JANUARY/FEBRUARY 1987

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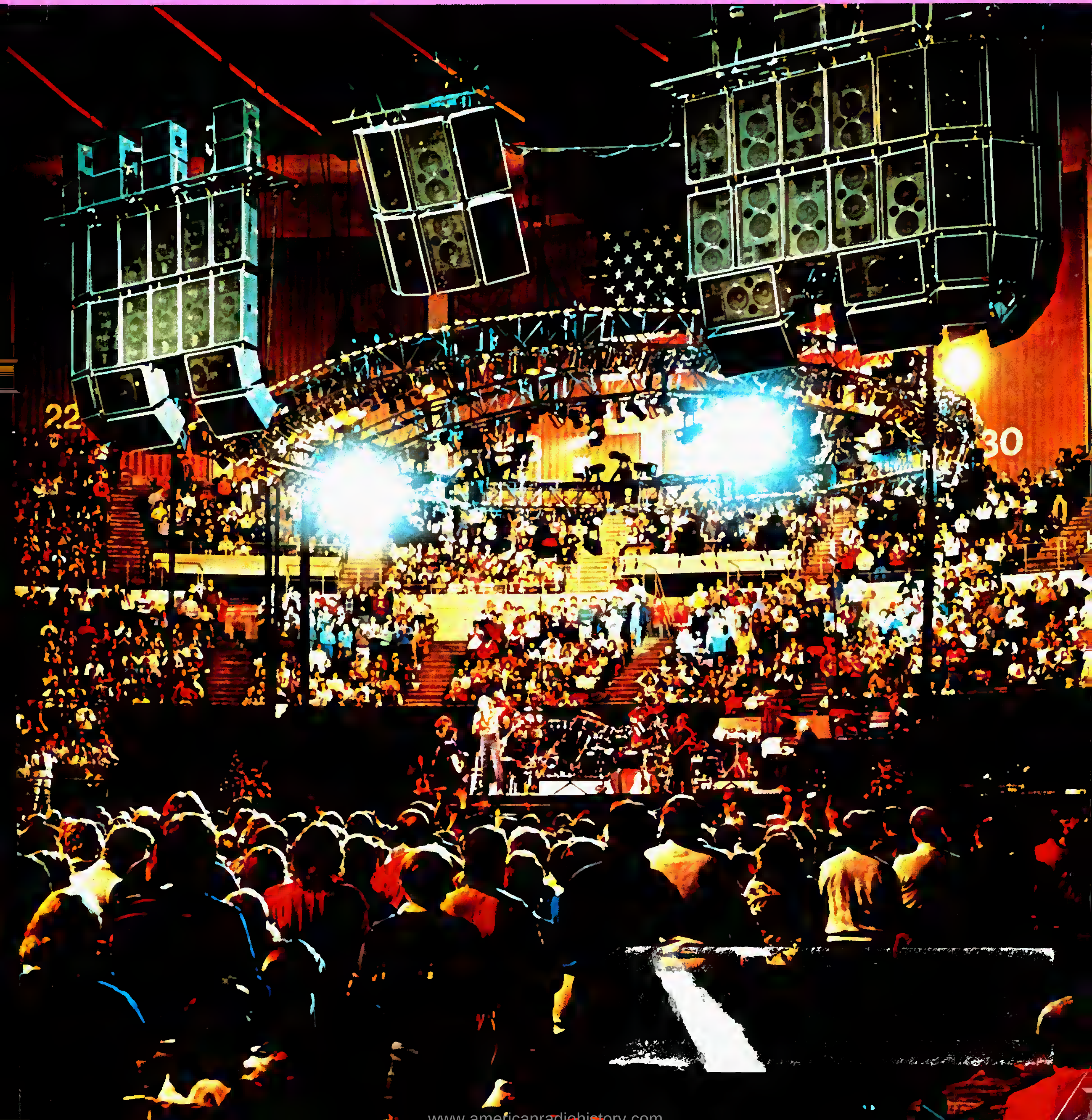
Featuring 2 to 8 trk-The Smaller Recording Studio
Guide: Studio and Performance Monitors



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JANUARY/FEBRUARY 1987 VOLUME 21 NO.1

the sound contracting engineer

the recording engineer

2 to 8 the smaller recording studio

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About the Cover

The cover is of a Neil Diamond concert that took place at the St.Paul (Minnesota) Civic Center in December, 1985. JBL Concert Series Speakers were used by Stanal Sound, as they were also used at L.A.'s Greek Theater. See our story in this issue.

About the 2 to 8trk Cover

Artists featured in the story on home recording studios. Clockwise from top right: Rick Derringer, Nancy Wilson, Leslie Howe, David T. Chastain, and Howard Leese. (Photo of Leslie Howe by Janusz Kawa. All other photos by Cheryl Lynne.)

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Letters

Dear Editor,
In re-reading Bruce Bartlett's article regarding the AES Workshop, "Recording Pop Music On Location" in the Jan/Feb, 1986 issue of db, there appears to be an error in the section "Interfacing with Telephone Lines." The article advises readers to "order program lines two or three days in advance." If anyone were to follow this advice, they might find themselves all dressed up with nowhere to go, come program time. Based on my years of experience in dealing with the phone company with both equalized and non-equalized circuits, you should place your order for an equalized line twenty to thirty days in advance. It should also be kept in mind that the telephone company does not have equalized lines available everywhere. If one were to do an event from a particular location for the first time, it would be best to contact the telephone company before finalizing anything. This way you will be certain the "unloaded" pairs are available for an equalized circuit for that location.
Sincerely,
Dan Kelley
Station Manager
KISZ AM and FM,
Cortez, Colorado.

Thanks for pointing this out to us and all our readers, Dan. Since the divestiture of AT&T, small companies have sprung up everywhere. The amount of time it takes to secure lines for remote broadcasts probably varies from as little as one day to as much as three months. But as you say, order early...

Dear Editor,
I am a former reader of **Modern Recording & Music**, and now a db reader. I have very high praise for your magazine, especially your columns and articles. They are very informative and enjoyable to read. I am a musician and am planning to go for sound reinforcement. I want to build my own cabinets to save money. My problem is that I don't have enough knowledge of cabinet design. Can you provide me with information on design and construction of cabinets?
Jeff P. Santos,
Yigo, Guam

Well Jeff, if you refer back to the July/August 1986 issue of **db Magazine**, *you will find an article entitled "Vented Loudspeaker Enclosure Design Made Easy." It includes a section on the most commonly asked questions, a plethora of formulas, an overflow of*

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flowcharts, and a bottleneck of a bibliography of recommended references. Please note: If you refer to the program on page 48 of the article, it is incorrect. A corrected reprint of the program, along with an additional one, appears in the "Letters" section of the September/October 1986 issue.

Our recent mail contains information on a company called Audio Systems Design who has released version 2.1 of its graphics based speaker cabinet design program. The program runs on a regular IBM PC or compatible, costs $29.95, and is available from Audio Systems Design, 1755 Jenkee Drive, Florissant, MO 63031. We suggest you write to them directly for more information.

Good luck, and happy building.

Author Bruce Bartlett writes:

I'd like to correct three errors I made in my articles in the November/December issue.

On page 12, under the *Sequencing* heading, 256 Hz should be 261.63 Hz.

On page 14, the rate of MIDI information transfer is 3.125 kilobaud (31,250 bits/sec or 3,125 bytes/sec, not 31.25 kbaud).

On page 39, under *Stereo Localization*, I said that if you turn down the right channel, the image shifts toward the right speaker. It should say, if you turn UP the right channel, the image shifts toward the right speaker. *Figure 3* shows the correct relationship.

CALENDAR 1987

Synergetic Audio Concepts (SynAudCon) is sponsoring a series of two-day seminars dealing with solving audio and acoustic problems.

Demonstrations will include signal alignment, measurements of %ALcons and RASTI, the fundamental differences between impulse and energy time curve measurements, and how to design loudspeaker arrays.

For more information, contact: Synergetic Audio Concepts, PO Box 669, San Juan Capistrano, CA 92693. Tel: 714-728-0245.

Studio City, CA–March 3-4, 1987
Sportsmen's Lodge

New Orleans, LA–April 2-3, 1987
Sheraton Inn-Airport

Dallas, TX–April 14-15, 1987
The Summit Hotel

Atlanta, GA–April 22-23, 1987
The Presidential Hotel

The UCLA Extension Department of the Arts will offer a wide variety of classes of special interest to recording engineers during the winter quarter, including several required and elective courses in the Professional Designation in Recording Engineering program. Some of the required courses will include "Introduction to Audio Engineering," taught by engineer Michael Braunstein. Class will meet on Mondays, in room 1344 Scoenberg, UCLA, 7-10 p.m., for a fee of $205.00.

Los Angeles, CA–January 5-March 16.

"Recording Engineering Practice I," meeting on Wednesdays, January 7-March 25, will be instructed by Van Webster, owner of Digital Sound Recording (609 N. Avenue 64, Los Angeles), where the class will be held. Class will meet from 7 to 11 p.m., for a fee of $875.00.

Los Angeles, CA–January 7-March 25.

Electives will include "Record Production I," taught by producer Jeffrey Weber. Supplemented by guest speakers and field trips, this class will analyze the production process. Class will meet in romm 1344 Schoenberg, UCLA, 7-10 p.m., for a fee of $225.00.

Los Angeles, CA–January 7-March 25.

"Multitrack Editing for Audio Recording," taught by engineer Sye Mitchell, will offer 12 students the opportunity to edit multitrack, 8, 16 and 24-track tape at Studio Masters Studios. Class will meet in room 1151 Schoenberg, UCLA, 7-10 p.m., on the 30th, and will meet at Studio Masters, 10 a.m. to 6 p.m. on the 31st and 1st. The fee will be $140.00.

Los Angeles, CA–January 30-February 1.

For details about these classes or the Professional Designation program, contact UCLA Extension, (213) 825-9064.

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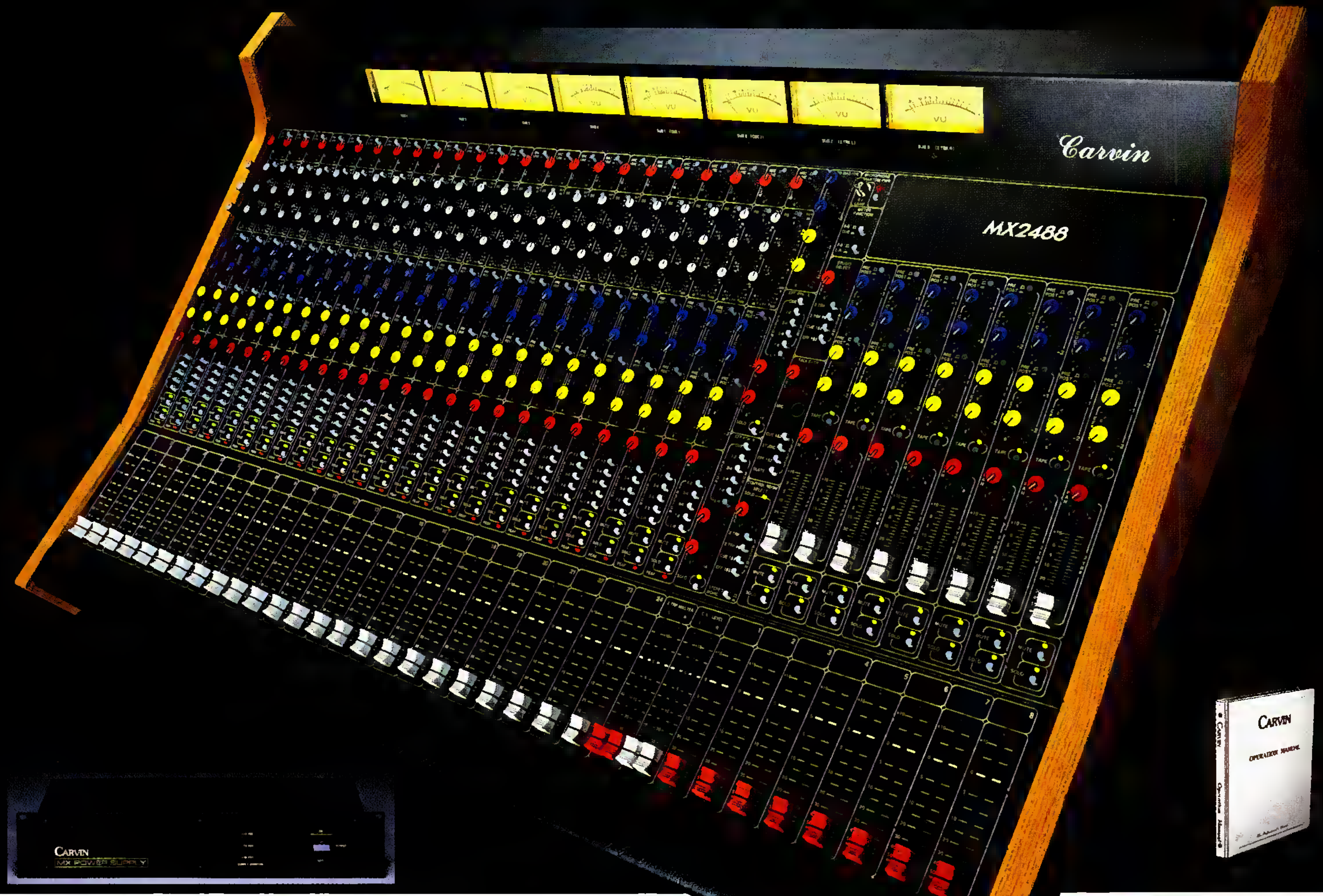

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Len Feldman—db magazine
September/October—1986

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Bruce Bartlett

Recording Techniques

Microphone-Technique Basics

Microphone selection and placement (mic technique) greatly affects the sound of a recording. Even if your tape recorder and mixer or PA system are the best available, the final result will be poor unless you choose and place your microphones carefully.

