-record (re-stripe) a defective SMPTE track, use the jam sync mode on the time code generator. This feature produces new code which matches the original addresses and frame count.

For example, suppose the slave tape needs to be re-striped. Patch the slave's time-code track into the generator set to Jam Sync mode. Patch the generator output into the time-code-track input on the slave machine. Play the tape. The time code reader built into the generator will detect a section of good code and will initialize the generator with that information. Then start recording the new, regenerated code over the bad data.

Jam Sync also should be used when you copy a tape containing time code. With jam sync in operation, the code will be regenerated to create a clean copy. This procedure is preferable to copying the time code track directly, because each generation can distort the code signal.

SYNCHING TO VIDEO

With the advent of MTV and other audio/video combinations, there's a widespread need to sync audio to video. Running audio and video tapes in synchronization for TV audio editing is a typical post-production method.

Synching audio and video machines also eliminates the dubbing step when transferring (laying back) the audio sound track to video tape. That is, you can mix the multitrack tape master directly to the video tape (keeping sync), rather than mixing down to 2-track and dubbing that to video tape.

When you sync audio and video, select Longitudinal Time Code (LTC) or Vertical Interval Time Code (VITC). Longitudinal code records along the length of an audio track on the video tape. Vertical Interval code is combined with the video signal and is placed in the vertical blanking interval—the black bar seen over the TV picture when it is rolling vertically. VITC frees up an audio track for other purposes.

If you record the time code signal on an audio or cue track of the videotape, do not use automatic level control because it may distort the SMPTE waveform. Instead, adjust the time code signal level manually.

When you play an audio tape synched to video, the time-code track on the audio tape will be delayed with respect to the video's code due to the spacing betweeen the record and playback heads in the audio recorder. This delay (about five frames) can be corrected by the offset function in the synchronizer.

Some time code systems include a character inserter which displays the address on the video monitor. If desired, these addresses can be recorded with (burned into) the picture—a feature called window dub.

OTHER TIME CODE APPLICATIONS

SMPTE time code allows video editing under computer control. To edit a video program, you copy program segments from two or more video tapes onto a third recorder. On a computer you specify the edit points (time code addresses) where you want to switch from one video source to another. You can rehearse edits as often as required without cutting tape.

Time code also is used as an index for locating cue points on tape. During a mixdown, you can use these cue points to indicate where to make changes in the mix.

Studios doing soundtrack work for film or video can use SMPTE time code to synchronize sound and picture for overdubbing narration, lip-sync, music, or sound effects. Time code also can be used as reference for console automation and MIDI instruments. With this latter application, MIDI synthesizers can be cued to any point within a sequence, rather than having to start at the beginning.

By using SMPTE time code to lock together multiple audio or video transports, you can greatly expand your operating flexibility.

For more detail on SMPTE time code, I highly recommend the following book: Time Code Handbook *by Walter A. Hickman, Cipher Digital, 150 Huntington Avenue, Boston, MA, 02115. This book is available for $8 from Bang Campbell Associates, P.O. Box 47, Woods Hole, MA, 02543.* ■

Rita Wolcott

Direct Mail Sales:

What's In It For Me?

YOU GET what you pay for. The old adage usually proves true. And it's always good to keep in mind when buying a piece of audio equipment. Yet, it's also wise to remember that when one decides to buy retail, half of what he's paying will go to the dealer, the distributor, and those in the warranty program. The other half will go towards the product.

Now consider buying the exact same piece of equipment directly from the manufacturer, a practice generally known as direct mail sales. With retail margins eliminated, buyer savings will total between thirty-five and fifty percent. With dealer and distributor shares eliminated, one will usually get more product for his money delivered in less time.

Buying directly also allows the consumer to obtain a higher quality product per dollar than what he would be offered by a dealer program. In addition, because the products are available direct from the factory, the manufacturer gives you the straight facts, and if you ever need assistance you can go directly to the source—the factory where the products are made.

DIRECT TO THE SOURCE

So why do so many choose to pay dealer prices? Lacy Thompson, president of LT Sound offers, "Some people buy retail because they have a relationship with a dealer whom they feel they can trust. Another explanation is the fact that direct sales makes up such a small section of the market, only about five to ten percent."