Instead of suggesting techniques for individual instruments, this article covers some fundamental guidelines to follow in any mic'ing situation. These are physical principles to keep in mind when using microphones, rather than artistic goals.

MICROPHONE SELECTION

How do you choose an appropriate microphone for an instrument or musical ensemble? Is there a correct microphone to use in each application? No. Every microphone sounds different, and you choose the microphone that gives you the best sound. Still, there are some guidelines that apply in most situations. To summarize these, I'll briefly describe how to choose a microphone based on its frequency response and polar pattern.

The frequency response of a mi-

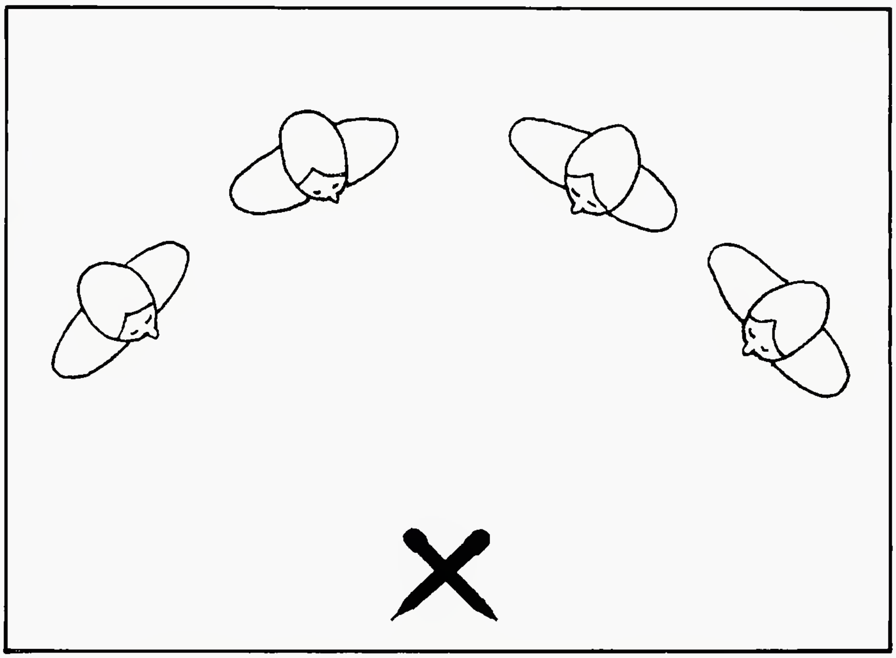

Figure 1. Overall mic'ing of a musical ensemble with two distant microphones.

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crophone affects the reproduced tone quality. Flat-response mics tend to sound natural; mics with emphasized high-frequency response sound brighter or more trebly. Microphones that roll off below the range of the instrument minimize room rumble; mics that roll off low frequencies within the range of the instrument tend to sound weak in the bass.

Most condenser microphones have an extended high-frequency response, making them suitable for cymbals or other instruments requiring a detailed sound, such as acoustic guitar, strings, piano, and voice. Dynamic mics have a response adequate for drums, guitar amps, horns, and woodwinds.

The polar pattern of a microphone affects the amount of leakage and ambience (room reflections) that are picked up. Omnidirectional microphones pick up the most ambience and leakage; unidirectional microphones pick up less.

The more ambience that is recorded along with an instrument, and the more an instrument's leakage is recorded by other mics, the more distant that instrument sounds. For example, a drum set recorded with a lot of ambience sounds distant. Also, if the sound of the drum set leaks into distant microphones, the drum set will sound even more distant.

Because of their greater pickup of ambience and leakage, omni mics sound more distant than unidirectional mics when both are placed the same distance from an instrument. Stated another way, an omni mic must be placed closer to an instrument than a uni mic to reproduce the same sense of distance.

Omni mics have some advantages over uni mics that may influence your choice: Omni mics tend to have less handling noise and breath popping than unidirectional mics. In addition, the up close bass-boost of many unidirectional mics does not occur with omni mics. This bass boost, however, may be a desirable effect with tom-toms or stage vocals.

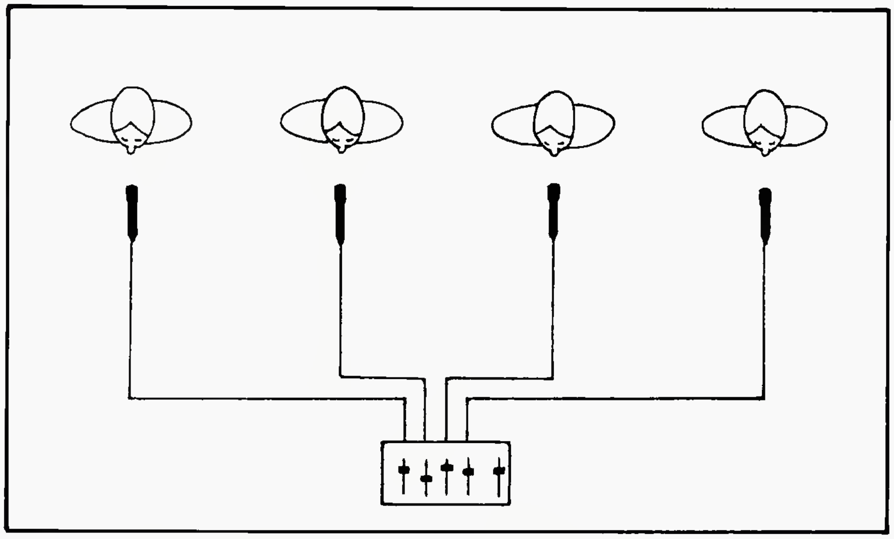

Figure 2. Individual mic'ing with multiple close-up microphones and a mixer.

MICROPHONE PLACEMENT

After choosing a suitable microphone, you place it in a spot that gives the desired tonal balance and ambience pickup.

Before placing microphones for a recording session, there are at least three questions to ask: How many microphones should be used? How far from the sound source should each microphone be placed? When mic'ing an instrument up close, what part of the instrument should be mic'ed? Let's answer each question in detail.

QUANTITY

The number of microphones required varies with the recording situation. Use just two microphones (or a stereo microphone) when you want to record an overall acoustic blend of the instruments and room ambience (*Figure 1*). Many ensembles can be recorded quite well this way: classical-music ensembles (such as orchestras, marching bands, choirs, string quartets, and pipe organs), small folk groups, and vocal quartets.

Pop-music groups, on the other hand, usually are recorded with multiple microphones–one or more for each instrument or instrumental section. Mic'ing every instrument lets you control the balance (relative loudness) among instruments at your mixer by adjusting the volume control for each microphone (*Figure 2*).

For the greatest clarity in a multi-mic recording, use as few microphones as are necessary to get a good sound. Don't use two microphones when one will do the job. To achieve this, sometimes you can cover two or more sound sources with a single microphone (*Figure 3*). A brass section of four players can be covered with just one microphone on four players, or with one microphone on every two players.

There's a disadvantage in picking up several instruments with

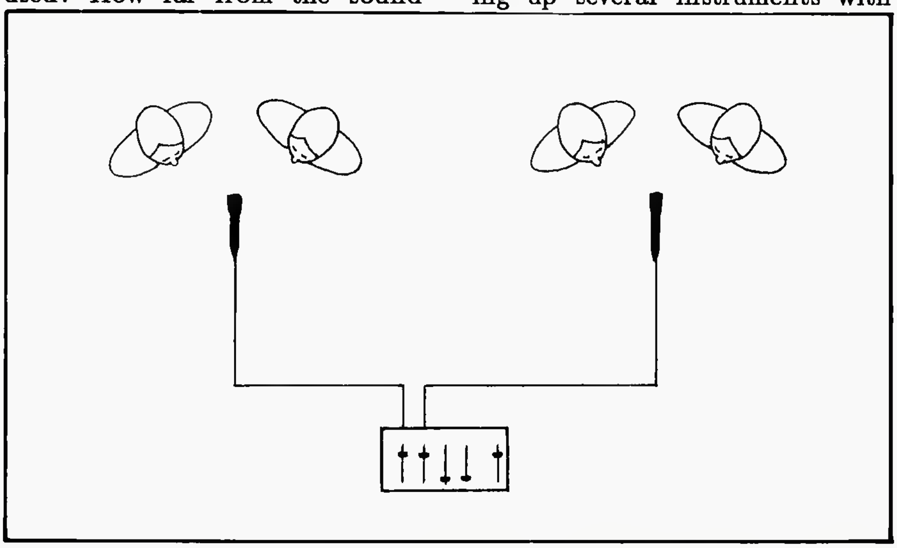

Figure 3. Multiple mic'ing with several sources on each microphone.

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one microphone: You can't adjust the balance among instruments after recording (during mixdown). The instruments have to be balanced acoustically in the studio for a proper blend before making the recording. Overly-quiet instruments should be moved closer to the mic, and *vice-versa.*

What's more, if you pick up several instruments equally with one microphone, you must place the mic relatively far from the instruments. This distant microphone may pick up too much room ambience and leakage from other instruments.

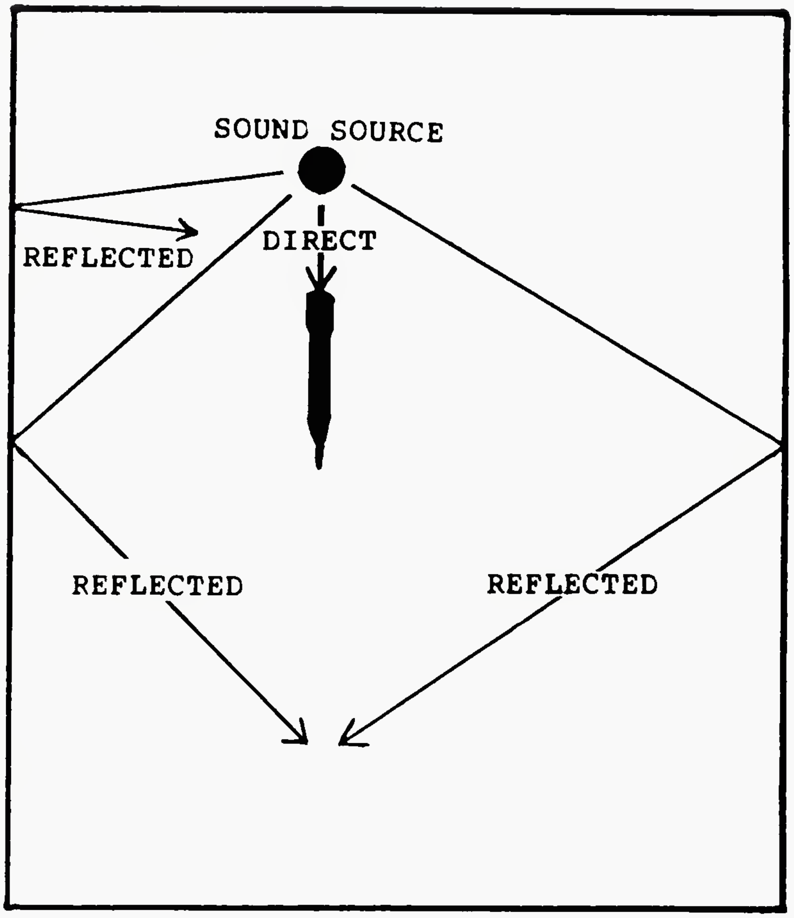

Figure 4. A close microphone picking up mainly direct sound results in a close sound quality.

MIC'ING DISTANCE

The closer a microphone is placed to a sound source, the less ambience, leakage, and background noises are picked up. In other words, placing a microphone a few inches from an instrument both reduces the recorded ambience or room reverberation, and rejects the sounds of other instruments leaking into that microphone, and also discriminates against noises entering the recording room. In sound-reinforcement situations, close microphone placement also permits maximum loudness before feedback occurs.

Here's why: The levels of reverberation, leakage, and background noise in a studio are fairly constant with microphone position. But the closer you place a microphone to a sound source, the louder that source is at the microphone. Consequently, close mic placement picks up a high ratio of desired signal (the instrument) to undesired signal (ambience, leakage, and noise), (*Figure 4*).

Conversely, the farther a microphone is placed from a sound source, the more room ambience is picked up. The sound coming directly from an instrument (direct sound) gets quieter as the distance from the instrument increases. But the loudness of the ambience is fairly constant throughout the room. So, a distantly placed microphone picks up a high ratio of ambience to direct sound (*Figure 5*). That is, it picks up a lot of room sound. It also picks up a lot of leakage and background noise (*Figure 6*).

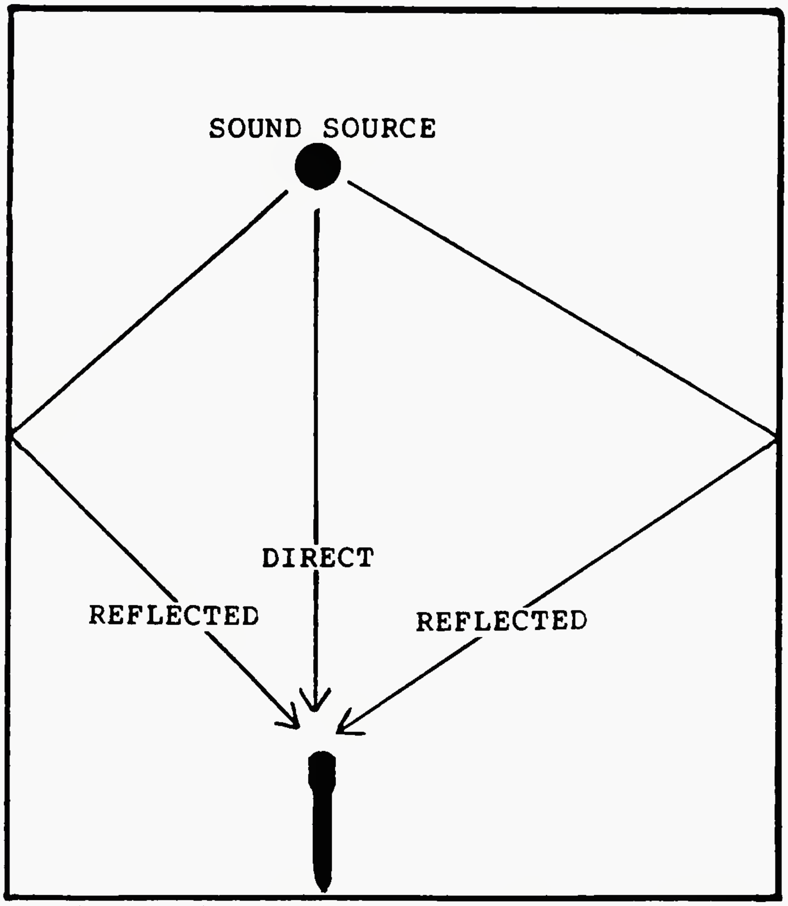

Figure 5. A distant microphone picks up mainly reflected sound, resulting in a "distant" sound quality.

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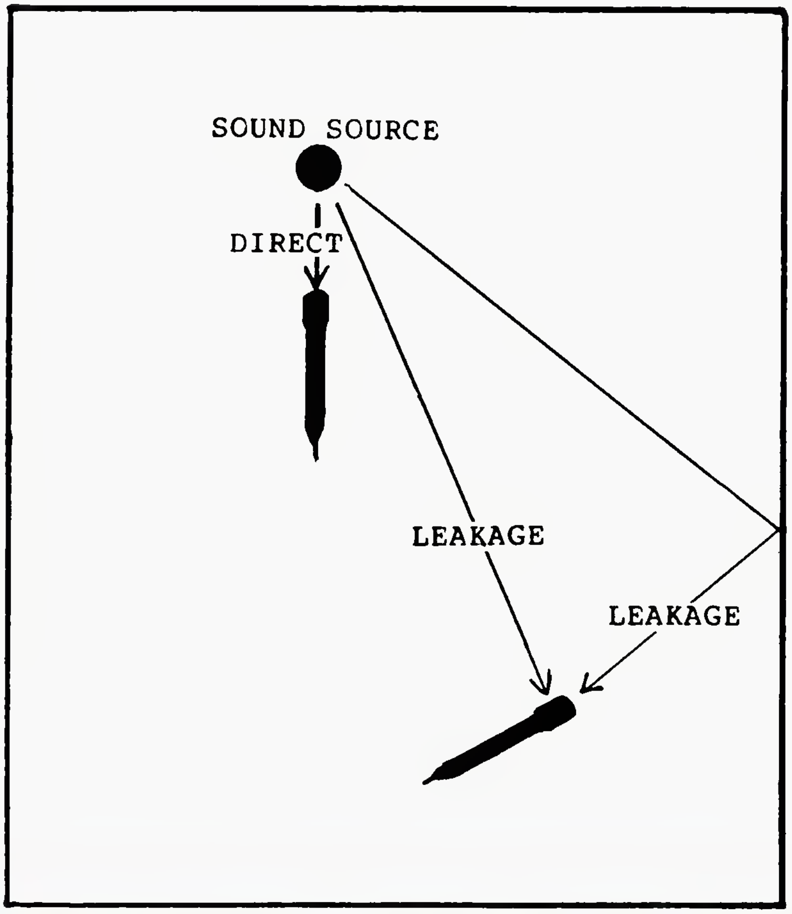


Figure 6. A distant microphone may pick up leakage, also resulting in a distant sound quality.

By mic'ing each instrument or vocalist at a different distance, you can create a sense of depth in the recording. Close-mic'ed instruments sound close to the listener (that is, they have presence), and distant-mic'ed instruments sound far away.

Distant mic placement adds a "live, loose" feel to a recording. This technique often is used when overdubbing strings and horns, and sometimes is applied to overdubbed vocals, electric-guitar solos, drums, and pianos. Classical music always is recorded at a distance because concert-hall reverberation is a desirable part of the sound.

An ambience microphone is a microphone placed about 10 feet or more from an instrument to pick up ambience. The microphone is usually omnidirectional. Its output is mixed with the usual close-placed microphones, adding an airy or spacious feeling to the sound of the instrument being recorded. Two are often used for stereo. In live concert recording, ambience microphones placed over the audience pick up audience reaction and concert-hall acoustics.

TONAL EFFECTS OF CLOSE MICROPHONE PLACEMENT

Although close mic'ing has several benefits, you should place the microphones only as close as necessary, not as close as possible. Mic'ing too close can color the recorded tone quality of an instrument. Why does this occur?

Most instruments are designed to sound best at a distance (say, 1 1/2 or more feet away). So, a flat-response microphone placed there tends to pick up a natural or well-balanced timbre (tone quality). But when leakage or poor room acoustics force you to mic in close, you emphasize the part of the instrument that the microphone is near. The tone quality picked up very close may not reflect the tone quality of the entire instrument.

For example, the sound hole of an acoustic guitar resonates strongly around 80 to 100 Hz. A microphone placed close to the sound hole "hears" and emphasizes this low-frequency resonance, producing a bassy, boomy recorded timbre that does not exist at a greater mic'ing distance. To make the guitar sound more natural when miked close to the sound hole, you need to roll-off

the excess bass on your mixer, or use a microphone with a bass roll-off in its frequency response.

In general, close mic'ing may give a tonal imbalance, which you can partially correct with equalization or careful microphone selection and placement.

A natural tonal balance usually can be found at a mic'ing distance equal to the size of the instrument. If the situation allows, place the mic as far from the instrument as the instrument is big. That way, the mic picks up all the sound-radiating parts of the instrument about equally. For example, if the body of an acoustic guitar is 18-in. long, place the mic 18-in. away for a natural tonal balance.

USING MIC PLACEMENT FOR TONE CONTROL

Musical instruments radiate different tone qualities in different directions, and they produce different tone qualities from different parts of the instrument. Thus, you can partly control the recorded tone quality simply by changing the micro-phone position relative to the instrument. (*Figure 7*).

For example, a trumpet radiates strong highs directly out of the bell, but does not project them to the sides. Thus, a recorded trumpet sounds bright when mic'ed on-axis to the bell and sounds more natural or mellow when mic'ed off to one side. Similarly, the recorded timbre of a piano varies widely depending on microphone position. The sound of a piano mic'ed one foot over the middle strings is fairly natural; under the soundboard is bassy and dull, and in a sound hole is constricted.

It pays to experiment with all sorts of microphone positions until you find a sound you like. There is no one right way to place the microphones because you place them to achieve your desired tonal balance.

To determine a good starting microphone position, try closing one ear with your finger, listen to the instrument with the other ear and move around until you find a spot that sounds good. Put the microphone there, then make a recording and see if it sounds the same as what you heard live.