STUDER A820 MASTER RECORDERS

• The Studer A820 analog mastering recorder, introduced in a standard 1/4-in. stereo configuration last year, is now available in two new versions for either 1/2-in. stereo or 1/4 in. stereo with center-track SMPTE code. The half-inch two-track format A820 is designed primarily for extremely high quality music mastering applicaations. The added tape width of the A820 2-1/2-in. provides significant improvements in both signal-to noise ratio and tape saturation characteristics. The A820-TC features the center-track SMPTE time code system first introduced on the Studer A810. Two separate heads are used to record and reproduce the time code, thus keeping code/audio crosstalk at better than -90 dB. A microprocessor-controlled delay line compensates for the offset created between code and audio heads, thereby maintaining exact coincidence of code and audio at all speeds, including varispeed modes. The basic A820 transport is designed for maximum flexibilty across a full spectrum of audio recording applications. The A820's ability to accommodate 14-in. reels permits longer playing times at higher tape speeds. Four speeds (3 3/4, 7 1/2, 15 and 30 in./sec.) are standard and front panel selectable. All A820 operating features are controlled by a network of microprocessor-based systems, allowing software control of virtually all operating parameters. Approximately 40 different user programmable functions may be assigned to various keys. All audio parameters are digitally set and stored in non-volatile memory. Microprocessors also monitor all dynamics of the transport, including tape tension, tape winding speeds, and reel inertia. A dual thumbwheel control is provided for fast editing: one wheel fast winds at variable speeds in either direction, while the other precisely positions the tape for the edit. For enhanced sonic performance, the A820 incorporates the latest Studer advancements in phase compensated amplifier technology. A choice of transformer or transformerless inputs and outputs is offered.

Mfr: Studer/Revox America, Inc.
Price: A820-2-1/2-in. is $11,000.00.
The A820-TC is $11,500.00.

Circle 52 on Reader Service Card

FOUR DESIGNS RACKKRATE

• Four Designs Company has introduced a new product called Rackkrate. This six space high rack unit is the lowest cost method of securing rack mounted equipment for onstage or road use, as well as in a musician's home studio. Rackkrate's design was inspired by those ubiquitous, indestructable plastic milk crates that seem to be used for everything but carrying milk. Four Designs Company puchases special crates new from the manufacturer and transforms them by attaching threaded mounting rail and padded handles onto the modified crate. In addition to low cost, Rockkrate features light weight, strength and good ventilation. The resiliant plastic crate also offers excellent protection by absorbing shocks and sharp jolts often encountered when loading and transporting equipment. Rackkrate comes with mounting hardware and a one year warranty.

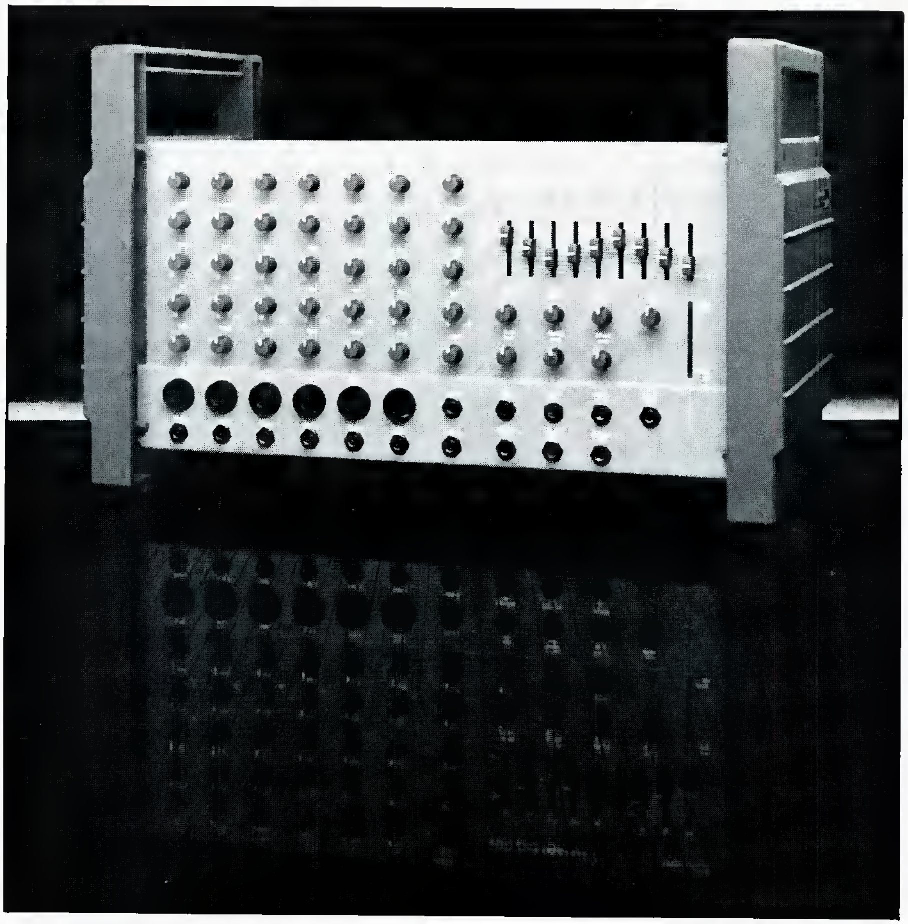
Mfr: Four Designs Company
Price: $49.00.