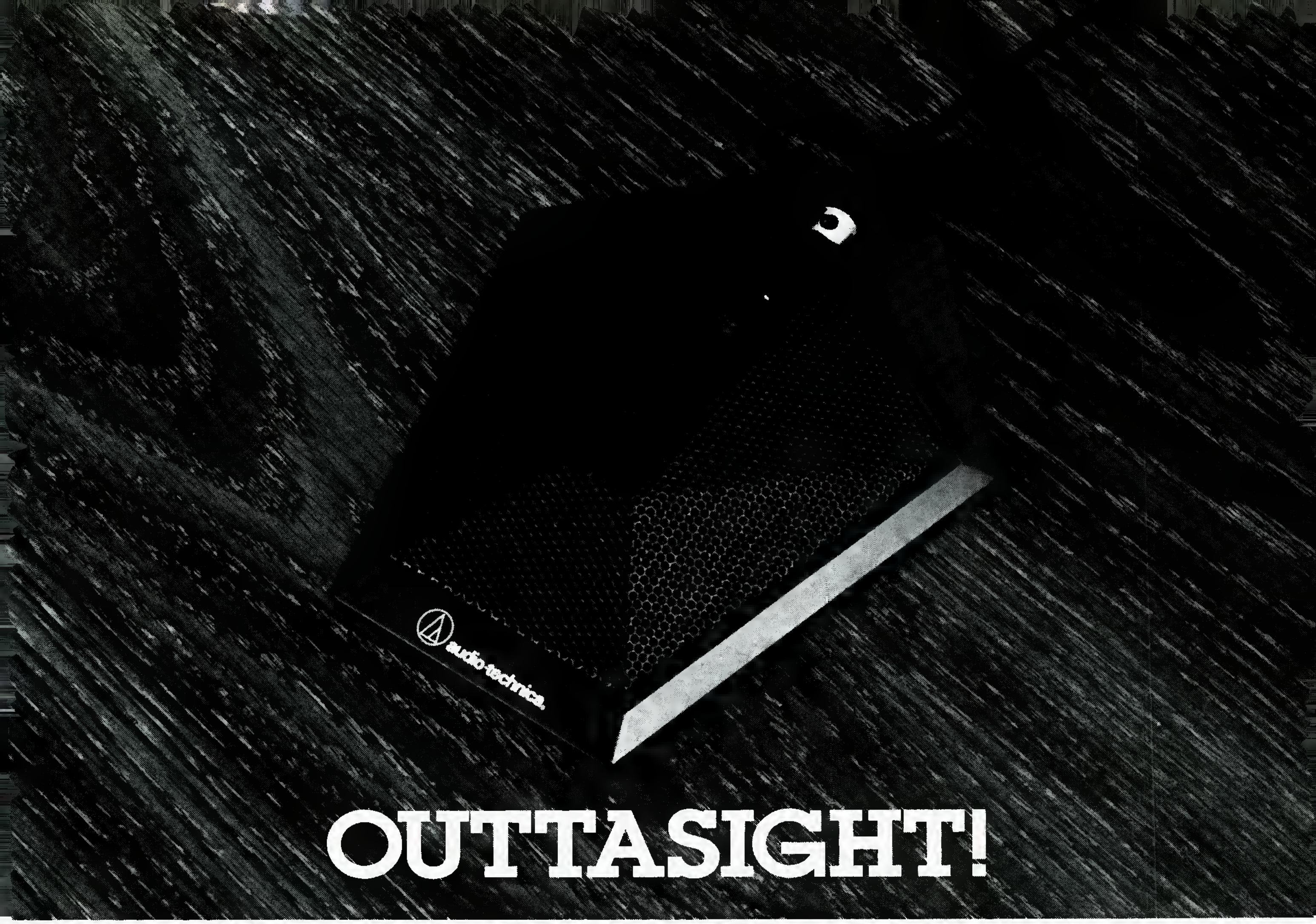
ON-SURFACE TECHNIQUES

Sometimes you're forced to place microphones near hard reflecting surfaces. Applications where this might occur are recording drama or opera with the microphones near the stage floor, recording an instrument surrounded by reflective baffles, or recording a piano with the microphone close to the lid. An unnatural, filtered tone quality can result. Here's why:

In these situations, sound travels to the microphone via two paths: directly from the sound source, and reflected off the nearby surface. Because of its longer travel path, the reflected sound is delayed relative to the direct sound. The direct and delayed sound waves combine at the microphone, resulting in phase cancellations of various frequencies. (*Figure 8*).

There is created a series of peaks and dips in the net frequency response called a comb-filter effect. The recorded tone quality in this case can be quite colored; it is similar to that achieved by phasing or flanging.

To avoid the tonal coloration caused by microphone placement near a surface, boundary



Until today, your choice of hemi-cardioid boundary microphones was limited. You could either choose thin sound to get articulation, or mid-range sound without the extended highs and lows needed for demanding sound reinforcement.

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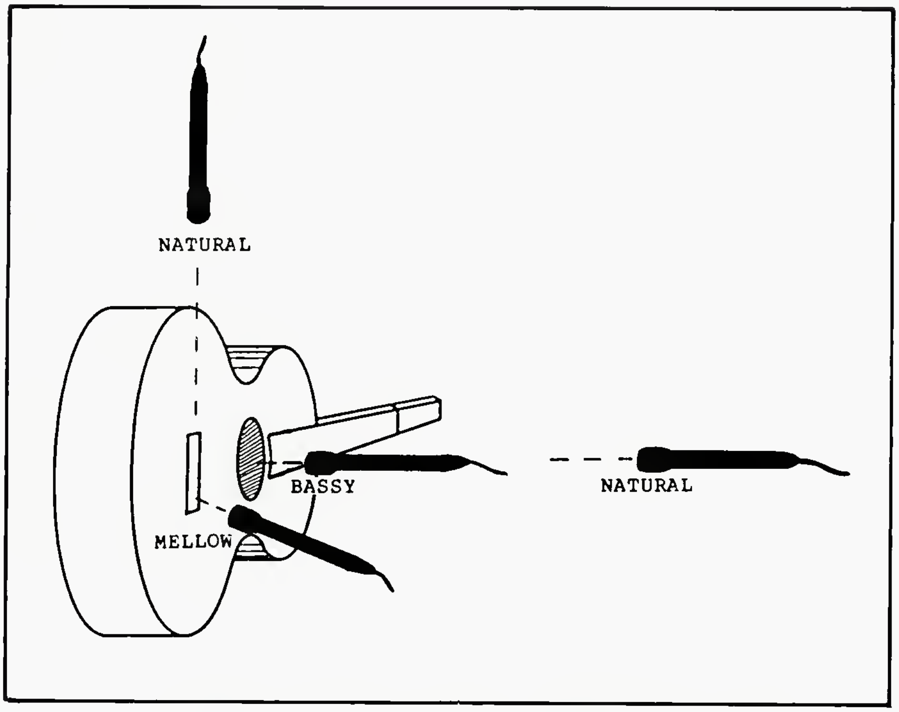


Figure 7. Microphone placement affects the recorded tonal balance.

Some microphones have off-axis coloration: A dull or colored tone quality for sources that are not directly in front of the microphone. Try to keep sound sources as on-axis as possible, especially sources that radiate strong high frequencies. For wide-angle sound sources, use microphones that have uniform response over a broad range of angles. Boundary microphones and miniature microphones have almost no off-axis coloration.

SUMMARY

Microphone techniques affect mainly two aspects of recorded sound: tonal balance and sense of distance. The frequency response and placement of a microphone affect the recorded tonal balance. The polar pattern and mic'ing distance affect the recorded sense of distance.

microphones (such as the Crown Pressure Zone Microphone series) have been specially designed for on-surface mounting. They are constructed with the microphone diaphragm very close to the reflecting surface so that there is no delay in the reflected sound. Direct and reflected sounds combine in-phase over the audible range of frequencies, resulting in a flat response.(*Figure. 9*).

Omni boundary mics are often taped to the underside of a piano lid, to a hard-surfaced panel, or to walls for ambience pickup. Uni-directional boundary mics are commonly used on stage floors near the footlights, on lecterns, or on news desks.

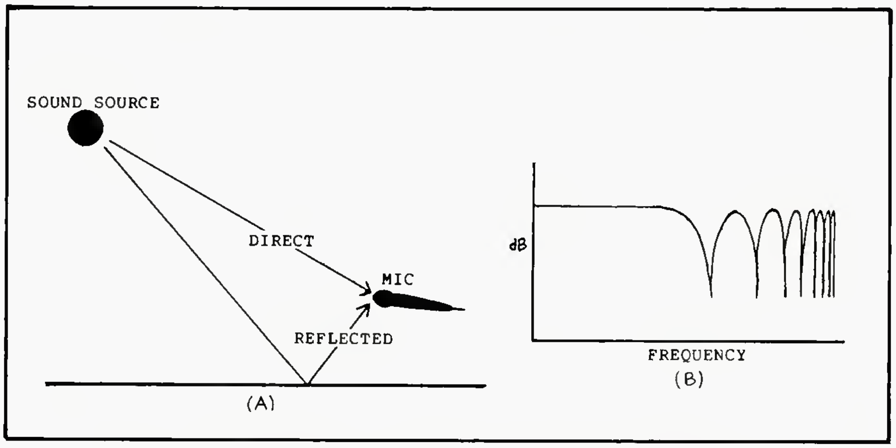


Figure 8. A microphone placed near a surface picks up direct sound and delayed reflection (A). The resulting phase interference gives a "comb filter" frequency response.

ADDITIONAL TIPS

When multiple microphones are mixed to one channel, the distance between microphones should be at least three times the mic-to-source distance. This is called the 3:1 rule (*Figure 10*). For example, if two microphones are each placed one foot from their sound sources, the microphones should be at least three feet apart. Following this rule will prevent phase cancellations and the resulting blurred, colored sound quality.

If the 3:1 rule is violated (many mics at a distance), clarity is reduced. If the 3:1 rule is followed (fewer mics at a distance, or many mics up close), clarity is enhanced.

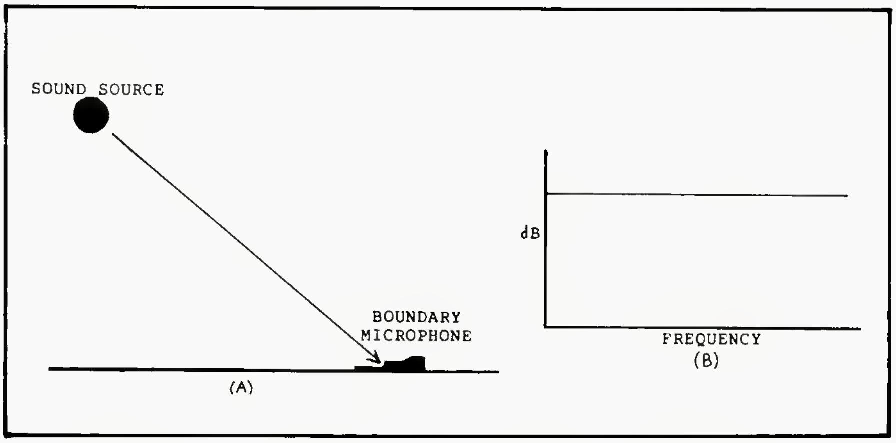


Figure 9.(A) A boundary mic on a surface picks up direct and reflected sound waves in phase. (B) The resulting lack of phase interference gives a flat response.

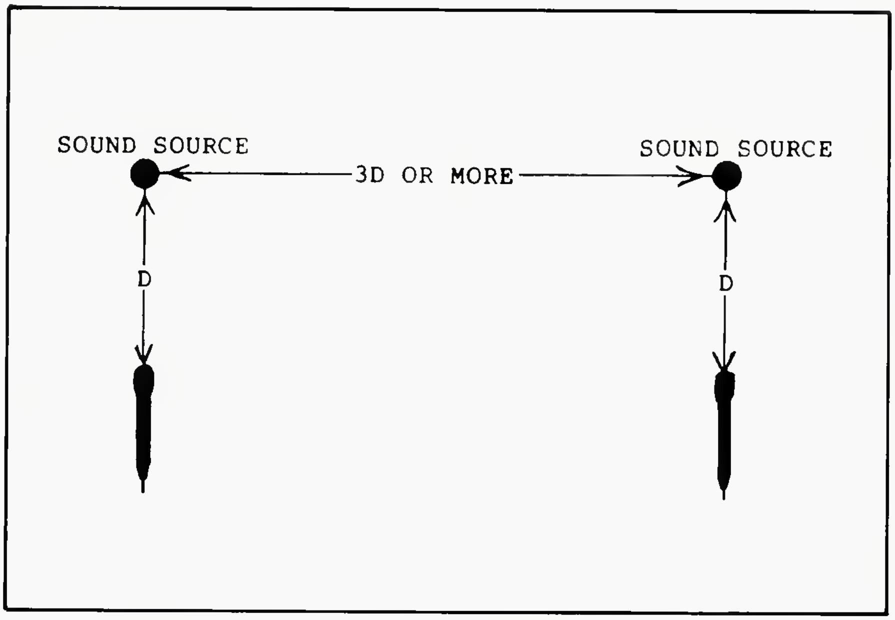
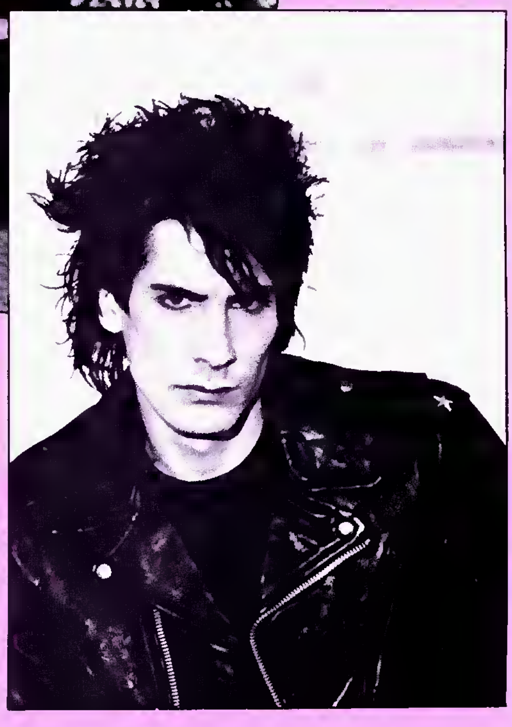


Figure 10. The 3:1 rule of microphone placement insures that you avoid phase interference between mic signals.

Follow the 3:1 rule to prevent phase cancellations between microphones, and use boundary microphones to prevent phase cancellations from surface reflections. Be aware of off-axis coloration when selecting and placing microphones. If you take advantage of these physical principles, they will become a tool to help you achieve the desired production style. ■

2 to 5

rk



Sammy Caine

Home Studios of Recording Artists

Technical Editor Sammy Caine visited several musician/studio operators and came away with a wide spectrum of usage.

HOME STUDIOS COME IN A WIDE RANGE OF SHAPES AND sizes. The type of equipment employed in a home studio can be a pair of stereo cassette decks or a pair of Sony PCM-3324 digital tape machines with a Neve console.

Since the average home studio owner cannot comfortably afford state-of-the-art recording equipment, we sometimes feel we must make do with our cassette multitrackers, with perhaps only a small reverb or antique echo device. We feel we must leave professional recording to the recording artists. But what do recording artists do when they want to record a song without the involvement of a producer and a professional recording studio? They use their home studios, of course.

I spoke to a number of professional recording artists to find out exactly what equipment they have in their home studios.

RICK DERRINGER

First, I visited the New York City apartment of producer/artist Rick Derringer. In his small "studio room," Rick has a 4-track recording system. At the center of the system is a Vesta Fire MR-10 4-track cassette recorder/mixer and JBL L11T and L112 monitors. He primarily uses the studio for song writing and recording demos. For mixdown purposes, Rick uses a Sony PCM-F1 digital converter with a Hitachi stereo hi-fi VCR, so in effect, he ends up with a 2-track digital master. In addition, Rick carries the digital set-up to the studio when he is working on a project so he can make high quality cassette copies to listen to at home.

In his home studio, Rick doesn't utilize extensive outboard equipment. He has a Yamaha SPX-90 effects processor, and an Alesis MIDI-Verb reverb unit.

Since he is primarily a guitarist, Rick naturally has

Rick Derringer with his Sony PCM-F1 and Hitachi Hi-Fi VCR used for digital 2-track recording.

Rick Derringer's home studio room.

a full complement of guitars and amps. His guitar collection includes a BC Rich Stealth, which he designed, a Steinberger, a 1958 Gibson Les Paul, and an old Rickenbacker. The pick-ups in these guitars are DiMarzio and Seymour Duncan. Guitar amps used are Peaveys, Seymour Duncan Convertibles and a Scholz Rockman. For drums, Rick uses a Roland TR-505 rhythm composer.

Soundproofing the room was not a concern of Rick's, since he close-mics with a Sony lavalier microphone. For vocals, he uses a Shure SM-57. This helps keep extraneous sounds off the tape, but the lack of soundproofing allows sound to escape from the room. In fact, the front wall of the studio is actually the front of the building, so anyone walking past the building, can possibly hear the next hit Rick is working on.

MICHAEL McDONALD

Rick's small and functional home recording studio is only one approach. At the opposite extreme is the home studio of former lead singer of the Doobie Brothers, Michael McDonald. Michael is currently involved in a solo recording career.

In Michael's California home, he has a full 24-track state-of-the-art recording studio. At the center of the studio is an API 36 x 16 console and a 3M-79 24-track tape machine. In addition, the remainder of Michael's equipment list reads as that of any major commercial recording facility.

McDonald is heavily into studio computer systems. He uses a Yamaha QX-1 computer sequencer and an Apple Macintosh computer. Software used in the studio includes Opcode's DX Librarian and DX Editor, and Mark of the Unicorn's Performer. He also uses a Roland SBX-80 sync box.

His pro mics are various AKGs, Sennheisers, Shures, and Neumanns. His studio's extensive list of outboard signal processing gear includes Lexicon and Yamaha delays and reverbs, an Eventide 949 Harmonizer and a full array of gates, compressors, limiters, and equalizers. For monitors, Michael has three systems: Meyer Sound 833s, Yamaha NS10s, and Mitsubishi DSs.

Of course, this fully equipped studio wouldn't be complete without a wide range of synthesizers. Among them are Yamaha DXs and TXs, LinnDrum machines, and a Roland Super Jupiter. (Full equipment lists appears at the end of this article.)

LESLIE HOWE

Another artist that feels that the large state-of-the-art studio is the way to go is Leslie Howe, one half of the Canadian group One to One. He is currently upgrading his home studio from an 8-track facility to a full 32-track digital studio.

During the recording of One to One's debut album, *Forward Your Emotions*, on Warner Brother's Records, Leslie used the 8-track studio for writing and recording all its demos. In addition, the band recorded and mixed one song from the LP, "Angel In My Pocket," in the studio. Leslie is also a producer and engineer, and that is no doubt why the song had high enough quality to be included on the LP. At the time of this recording, the studio, located in Leslie's home in Ottawa, Canada, had a 16 x 2

Series II Soundcraft mixer and an MCI JH110-A 1-inch, 8-track tape machine. His monitors, which are being replaced by UREI 811C time aligned monitors, were JBL 4315B 4-ways.

Since he uses extensive MIDI-equipped synthesizers, Leslie utilizes a Sycologic M-16 16 x 16 MIDI matrix for enhanced MIDI control. The matrix is used with its remote keypad to control and display the MIDI configuration being used. It allows up to 32 matrix patches to be edited, stored and recalled.

Most of his current equipment will be retained for use in the new digital studio system. The studio will include the recently purchased Mitsubishi Westar 28-input automated console and a Mitsubishi X-800 32-track digital tape machine. In addition, Leslie has ordered a wide range of outboard and processing gear to add to his current equipment.

DAVID T. CHASTAIN

Leslie's type of studio obviously takes up a large amount of room. One artist that doesn't have all the extra room for a large studio is David T. Chastain, the lead guitarist for the hard rock group CJSS.

David cordoned off and soundproofed an upstairs bedroom of his Cincinnati, Ohio home for the control room, and soundproofed his basement to use as the studio. He uses the studio for writing and making what he calls "very good quality demos."

At the heart of the studio is a Yamaha 1602 16-channel mixer and a Tascam 38 8-track tape machine. Although the mixer has 16 inputs, David did feel the need for a bigger one.

David uses a Teac Model 1 mixer for his pair of monitor systems which are JBL 4311Bs and Auratones. He has a small selection of various reverbs and digital delays which include a Yamaha REV-7 and a Lexicon Prime Time digital delay. He also has an Eventide H910 Harmonizer.

His small, but varied microphone collection, includes Sennheisser 441s and 421s, AKG D-12Es and Shure SM-57s and SM-58s.

NANCY WILSON AND HOWARD LEESE

Nancy Wilson and Howard Leese, of the rock group Heart, also have home 8-track studios. As the other artists, their studios are used primarily for writing and recording demos.

In Nancy's studio, there is a Tascam Studio 8 mixer and an Otari 8-track tape machine. She has a very limited processor selection, but since she is a guitarist and keyboard player, she has many pieces of musical equipment. Synthesizers in the studio include five Yamaha DX-7 controllers, a Jupiter 8 keyboard, and a Yamaha KX-88.

Howard Leese, on the other hand, has a bit more extensive collection of equipment in his home studio, which he calls "Kent Manor."

Around the control center of a Tapco 8-track mixer and Tascam 8-track tape machine, Howard has a large selection of outboard gear, and an extremely extensive keyboard and synthesizer collection.

He utilizes the Yamaha QX-1 digital sequencer computer and a Pitchrider 7000 guitar-to-MIDI converter as well as state-of-the-art synthesizers and older models such as Mini-Moogs, Prophets and Mellotrons.

As if that weren't enough, Heart's lead guitarist also has over one hundred modern and vintage guitars and a wide selection of guitar amps and effects.

CONCLUSION

As you can see, most recording artists have some type of recording system in their homes. They can range from a major to a small studio. Some are simply functional facilities used by the artist to record their ideas. They employ no other equipment than the bare necessities, such as artist Howard Jones whose modest equipment includes an Akai 1212 12-track recorder/mixer and a pair of JBL monitors. Others, such as the aforementioned Leslie Howe and Michael McDonald have studios capable of producing high quality recordings–in some cases digital recordings.

What is really important is not the equipment–it's the creative entity *behind* the equipment.

EQUIPMENT LISTS

LESLIE HOWE
Soundcraft Series 2 16 x 8 mixer
MCI JH110-A 1-inch, 8-track tape machine
JBL 4315B 4-way monitors
Yamaha NS-10 monitors
Yamaha REV-7
Lexicon PCM-70
Revox PR99 2-track
Neumann U-89 microphone
Ashly stereo noise gate
dbx 165 compressor
BGW 750 power amp
Yamaha TX-216 synthesizer rack
Oberheim Xpander
Yamaha DX-7 synthesizer
Roland JX-8P synthesizer
Emulator II
Roland Juno 106 synthesizer
Emu SP-12 drum machine
LinnDrum
Sycologic M-16 MIDI matrix

Upgrading to:
Mitsubishi Westar 28-input automated console
Mitsubishi X-800 32-track digital tape machine
Complete Monster Cable wiring
Lexicon 480L multi-effects processor
UREI 811C Time Align monitors
Aphex Compellor
Valley People Kepex rack with DSP modules, two Gain Brain modules and two gate modules
Lenco MPA-2100 power amplifiers
AMS RMX-16 digital delay

DAVID T. CHASTAIN
Yamaha 1602 16-channel stereo mixer
Tascam 38 8-track
Yamaha REV-7
Master Room reverb
Eventide H910 Harmonizer
(2) Ibanez digital delays
Teac Model 1 mixers for the monitors
Lexicon Prime Time digital delay
Tascam MH40 headphone amplifier
(2) Yamaha K-2000 cassette machines

JBL 4311B monitors
Auratone monitors
BGW power amps
Sennheiser 421 and 441 microphones
AKG D-12E microphone
Shure SM-57 and SM-58 microphones

MICHAEL MCDONALD
API 36 x 16 console
3M-79 24-track tape machine
Yamaha C200 cassette machine
Yamaha QX-1 computer/sequencer
Apple Macintosh computer
AKG 414EB, 451, C-28, and C-12 microphones
Mark of the Unicorn Performer software
Opcode DX Librarian software
Opcode DX Editor software
Roland SBX-80 sync box
Meyer Sound 833 monitors
Yamaha NS10 monitors
Mitsubishi DS monitors
Macintosh 2300 power amplifier
Yamaha 2002 power amplifier
Cerwin Vega TO-1 monitor eq
GML 8200 parametric eq
Lang PEQ-2 eq
UREI 1176 limiters
dbx 165A limiters
Valley People TR-806 gates and compressors
Lexicon Prime Time digital delay
Yamaha SPX-90 effects processor
Yamaha REV-1
Yamaha REV-7
Eventide 949 Harmonizer
Hammond B3 and Leslie organ
Yamaha DX-5 synthesizer
Yamaha DX-7 synthesizer
Yamaha TX-7 synthesizer
Yamaha TX-816 synthesizer rack
Yamaha RX-11 drum machine
LinnDrum II drum machine
Roland Super Jupiter synthesizer
Roland JC-120 guitar amplifier
Fender Twin guitar amplifier

HOWARD LEESE
Tapco 8-track mixer
Tascam 8-track tape machine
Nakamichi and Tandberg cassette machines
Yamaha REV-7
Full array of graphic and parametric eqs, vocal stressers, delays, and compressors
Auratone monitors
E.S.S. monitors
Sony DX-7 digital headphones
Yamaha QX-1 digital sequencer/computer
Yamaha TX-816 synthesizer rack
Yamaha RX-11 drum machine
Pitchrider 7000 guitar-to-MIDI converter
(3) Mini Moogs
Prophet V
Solina
Mellotron
Clavioline
KX-5 remote keyboard
CP-70 electric grand piano
Marshall, Fender, Carvin and Seymour Duncan guitar amplifiers
SRD Rock modules
Assorted guitar effects

NANCY WILSON
Tascam Studio 8 mixer
Otari 8-track tape machine
Otari 2-track tape machine
Yamaha NS10 and NS50 monitors
Yamaha SPX-90 effects processor
Yamaha REV-7
Audio & Design limiter
Yamaha DX-7 controllers
Roland Jupiter 8 synthesizer
Yamaha KX-88 keyboard
SRD Rockman and rackmounted Rockman

RICK DERRINGER
Vesta Fire MR-10 cassette multitracker
Sony PCM-F1 digital processor
Hitachi Hi-Fi VHS recorder
JBL L11T and L112 monitors
Yamaha SPX-90 processor
Alesis MIDI-Verb
Mark Levinson ML3 power amplifier
Mark Levinson JC2 pre amp
Roland TR-505 Rhythm Composer
Shure SM-57 microphones
Sony lavalier microphone
SRD Rockman
Seymour Duncan Convertible guitar amplifier
Peavey guitar amplifier ■

On Taxes

Mark E. Battersby

The majority of home recording studio owners and sound engineers who lease equipment–on both sides of the transaction–were looking forward to January 1, 1988. On that date, new leasing rules were scheduled to go into effect that would have allowed fixed-priced purchase options to be ignored when determining whether a transaction was actually a lease or whether it was merely a disguised sale.