Circle 53 on Reader Service Card

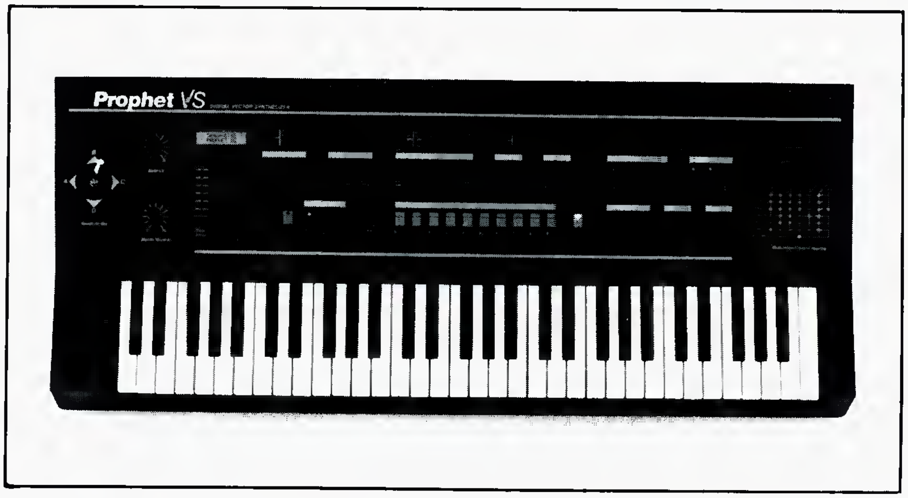
BIAMP SYSTEMS MIXER PLUS

• The new Biamp Systems of Portland, OR, has just launched their first new product, the Mixpak 6Plus+, a powered mixer. The Mixpak 6Plus+ features six input channels plus an electronic drum/synthesizer input channel. The design incorporates high impact molded plastic ends and a rugged steel chassis. The Mixpak 6Plus+ power supply produces 250 watts into 4 ohms and utilizes a refined compressor action auto-limit to protect speaker systems from accidental overload damage. The Mixpak 6Plus+ features a 9-band graphic equalizer, individual reverb and effects controls for main and monitor outputs, complete patching capability, high and low impedance inputs, and LED power meter, power-on and auto limit indicators, and optional snap-on protective front cover. The unit's flow-through venting design results in cooler running electronics for longer life and reliability. The Mixpak 6Plus+ is offered with an exclusive five year limited warranty.
Mfr: Biamp Systems, Inc.
Price: $599.00.

Circle 54 on Reader Service Card



SEQUENTIAL DIGITAL SYNTH

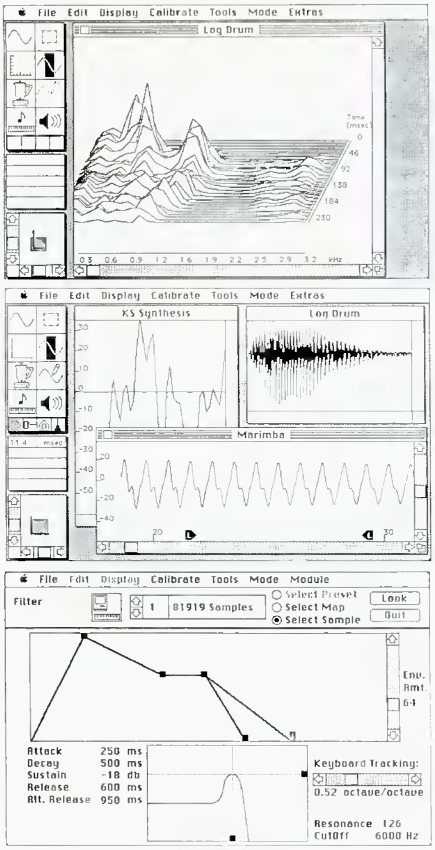


• Sequential's Prophet V5 pioneers a technique called Vector Synthesis, a unique concept in dynamic digital sound creation using an algorithm enabling complex sounds to be created—and most imporatantly—manipulated in real time. As a result, the 8 voice Prophet V5 delivers more real time control than any other synthesizer ever designed. The Prophet V5 Digiital Vector Synthesizer features eight voices, with each voice composed of four digital oscillators. The V5 allows for extensive control over a multitude of parameters as well as the creation of user definable waveforms. Every V5 voice is built from four 12-bit oscillators. Each oscillator's frequency can be controlled independently, using any one of 128 complex waveforms—including white noise. New waveforms are easily created by mixing as many as four existing waveforms using the joystick. Any waveform can be stored internally, or to an external cartridge. Easy waveform construction is inherent to the Prophet V5. Sophisticated electronic and software techniques provide significant control over a comprehensive number of timbre governing parameters. Five stage envelopes allow manipulation of filters, amplitude, and the relative mix of the 4-oscillators per voice. Looping and repeat functions enhance the power of patch programming. Other features include real time control of stereo panning, voice oscillator mixing, stereo chorusing and instant access to 200 programs via ROM and RAM cartridges. The five-octave synthesizer is velocity and pressure sensitive and can be programmed split and stacked. The V5 also includes a versatile arpeggiator, offering new options such as polyphonic voicing, rests and layering. The V5 is built with a rugged steel chassis and uses high quality switches and LED indicators. It is backed by Sequential's full one year warranty.
Mfr: Sequential
Price: $2,599.00.

Circle 56 on Reader Service Card

SOUND DESIGNER SOFTWARE

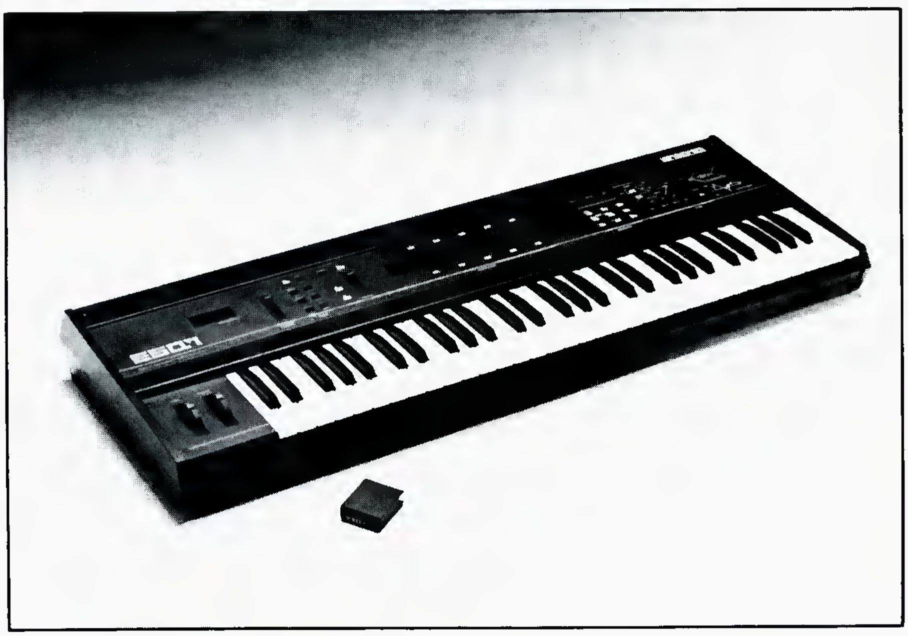
● The Sound Designer 2000 is an advanced computer music system for the Apple Macintosh and Sequential Prophet 2000 digital sampling keyboard. It allows sampled sounds to be transferred between the Macintosh and the Prophet 2000 using a standard Macintosh MIDI interface. Sound data is transferred at twice the normal MIDI data rate (63K baud) to minimize waiting. Sound waveforms are displayed on the Macintosh's screen (up to three waveforms may be displayed simultaneously). The software provides extensive sound editing capabilities, including Macintosh style "cut and paste" editing. The waveform display can be magnified to show extremely fine detail, with editing accuracy to 1/40,000th of a second. Calibration scales provide exact readouts of time and amplitude values at any location in the waveform, and the waveform display can be horizontally and vertically scrolled. Sound Designer 2000 also includes FFT (Fast Fourier Transform) based frequency analysis and modification of sound, digital equalization, enveloping, digital mixing and digital merging, as well as a variety of other digital signal processing functions for modifying sampled sounds and creating unique sounds. Sound Designer's visual looping aids greatly simplify looping. Both sustain and release loops are provided, as well as a crossfade looping program for looping sounds that lack a natural loop area. Direct digital synthesis can be performed on the Macintosh, and the resulting sounds transferred to the Prophet 2000 for playback. Programming of all Prophet 2000 parameters is greatly simplified by Sound Designer's Prophet 2000 graphic programming aids. Filter response curves, ADSR curves, and other graphically represented parameters can be drawn using the Macintosh mouse, and keyboard set-ups, MIDI assignments, controller assignments, etc. can be quickly programmed using on-screen menus.



Mfr: Digidesign Inc.
Price: $495.00.