Unfortunately, the Tax Reform Act of 1986 effectively dashed the hopes of those involved with leasing by simply repealing the so-called "finance leasing" rules. But leasing as an alternative to purchasing studio or sound equipment is not dead by any means, although it should certainly be re-evaluated under the new tax law.

First, it is still possible for a home studio or the self-employed sound engineer to deduct, in full, payments made under a bona-fide leasing agreement if the operation is considered to be a business for income purposes.

Individuals, regardless of how rich, cannot deduct leasing expenses except as legitimate business expenditures. On the negative side, however, those tax deductible leasing payments will offset income that will be taxed at much lower rates under the new legislation.

Eventually, after staggered effective dates, corporate income will be taxed at a maximum 34 percent tax rate, far lower than the present 46 percent maximum rate. Sound engineers or home studio operators as sole proprietorships or partnerships will face tax rates of either 15 or 28 percent under the new rules.

When comparing the bottom-line costs of leasing as opposed to purchasing, the new tax rules limiting deductions, repealing the investment tax credit and governing depreciation all must be considered.

The most important factor contained in the new tax law is the retroactive repeal of the investment tax credit. That's right, new equipment acquisitions made after January 1. 1986, no longer qualify for the 10 percent investment tax credit. Instead, Congress instituted a new first-year write-off as partial compensation.

The old tax rules permitted a write-off of up to $5,000 of the cost of any equipment acquired during the tax year. Although the first-year write-off was scheduled to increase to $7,500, Congress jumped the gun and created a $10,000 first-year write-off for equipment acquisitions.

Of course, like many of the other provisions of this massive tax bill there is a problem. The 10,000 ceiling is only for those studio operations whose total investment in equipment is $20,000 or less. For others, for every dollar of equipment investment in excess of $200,000, the $10,000 would be reduced by one dollar. Plus, the first-year write-off would be limited to the taxable income of the sound or studio operation.

A similar situation exists with the depreciation allowance. Although next year a new, faster method of depreciating equipment acquisitions will be the accepted norm, new useful laws have been established. Beginning January 1, 1987, the 200 percent declining-balance method of depre-

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ciation will replace the present 150 percent declining-balance method used under the Accelerated Cost Recovery System (ACRS). However, depreciable property would now be grouped into new categories:

Three-year-class: Most short-lived equipment, but not automobiles and light trucks. This class uses the 200 percent declining-balanced method, switching to straight-line.

Five-year-class: This category encompasses automobiles, light trucks, qualified technological equipment and research and experimentation property. 200 percent declining-balance depreciation all the way.

Seven-year-class: Into this category falls most equipment employed in the recording industry.

Commercial and industrial real estate, such as the studio itself, has a life of 31 1/2 years under the new rules and can only use the conservative straight-line method of depreciation.

Most equipment dealers will soon be able to offer their customers only two options, outright sale or leasing. The once popular installment sale has also been the victim of tax reform.

According to the new rules, the use of the installment method will be denied for a portion of the sales of dealers (other than sale under a revolving credit plan) as well as a portion of the sales business or rental property whose selling price exceeds $150,000. The disallowed portion would be that portion that bears the same ratio to total installment sales that the instrument or sound equipment dealer's outstanding debt bears to the sum of the dealer's installment obligations and the adjusted basis or book value of his other assets.

Why all of the emphasis on the installment rules? First, it affects everyone as a customer. Second, the rules as they are written in the new tax law, also will apply to everyone who attempts to sell business assets or equipment by giving the buyer a break. Time payments fall under the installment rules as do transactions involving a down payment and the balance payable upon delivery.

Since provisions of the new tax law are expected to bring the cost of borrowed money down, leasing, like most business transactions, becomes more of a financial decision rather than a tax one. Thus, the lease/buy decision now hinges on what figure the leasing company offers as opposed to the financing rate offered by the bank or finance company, less the much diminished tax consequence, of course.

With the experts predicting lower interest rates as a direct result of this new tax legislation, money to finance equipment acquisitions should be more plentiful and less expensive. Of course, these two effects should also trickle down to the leasing company whose rates will, in all likelihood, remain competitive.

Quite simply, the lease/buy decision is now (or soon will be) a business one. Will it be better for your operation to lease a given asset or buy it? What will each strategy cost your operation both before and after taxes? And, of course, what do you want to happen to the asset when it is paid for or the lease expires? How does that affect the lease/buy decision?

Leasing or buying under the Tax Reform Act of 1986 certainly will not be boring. ■

Drew Daniels

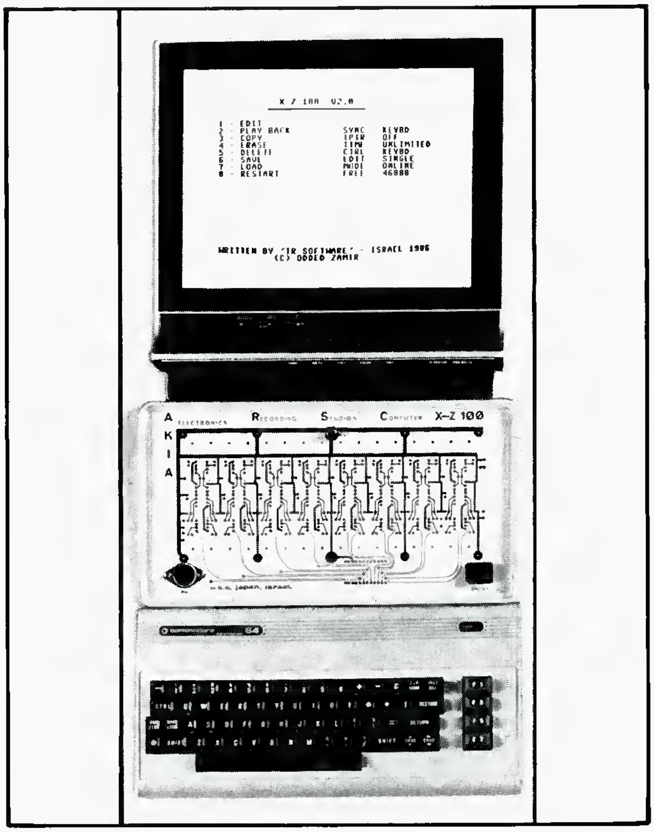
Hands On

Hands-on Equipment Review of A-KIA XZ-100 Recording Studio Computer.

The XZ-100 Recording Studio Computer by A-KIA Electronics is a stunning example of human ingenuity. That ability of enthusiasts of a particular discipline to make the best use of things at hand.

The XZ-100 is based on the ubiquitous Commodore 64 home computer, which is the biggest selling computer, and which has been known to sell at discount stores for under $100.00 making it utter foolishness to pull your Macintosh or IBM out of service to use for controlling recording processes. Instead, the investment of a few hundred dollars in "dedicated" computer and B&W or color monitor equipment allows the XZ-100 user to build up a permanent automated mixing facility in the control room at the most minimal cost.

The price of the complete XZ-100 system is $1799.00. This includes the special software which I will describe later, one VCA controller box which has 16 independent throughput channels, and the interface unit which is capable of interfacing up to eight boxes, giving the user up to 128 independent channels of VCA control under the direct access of the one Commodore 64 keyboard. In addition to the basic unit, A-Kia Electronics has available a 16-fader control console for use with the "Status-Maker" program within the XZ-100's software package which enables slide pot control of mixes in real time the way it would be done on an automated console with built-in VCAs.



1. Main menu of the X-Y 100 computer, showing various functions.

SOFTWARE IS THE KEY

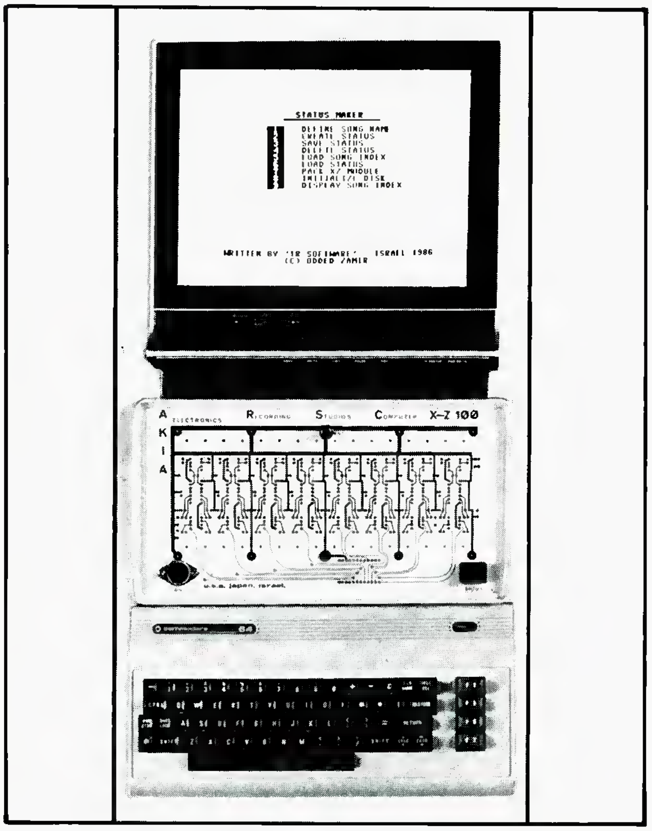
Hip computer enthusiasts know that the sophistication of microprocessors is of secondary importance to the sophistication of the software driving them. A good hacker can make even the dumbest micro do amazing things. In the case of the XZ-100, the special software makes the Commodore 64 a completely different computer. The owner of A-Kia Electronics, Odded Zamir, an Israeli-born computer scientist and audio fanatic, has written from scratch what he describes only as a completely new and unique computer language, "a totally new type of computer programming," that is used exclusively for the machine-control oriented tasks involved in the timing and manipulation of gain adjustments similar to those used to drive machine tools, such as computerized milling machines or lathes, and not for word processing, spread sheets or computer video games. The software makes use of the most basic level of operation of the Commodore's microprocessor or CPU. The CPU is addressed in such a way that the access time (the time it takes for the CPU to receive, process and re-issue instructions to control the VCA circuits) is reduced to 2 milliseconds (0.002 seconds), or about the length of time it takes for sound from a snare drum to reach the drummers ears! In contrast, the human brain takes about 150 ms or more just to respond to a stimulus and produce a physical reaction.

How short a time is this? Well, consider a very fast metronome setting, say 180 beats per minute. This makes the rate 3 beats per second with a period of 333 ms between beats. Even sixteenth-note triplets at this metronome setting (faster than a drum roll) will occur 28 ms apart!

In human terms, 2 ms is virtually instantaneous. Writing and updating mix information is therefore as accurate as the operator's ability to move fingers. The XZ-100 is MIDI controllable, and in October, Mr. Zamir made a trip to Israel and returned with SMPTE time code control capability added to the XZ-100's software.