Circle 57 on Reader Service Card

ENSONIQ SEQUENCER/SYNTH



● Ensoniq Corporation's 8-voice MIDI-based instrument, the ESQ-1 Sequencer/Synthesizer is a powerful complex waveform synthesizer and multitrack MIDI sequencer in one compact keyboard instrument. It is an 8-voice polyphonic, polytimbral synth with the rich sound of three oscillators per voice. There is a choice of 32 multi-sampled and synthetic waveforms for a nearly unlimited variety of sounds and effects. Included are sampled waveforms of pianos, strings, and brass instruments, in addition to a wide variety of synthetic waveforms. There are 40 programs on board with an additional 80 cartridge programs available, giving the player instant access to 120 distinct sounds. The ESQ 1 also features a polyphonic, velocity sensitive 61 note weighted-action keyboard with programmable split points and sound layering on either or both keyboard halves. It also features a "MIDI Overflow Mode" which permits slaving other MIDI units together to create a 16-voice synthesizer. There are also Poly, Omni, Multi and Mono modes plus 8 simultaneous polyphonic channels with separate programs. The ESQ-1 contains a sophisticated sequencer with 2400 note internal storage (expandable to 10,000 notes) and 8 discrete tracks—each with separate program and MIDI channels. Each track has 8 voices dynamically assigned. The sequencer also features a mixdown facility for balancing individual tracks, sync to tape selector, built-in metronome and autolocator, plus 30 separate sequences chainable into 10 songs. External storage of notes from the sequencer can be made on audio tape or on 3 1/2-in. diskettes using the Mirage Digital Sampling Keyboard or Digital Multi Sampler. Standard accessories for the ESQ-1 include the Owner's Guide, Programming Guide, Footswitch/Sustain Pedal, and detachable power cord.

Mfr: Ensoniq Corp.
Price: $1,345.00.

Circle 58 on Reader Service Card

JBL INSTRUMENT SPEAKERS

• The overdrive sound character found in much of today's music has led JBL Professional to introduce a series of musical instrument loudspeakers specifically designed to deliver excellent tone, power and sensitivity. Whether a musician is playing guitar, bass or keyboards, the new JBL G 125 and G 135 speakers combine a smooth, warm overdrive sound character with lyrical sweetness for clean settings. The G Series has a 200-watt continuous pink noise power capacity, allowing performers to crank up amp settings without worry. In order to transfer all the power from the amp with maximum efficiency, a large 3-in. diameter voice coil that is made of aluminum ribbon wire and and precisely edge wound with special high-temperature adhesives is used. This manufacturing process reduces gapping and improves conductivity resulting in greater efficiency and increased power handling. A deep cone and frame as well as a paper dome provides wide, extended frequency response. Unwanted harmonic distortion normally found in conventional speakers is eliminated by JBL's SFG (Symmetrical Field Geometry) magnetic structure. SFG also is keyed in producing deep, powerful clean bass. The JBL G 125 has a high sensitivity rating of 102 dB while the G 135's rating is 104 dB (1 watt input at 1 meter). Additionally, the cast aluminum frame of the G Series provides much better ruggedness, reliability and heat sink action than stamped frames.

Mfr: JBL Professional

Prices: G125: $165.00; G135: $177.00.

Circle 59 on Reader Service Card

SHURE CONDENSER MICS

• The Shure Brother's SM94 and SM96 are electret condenser microphones. The SM94 is designed primarily for instrument mic'ing and recording applications, while the SM96 is best suited for use by vocalists. Both are designed to provide good value and high performance. The mics are intended to fill the gap between very costly condenser microphones and the lower priced, lesser performing condenser models on the market. Both units have frequency responses that are specially tailored for their intended application. The SM94's frequency response is essentially flat, while the vocal-oriented SM96 has a slight presence rise and smooth low-end roll-off to correct for proximity effect and enhance vocal performance. Both microphones have unidirectional (cardioid) polar patterns that will not "collapse" at high frequencies, permitting uniform off-axis response throughout the audio spectrum. Both mics feature a "space frame" shock-mounting system that effectively isolates the transducer element, protecting it from handling noise and stand thumps. They can be powered by virtually any standard phantom power source (12-48V dc), or internally by a standard 1.5V AA battery. When a battery is installed in either mic, it can act as a backup power source, taking over automatically if the phantom power source should fail. To provide pop and noise protection for vocalists, the SM96 is equipped with a built-in, 3-stage pop filter. An accessory windscreen is also available for the SM94. Both models have non-reflective gray finish and come supplied with a vinyl storage bag.

Mfr: Shure Brothers, Inc.

Price: $250.00.

Circle 60 on Reader Service Card

People, Places...

• **Robert L. Rosipajla** has been appointed senior vice president/service division at **Mitsubishi Electric Sales America, Inc.** His responsibilities will now include the management of the national service network for all MESA consumer electronics audio/video products. Rosipajla joined MESA in January of 1980. Prior to MESA, he worked for GTE in New York. MESA is a leading manufacturer and supplier of high performance audio/video products.

• **Maria A. Curry** has been named vice president and general manager of **Agfa Gevaert's** magnetic tape division and a member of the Executive Council of Agfa Gevaert. Curry has been with AG since 1959 when she started in the magnetic tape factory working in technical applications in the laboratory. Born and educated in Austria, she is a graduate of the Institute of Technology in Vienna. She came to the US in 1960 to work at AG as a technician and technical liaison with headquarters in Leverkusen.

• **Alpha Bibbs** has been appointed product manger in ADC Communications' data transmission products group. In his new position, Bibbs is responsible for managing product development and implementation of ADC's products in the data transmission area. Bibbs has over 12 years of experience in the Bell System.

• **Encore Studios**, of Burbank, CA, has just completed substantial facility renovations, spearheaded by leading acoustical consultants, Lakeside Associates. Once home of the Kendun Recorders, whose "Super D" room attracted the likes of Quincy Jones, Michael Jackson, Jefferson Starship, and George Benson, Encore is once again offering readily available technology to the recording industry.

• **Music Graphics, Inc.**, a newly established firm specializing in music-oriented film production for television and the home video markets, has opened offices at Kaufman Astoria Studios. Headed by Joe Bilella, formerly of Picture Music, the company is developing a variety of musical subjects for production later this year. Music Graphics will work closely with Master Sound Astoria, the 48-track digital/analog recording facility recently opened at KAS.

& HAPPENINGS

The 128th SMPTE Technical Conference Scheduled

The 128th technical conference and Equipment Exhibit of the Society of Motion Picture and Television Engineers (SMPTE) will be held October 24-29, 1986, at the new Jacob K. Javits Convention Center in New York City. The conference and exhibit will split two weeks, beginning on a Friday and concluding on a Wednesday. This marks the first time that the SMPTE has pust such a format into effect.

THE AUDIO TECHNOLOGIES RESEARCH GROUP

The Audio Technologies Research Group consists of a small representation of students from the Institute of Audio Research in New York City. To further expedite their goal of becoming the most experienced and technologically advanced engineer/producers possible, they created the following program. Weekly seminars will take place until June within the appropriate settings of recording studios, broadcast facilities, concert halls, and other related industry locations. Their goal is to provide members and their guests with vital insight into the expectations of future clientele.

INTERLOCHEN'S NATIONAL MUSIC CAMP TO OPEN RECORDING ARTS AND BROADCASTING INSTITUTE

The National Music Camp at Interlochen, Michigan, now in its fifty-ninth season, will offer a broad range of courses in audio-recording and broadcasting during the 1986 summer season. The Recording Arts and Broadcasting Institute of Interlochen is an intensive, full-time clinic workshop which will help prepare students for the occupation of recording engineer in the audio related industries of radio, television, films, theater, etc. Taught by top professionals in the recording industry, students will have full use of professional recording equipment. Conducted in three three-week sessions throughout the summer, classes combine work experience with lecture/demonstrations by a faculty drawn from noted representatives of the recording industry.

SYNTERGETIC SPONSORS STUDIO DESIGNER'S WORKSHOP

Synergetic Audio Concepts will sponsor a Studio Designer's Workshop to be held May 7-9, 1986, at the new Tele-Image Studios, a 35,000 square foot video and audio production facility located at the Dallas Communications Complex. The facility contains three video editing suites, a screening room, a sound stage and a multi-track recording studio and a control room.

• **The Audio Engineering Society (AES)** will be holding their fourth international conference May 15-18 at the Westin O'Hare Hotel, Rosemont, IL. The following are programs and seminars which will be featured:

THURSDAY, MAY 15
12:00 Noon Registration
5:00 PM Hospitality/6:00 PM Dinner

SESSION ONE/INRODUCTORY SESSION
7:30 PM
Chairman: Robert B. Schulein, Shure Brothers Inc., Evanston, IL.

The conference will begin with an overview of what is planned, and will include a short summary of each speaker's presentation, accompanied by demonstrations.

FRIDAY, MAY 16/SESSION TWO: TRANSMISSION
9:00 AM

Chairman: Don McCroskey, Consultant, ABC Television, Burbank, CA.

The Real World of Video/Broadcast Program Transmission. Don McCroskey.

The Operating Plant: Problems and Traps. Peter Butt, Technical Reviewer, *Recording Engineer/Producer*, Hollywood, CA.

Equipment Performance Specifications. Mike Davis, Audio/Video Systems Engineering, ABC Television, NY.

Monitoring and Mono Compatability. Randy Hoffner, NBC Technical Staff, NBC Television, NY.

The Operational MultiChannel Television Sound Facility. Louis Bardfield, Engineering Supervisor, KTLA, CA.

SESSION THREE/DUPLICATION
Friday, May 16 1:30 PM

Chairman: David Robinson, Dolby Laboratories, San Francisco, CA.