USE AND LISTENING TESTS OR "I'LL BELIEVE IT WHEN I SEE (HEAR) IT"

Mr. Zamir brought the XZ-100 to my back-room recording studio and hooked up all my tracks in about as much time as it takes to plug an 8-track into a patchbay. We set the XZ-100 box and the Commodore system on my producer's desk and started mixing with it immediately. From the beginning, the operation was simple and smooth. I tried challenging Mr. Zamir with "what about this..." problem situations for complicated mixdowns and tricks, but at every turn, he instantly smiled and said "no problem, all you need to do is..." and as quick as pushing a key, he would have the XZ-100 doing what was needed. The first problem seemed to me to be all but impossible. How could the XZ-100 possibly synchronize its programmed mix to my old Tascam 80-8. My machine has a one-speed synchronous motor–no speed control–and no way to read or be controlled by a speed control device.

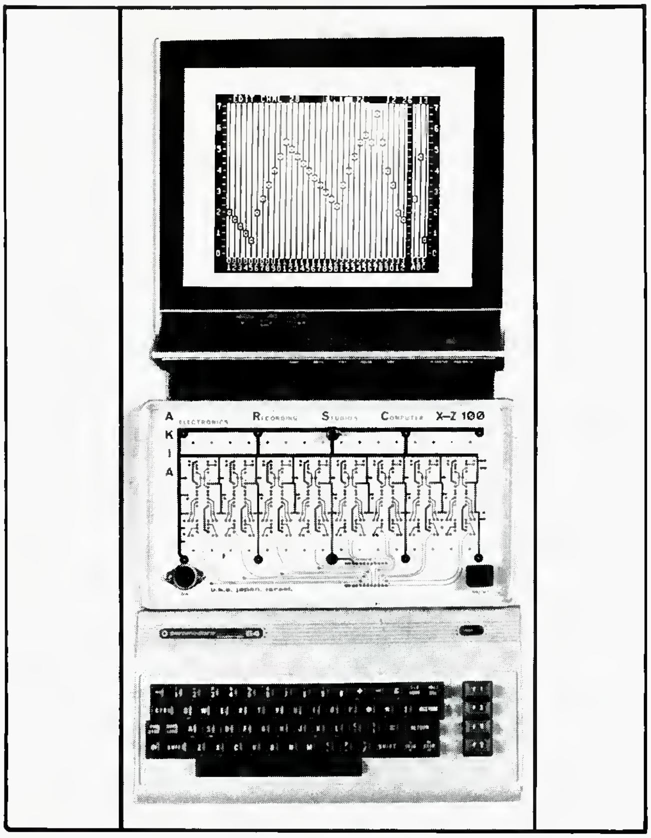


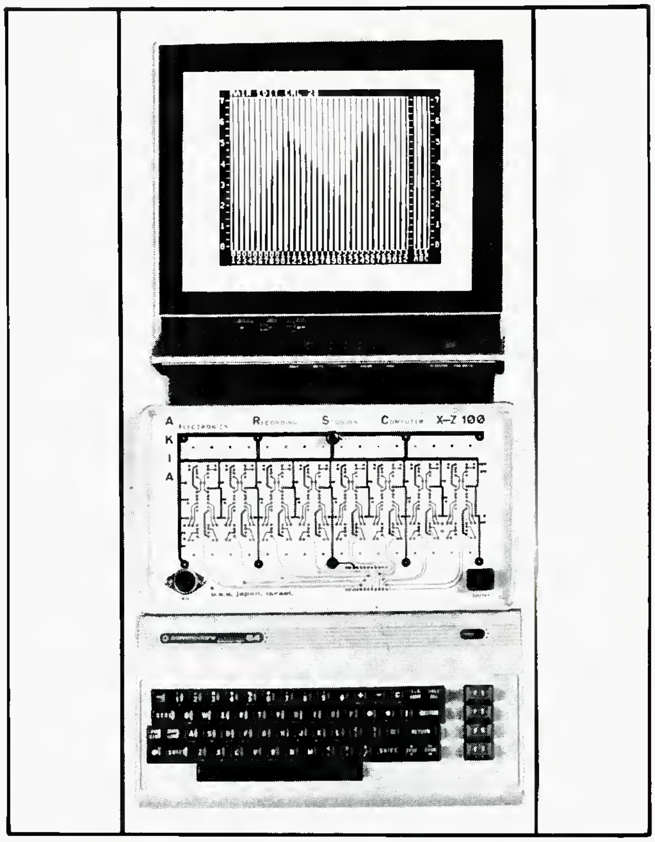
2. Status Maker menu is used to recall various sections.

Mr. Zamir pointed out that any well-maintained tape deck runs at a very constant speed. The XZ-100 can start its sync-to-tape by recording a short tone burst code pulse or "time pointer" on the tape. This can be done on any track, at any point, e.g. the count-in before the tune starts. The pulse can therefore be added to old tapes without disturbing any existing taped information. During subsequent mix programming, the XZ-100 instantly grabs this one little "blip" off the tape and starts its internal quartz-controlled clock. The error in tape timing for a five-minute pop tune we worked on was not audible–it might well have split a 64th note by the end of the tune–it was certainly less error than a punch-in produces. He demonstrated how the mixed tracks could be level-controlled and how parts, notes, clicks and noises could be edited out by the muting function of the XZ-100, and how the XZ-100 could operate as a gate and provide many other functions. You can gate reverb or echo by simply patching the XZ-100 into an echo return bus. If you use direct outputs or auxiliary buses to send to effects, you can have complete individual track/effect control by inserting XZ-100 channels into the effect send lines.

The software features a screen graphic representation of faders arranged as you would see them on the top panel of your mixer. You can watch as the computer-memorized mix moves the screen faders or mutes the channels. The most interesting facet of the software is its ability and method of storing "events." An event is any change or mix decision that the operator programs in, whether it be a fader move, a fader jump, a mute or an un-mute. The XZ-100 can store up to 12,000 events, making it unlikely that you would over-program the disk-based memory even if you were doing a three-act stage play. The way the XZ-100's software is written allows non-destructive additive or subtractive editing–updating of the mix (the program you are writing as you memorize) without disturbing the previously recorded data. You can easily add or delete events as you go along and save the updated programming to disk at any point during the session so that you can load the saved program into the XZ-100 again at any time. This latter feature also allows you to make separate mixes that you can use to test balance preferences for mixdowns, each with its own recallable program. You could, for example, make a separate "AM mix," "LP mix" and "Demo mix" without having to remember settings. You can trust your fingers to move each switch and fader to make copies of certain mixes the way I almost always do when I have to make lots of cassette master copies. Rejoice! The end of tape boxes filled with non-decipherable, chicken-scratched track sheets is at hand!

The "Status-Maker" program portion of the XZ-100 software package lets you create "modules" of mute/un-mute and fader event decisions that you can then edit, by module, in any order. For pop songs with verse-chorus-verse-chorus... configurations, this is particularly useful since all you need to do is program a verse module and a chorus module and then daisy-chain them according to the tune's structure. The Status Maker program would also be useful for live on-stage performances or audio blocking setups within scenes of stage plays or musicals, or even in blocking out opera performance. I





3. A display of changes made to a specific channel; in this case channel 28.

have sung in rock, country and folk groups, played in legitimate plays and sung lead roles in grand opera, and the XZ-100 gives me a lot of ideas of how I might automate production in every case.

The tradeoffs? In case you're wondering "this thing can't be perfect and so inexpensive too?" Well, there is one thing I would change (Mr. Zamir, for all I know, might be doing this as I write). The vertical volume scale on the on screen fader display is numbered from 0 to 7, which belies its relationship to linear auditory volume increments such as decibels. No doubt a very minor point, but for an old timer like me, it's a bit annoying not to be able to relate volume to a dB scale. I have lived with dBs for so long, it's hard not having them as my reference. The volume increment/decrement range is 66 discrete steps which are inaudible with most any type of program material you might care to record. Steps are necessary because the updating makes each step one event, so if there were many steps, e.g. 65,000 steps, one fader move could fill up the available memory. However, even a trained ear will not pick out these steps on music or speech material or, I submit, even pink noise, only perhaps on steady sine-wave test tones. It is interesting to note that the software, like direct access TV tuners, allows you to jump directly from one fader position to another. I suspect that a jump takes up only one event in memory rather than the several that would be required to step through the fader range from point A to point B. If so, it is indicative of the extremely economical programming style of the XZ-100's software.

CONCLUSIONS

I found the XZ-100 Recording Studio Computer to be everything it claimed to be in its brochure and advertisments. The unit and its concomitant literature are rather unsophisticated visually, but this belies the XZ-100's power as a production tool. The XZ-100 seems to support most of the functions I appreciate in the large and expensive automated console mixing computers and offers easy program editing and non-destructive updating, including some very handy operational functions not found in the megabucks systems. I would recommend the XZ-100 to those who have a few tracks and are thus forced to make complicated drop-in recordings using every blank track space available. These are usually difficult to mix down. Those who make plain complicated recordings such as sound tracks or shows with sound effects, commercial jingles, and combination live/mixdown recordings (I do this a lot–fill up eight tracks then use live players to make a live to 2-track plus 8-track mixdown transfer-collapse). Those who have a lot of tracks and effects, and need some way to manage them and stay sane will also find myriad uses for the XZ-100.

For information about the XZ-100:
Circle 50 on Reader Service Card.

The basic specs of the XZ-100:

Total Harmonic Distortion:	0.008%
Fader attenuation:	0 to -100 dB
Mute attenuation:	-100 dB
Dynamic range:	108 dB
Input level range:	-80 to +20 dBu
Inter-channel crosstalk:	-95 dB @ 1kHz
VCA response time:	100 dB/2 ms
Output noise:	-126 dB

[20 kHz bandwidth, unweighted, referred to 0 dBu (0.775 V)] ■

Brian Battles

Ad Ventures

Let me be one of the first zillion people to wish you a Happy New Year. Welcome to 1987! As you read this, we have finished with the holiday advertising crush period (including the "after Christmas" sales) and are heading into a new prosperous year.

I hope that you were able to undertake your own successful Ad Ventures in time to take advantage of the big dollars spent in fourth quarter budgets. If so, you have most likely undergone a baptism of fire and now consider yourself a seasoned veteran in the world of commercial advertising production. Please keep an open mind, however, because you might be able to wedge a couple of morsels of new information into the cracks those brain cells. Believe it or not, it still happens to me; occasionally, even I learn something new from time to time.

Let us review some basics which may have been washed out via your kidneys along with the holiday cheer. As we have seen, there are four main types of radio commercials:

1)Script read by itself (*a capella* voice-over or "straight read");

2)Script read over an instrumental bed;

3)Dialog or character skit;

4)Script mixed with specially-produced musical "jingle."

Obviously there are numerous variations on the above four kinds of ads, and many techniques can be combined, blended, concatenated, and alternated with one another. In addition, you can (and should) experiment with technical effects such as echo, reverb, pitch and/or speed variation, synthesized sound, eq, filtering, flanging, phase shifting, and so forth. The only limits are your available equipment and your personal creative engineering abilities.

At this point I would like to introduce a fifth type of commercial which I politely term the Amateur Hour spot. Actually, this shouldn't be in a separate category since it is an idea that is generally most effective when used in conjunction with one of the four listed above.

My definition of an Amateur Hour commercial is one in which you are using the voices of people who never had the opportunity to learn the ways of the arcane art of public speaking. These individuals can be our lovable client himself, his or her spouse, children, friends, neighbors, customers, sales personnel, civilians. average joes, or worse yet, you could be working with a businessman who knows more than everything about announcing because he spent "a whole semester doing a show at a college radio station" ten or fifteen years ago.

While it is almost always preferable to record these ads in the controlled and Utopian environment of your own studio, for one reason or another, you will often have to take a tape deck out "on location" to get on-the-spot live ambience, especially if you want to try interviewing customers extemporaneously.

By the way, I had a hard time working up the nerve to bring up this whole subject of recording with amateurs because it unleashes a Pandora's box of production ills. Incorporating the use of individuals who are not experienced announcers, can lead to Amateur Hour productions, which offer you the exciting opportunity to open up a really fine ulcer and can directly contribute to hypertension, insomnia, nausea, fatigue, and irresistible urges to beat the living daylights out of some poor, wretched, brain-damaged boob. Yet, nobody said this life would be easy, and as a devout pragmatist I am always willing to try anything that might remotely prove to be financially successful.