This session will cover video tape and video disk duplicaiton techniques starting with a master tape and then following the product through the premastering and duplicaiton processes.

Film-to-Tape Audio Transfer Considerations. Moshe Barkat, President, Modern Vdieo Film, CA.

Professional Video Format Audio Duplicaitons. Bob Liftin, President, Regent Sound, NY.

Consumer Video Tape Duplication Techniques. David C. Cuyler, Vice President, Bell & Howell/Columbia Pictures Video Services, NY.

Consumer Video Disk Duplication Techniques. Takeo Yamamoto, Pioneer Electronic Corp., Tokyo, Japan.

5:00 PM Hospitality Dinner

BREAKOUT SESSION FOUR
Friday, May 16 7:30 PM, 8:00 PM, 8:30 PM

Satellite Up/Down Link Audio Demonstration

Cooridinator: Brian Homans, Shiner + Associates, Inc., IL.

This session will demonstrate the performance and features of audio signal processing electronics used to transmit and receive stereo audio information from point A to B via satellite.

Braodcast Television Transmitter-Receiver Link Demonstration

Coordinator: Rober Cochran, Tellabs, Inc., IL.

This session will feature Richard Carpenter of Broadcast Electronics and will cover the important performance parameters involved with this process with an emphasis on problems affecting signal quality.

Live Stereo Remote Demonstration

Coordinator: John Phelan, Shure Brothers Inc., IL.

This session will feature Patrick Bradbury of RF Technology Inc. and will center on a 13-GHz short haul transmission link as well as various techniques for generating live stereo audio in the field.

SESSION FIVE/PRODUCTION
Saturday, May 17 9:00 AM

Chairman: Bill Varney, Universal Studios, Universal City, CA.

This session will cover all major areas of stereo audio production for television and video.

Live Stereo Audio Production Techniques for Broadcast Television. Shawn Murphy, Production/Film Music Mixer, Disney Studeios, CA.

Stereo Audio Production Techniques Within the Television Plant. Ed Encona, Director, Film and Tape Post-production, NBC, CA.

Post-production Stereo Audio Techniques for Video Production. Philip Mendelson, Chief Engineer, Post Group, Los Angeles, CA.

Stereo Audio Synthesis Techniques for Broadcasting and Video Production. James Cunningham, President, Studio Tehcnologies, IL.

SESSION SIX/CONSUMER PRODUCT TECHNOLOGY
Saturday, May 17 1:30 PM

Chairman: Emil Torick, CBS Technology Center, CT.

This session will cover the principal consumer product categories that relate to the reproduction of stereo audio with broadcast television or video playback.

Broadcast Television Stereo Audio Receivers. J. James Gibson, RCA Laboratories, NJ.

Stereo Audio Characteristics of Consumer Video Tape Formats. Edward Foster, Diversified Science Laboratories, CT.

Stereo Audio Characteristics of Consumer Video Disk Formats. Cooridinator: Takeo Yamamoto.

Stereo Audio Signal Processing Electronics/Intergration of Stereo Audio Into the Home. Paul Jenrick, Manager, Consumer Product Development. Shure Brothers Inc., IL.

5:00 PM Hospitality 6:00 PM Dinner

BREAKOUT SESSION SEVEN/POST-PRODUCTION AUDIO EDITING DEMONSTRATION: Analog Techniques

Cooridinator: Norm Relich, Purdue University, IN.

This breakout session will feature a demonstration prepared by Douglas Ordon of AVC Systems Inc., IL.

7:30 PM, 8:00 PM, 8:30 PM

POST-PRODUCTION AUDIO EDITING DEMONSTRATION: Digital Techniques

Cooridinator: Norm Relich.

This session will feature a demonstration of a post-production digital recording and editing system.

CONSUMER STEREO AUDIO/VIDEO PLAYBACK DEMONSTRATION

Coordinator: John Bullock.

This session will feature a demonstration of a state-of-the-art audio/video home system reproducing stereo audio using broadcast and recorded audio and video signals.

SUNDAY, MAY 18

In addition to this breakout session, a home media room audio/video demonstration will run during all break and relaxation periods on Friday and Saturday. This demonstration will be presented in a smaller room recreating a realistic home environment.

SESSION EIGHT/ECONOMIC AND BUSINESS CONSIDERATIONS

10:00 AM

Chairman: Martin Polon.

This session will feature speakers who will cover five major areas of economic and business concern to the success and growth of stereo audio for television and video.

Economic Impact of Stereo Television and Video on the Motion Picture Industry. Jason Squire, Squire Companies, CA.

Sales and Marketing Impact of Stereo Television and Video on the Pro Audio Industry. Paul Gallo, Publisher, Pro Sound News, NY.

Marketing Considerations for Consumer Acceptance of Stereo Audio for Television and Video. Almon Clegg, General Manager, Matsushita Technology Center, NJ.

Recording Studio Economics for Stereo Television Audio Production. Murray Allen, President, Universal Recording Corporation, IL.

NOON CLOSING COMMENTS/ 12:30 PM brunch

Classified

FOR SALE

"THE PLATE"

2 Live Echo Plates at Quadraphonics 24-track studio. 2040 Beaver Ruin, Norcross, GA, 30021. (404) 263-7170. $2,500 each NEGOTIABLE!

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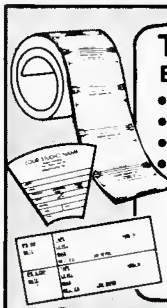
MCI 24-TRACK JH114, 12 DOLBY 361, 3M M64 2-TRACK, 3M M79 2/4-TRACK, SCULLY 280 2-TRACK, EVENTIDE H910 HARMONIZER. ALL EQUIPMENT PROFESSIONALLY MAINTAINED AND IN SERVICE IN MAJOR STUDIO. BROKER PARTICIPATION WELCOME. **CONTACT PAT SCHOLES AT (901) 725-0855, ARDENT STUDIOS, 2000 MADISON AVE., MEMPHIS, TN, 38104.**



YESTERDAY'S CRAFTSMANSHIP/PRICE—TOMORROW'S TECHNOLOGY! Inexpensive, professional quality Mixers, Amps, Effects! Unique Demonstration Tape/Literature-$4.00 (refundable)! **MCP/DaviSound, Box 521, Newberry, SC 29108.**

$29.95 TIMECODE generator software. Generate SMPTE timecode on your IBM PC. Software requires no modification of your PC. Send $29.95 plus $3.00 handling to: Kelly Quan Research, **55 White St.,San Francisco, CA 94109 (415) 771-6716.**

CLOSING JUNE 1986. Will have for sale Ampex 440Cs, Ampex 3200 Duplicator (1 master, 5 slaves), Dolby A & B units, bulk erasers, shrink tunnel, etc. Would appreciate indication of interest. **Barclay (914) 454-0068.**

CROWN, MICROTECH, Nakamichi, Electro-Voice, Professional Services. BEST PRICES. AmeriSound Sales, Inc. East Coast: **(904) 262-4000**; West Coast: **(818) 243-1168.**

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CUTTERHEAD REPAIR SERVICE for Westrex, Haeco and Grampian. Convert your old Westrex to model 3DIIH. New and rebuilt cutterheads available. **International Cutterhead, 194 Kings Court, Teaneck, NJ 07666. (201) 833-4421.**

FINISH UP ON TIME WITHOUT SACRIFICING QUALITY.

You want it quick and you want it good. In today's competitive post-production audio/visual scene, the rewards go to those who can produce results that are quick *and* good. That's why TASCAM designed the MS-16 1" 16-track recorder—to bring together top-notch audio quality plus premium features that streamline production and move you ahead of schedule.

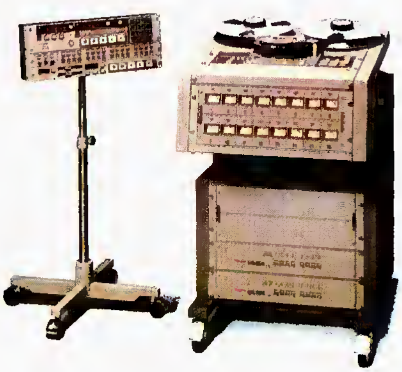
Quality reproduction starts with the heads, and TASCAM has three decades of design experience behind the MS-16's new micro-radii heads. They bring "head bumps" under control and ensure flat frequency response. And unlike most tape machines, the MS-16 record/sync and playback heads are identical in performance. Because sync response equals repro response on the MS-16, you can make critical EQ and processing decisions on overdubs or punch-ins without having to go back and listen a second time. You get what you want sooner and with fewer headaches.

The MS-16 cuts down on the time you spend locking up with other audio and video machines as well. A 38-pin standard SMPTE/EBU interface affords speedy, single-cable connection with most popular synchronizers and editing systems. It's the easy, efficient way to get the most out of today's sophisticated synchronization equipment. The MS-16's new Omega Drive transport is tough enough to stand up to long days of constant shuttling...while handling tapes with the kid-glove kindness they deserve.

Record/Function switches for each track allow effortless, one-button punch-ins. Input Enable allows instant talkback during rewinds, fast forwards and cue searches. These features speed you through sessions and let you concentrate on the project at hand...not on your tape machine.

Take a closer look at the MS-16. See your TASCAM dealer for a demo or write us for more information at 7733 Telegraph Road Montebello, CA 90640.

THE TASCAM MS-16 SIXTEEN TRACK



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