Have you ever heard a radio commercial or seen a TV spot that feature clips of satisfied customers, i.e., real people who are apparently thrilled with the chance to extol the virtues of doing business with a particular firm? Better yet, how often have you found yourself listening to the droning voice of a local merchant rattling on about his marvelous values and extraordinary service? Now that I brought it up, if it seems like these ads are all over the place, I'll tell you why: It's because they work. Not always, and I'm sure some of them do more harm than good, but if produced properly, "live" commercials can stand out to a listener and create a favorable marketing image. (They can also stand out to a listener and create a sensation of boredom, disgust, anguish, and downright hostility, but I like to think that those bad spots are produced by other people).

There are two fundamental types of spots that may be placed under the Amateur Hour heading: those which make use of written script, and those which rely on the candid sound of natural ad libbing.

Both of the above kinds of advertisements are usually produced by editing bits and pieces of the "live" stuff around a professional announcer's voice-over, and they both can be (and often are) faked. By that I mean recorded in a studio with background sound effects added later, and/or by using experienced talent to intentionally sound like just plain folks. (If you are somewhat short on patience, I recommend this "theater of the mind" approach.)

One of the greatest things we learn in this industry is that virtually everybody is amazed at what their voices sound like on tape. Not only that, but you will soon find out that most humans, especially local businessmen, consider their tonal gifts and ability to read copy to be equal to the likes of Gary Owens, Mason Adams, Bert Berdis, Elaine May, Casey Kasem, Bob Perry or Alan Freed. This self-assessment is unfailingly in reverse proportion to their level of articulation. You may be faced with the task of informing the bozo who's going to be writing your check that he sounds worse than the bumpkin on the Bartles & Jaymes Wine Cooler commercials. I soberly advise you to withhold suggestions that your client would have been rejected from auditions for *Hee Haw* or *The Gong Show*. So what are you going to do with all this "talent"? Simply start taping.

You see, there are two distinct schools of thought on the subject of recording the voices of amateurs for use in professional productions. One way is to write notes or even a complete script for them to work from, and then go through about 25,000 takes, coaching, prodding, and cajoling the dimwit behind the mic to sound only fractionally retarded as they mumble along in a loud, nasal monotone.

The second way, the more practical way, is to roll tape and let 'em rip. Using this method I wind up with two or three thousand feet of useless drivel and maybe twenty five to thirty feet of good material. Then, several Margaritas later, I can relax and separate the wheat from the chaff in my own peaceful editing room. And since I took down notes on where the verbal gems appear on the tape, I can normally assemble a passable thirty to sixty seconds of audio to work with in finishing the spot. If you can figure on about fifteen to twenty seconds for your professional announcer's voice-over portion of the ad, and a bit of music to fill in the gaps, you should have a decent commercial in the can. And it may even air someday, if the client can be convinced to approve your masterpiece.

Here's an important tip: Whenever you want to produce an ad featuring amateurs, suggest a concept which requires a sound collage of several different voices. Just remember whose voice you chop up. If the daughter of the owner of Cecil's Ceiling Center has a horrendous lisp or a marked sinus condition and you hack her part out of the piece altogether you will most likely spend the next few days rummaging through your wastebasket for those precious yards of Mylar you smugly axed. A good way to avoid this is to record Cecil's whole godforsaken family, his best customers, his neighbors, etc., just so there will be plenty of loved ones to choose from.

Almost without fail, sponsors demand to hear their commercials before they are spewed out onto the airwaves, and they are remarkably adept at pointing out minute but enormously critical weaknesses in your work of art. The folks who are paying for your services are uncanny in their skill to isolate a barely audible production blemish, and you will be absolutely astounded at the convoluted logic they employ in claiming that the advertisement is worthless unless you can wriggle a few more vital words into the allotted sixty seconds. In the next column, I'll take a look at some good techniques you can use to secure client approval on finished spots with minimal damage to your nervous system.

Note to faithful **Ad Ventures** readers: By the time you read this I will be in my new office as Audio Production Manager at CareerTrack, Inc. In addition to its famous management training and self-improvement seminars, CareerTrack also markets an outstanding selection of books, video tapes and audio cassette programs on a wide variety of valuable topics. My job will be to handle all aspects of audio recording, engineering, production design, and quality assurance. I will also have a hand in writing and editing copy as well as providing voice-overs and covering miscellaneous creative and administrative responsibilities. In the near future, I plan to offer you an inside look at the new recording facilities we are building with some details on our operations. For a catalog on CareerTrack's products and services write to 1800 38th Street, Boulder, CO 80301 (Tel.303-447-2323.) Tell 'em you heard about it here. ■

Jim Fiore

A High-Performance Direct Box

Here's a handy little item that every small studio should have available

MANY TIMES THE AUDIO PROFESSIONAL IS REQUIRED TO patch a high impedance unbalanced signal (such as a guitar) into the low impedance balanced system used in consoles. A device which accomplishes this is usually referred to as a direct box. Generally direct boxes come in two forms, either passive or active. The active version usually provides superior performance but has the liability of requiring a d.c. power supply. The intent of this article is to present a high performance active design that db readers can build themselves at a modest cost.

At the outset let's decide exactly what our unit should do and what performance specifications we desire. Functionally, the unit should present a very high input impedance to the musical instrument to prevent loading effects. These effects may include a change in timbre, an increase in noise or distortion, and other sonic degradations. The unit should also provide a return path to the musician's amplifier for normal monitoring. The unit must then produce a pair of outputs from the single input; one output being 180 degrees out of phase, i.e., inverted. We would like the system to have flat response from below 20 Hz to greater than 20 kHz and have minimal phase change as well. Of course, low noise and distortion are also required. Physically, a small, easy to move around, *destruction proof* construction is desired. Finally, a low cost would be nice (as always)!

THEORY OF OPERATION

For the details of circuit design, please refer to *Figure 1*. As you can see, the parts count is fairly low.

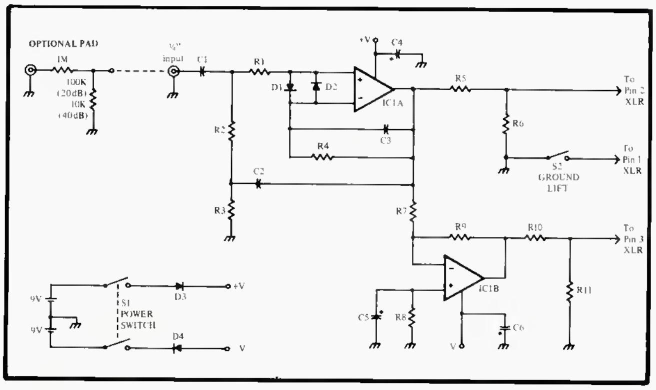


Figure 1. Schematic of circuit for active direct box.

First off, for power, two nine-volt batteries were chosen despite the fact that batteries have received somewhat of a *bad rap* lately. There are several good reasons for their inclusion here:

First, batteries are inherently free from hum unlike a.c. powered supplies and since we are operating in a high-impedance environment, this is particularly important.

Second: Batteries are relatively small and take up little cabinet space. The obvious strike against them is their life span. To combat this, one can always use rechargables. (Phantom powering was not used as many small mixers simply don't have good quality phantom supplies.) In series with each battery, we find a small signal diode. The diode pair is used as protection since many ICs will go to silicon heaven if the supply polarities are mistakenly reversed. If this happens, the diodes will be reverse biased and prohibit a possibly fatal reverse current flow. At this point the supply rails are both bypassed with uf low leakage tantalum capacitors to insure a good a.c. ground reference. Now with the power supply out of the way, let's take a look at the audio circuitry.

Op amp 1A forms a very high input impedance buffer amplifier. Note that the three capacitors in the audio line are all high sound quality polypropylene types. Polypropylene caps, while larger and more expensive than the typical aluminum electrolytics or polyester types, exhibit superior sonic performance. This is attributable to their low dissipation factor, low dielectric absorption, and high stability. Their superior sound has caused many audiophiles to rework existing equipment by replacing old caps with polypropylene types. Fortunately, their inclusion here will only add two or three dollars to the total cost. C1 is the input coupling cap and is used to block d.c. from any instrument or effects device. R1, D1, and D2 comprise an input protection network for our op amp. If the input levels get too high, one of the two diodes will become forward biased and thereby hold the differential input to a safe 0.7 volts. R1 serves as a current limiter. The op amp is set up as a follower, meaning that the voltage gain equals unity. In order to produce minimum d.c. output offset, it is imperative that the input bias currents see equivalent resistances. Here R4 is set to equal the series combination of R2 and R3. Unfortunately, R4 will have a noise current flowing through it as well, and this is undesirable. To fix this problem, R4 is bypassed by C3. C2, R2, and R3 set up an impedance while still having a reasonable d.c. bias resistance.

For d.c., C2 is open and we simply see the two resistors to ground. For audio frequencies, point A is held at just a hair below Vin. The differential across R2 is very small, and, therefore, the a.c. signal current will also be very small. Input impedance then becomes dependent on the op amp and stray input capacitance. With reasonable layouts, the input impedance can easily be over 10 megohms.

For good transient response, C1 must be larger than C2. The op amp chosen is a Signetics NE 5532. This is a dual, compensated amplifier with wide bandwidth, high slew rate, low noise, and excellent linearity. The chip is not hard to find and typically runs below $2. The other half of the 5532 is used as an inverting follower.

R7 and R9 set the gain. R8 is used for offset compensation and is bypassed with C5 for minimum noise. At the output of each section we see two resistors, R5, R6, and R10, R11. These resistors set up 21 dB attenuators. This is included for individuals who do not have input pads on their mixers and who may have preamp overload problems. Otherwise, these four resistors may be omitted. Use jumpers to replace R5 and R11. (For optimum noise performance 50-100 ohm resistors may be used in place of the jumpers.) All resistors used are one-percent tolerance metal-film types. These will guarantee minimum noise and maintain the gain matching required for a reasonable common mode rejection ratio. (To optimize further, set R7/R8 as close to unity as possible and set R5/R6 equal to R10/R11. Buy several resistors of each value and hand sort them. A four and a half

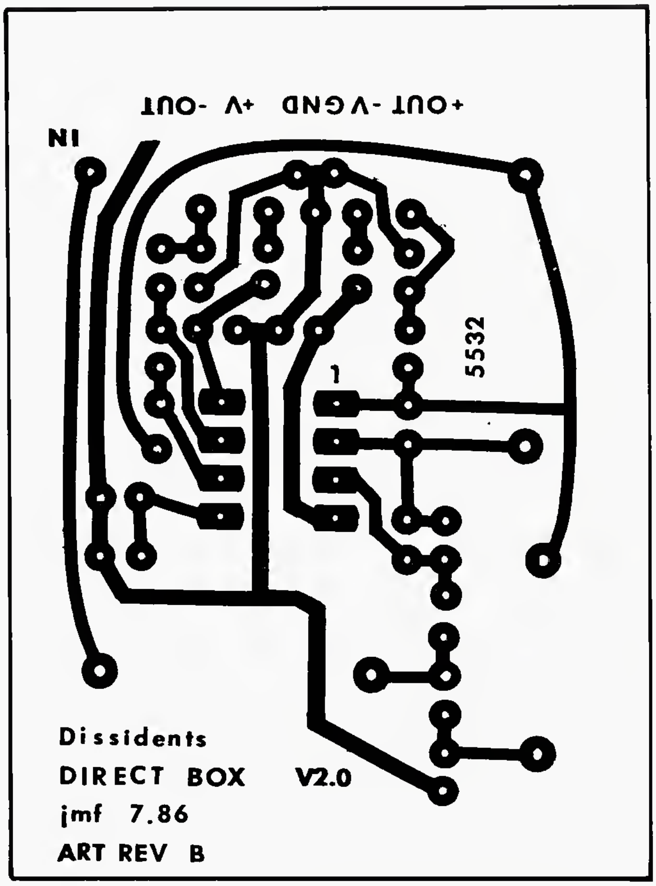


Figure 2. Foil side of the printed-circuit board.

digit multimeter will be invaluable for this.)

In the lab the circuit shows a deviation of +/- 0.5 dB from 1 Hz to 100 kHz. This insures a phase shift of only a few degrees at the audio band extremes. At very high frequencies the input protection circuit will be activated due to the reduction in loop gain. This makes finding the -3 dB point a little more difficult; however, I don't think that anyone really wants to amplify signals over 1 MHz! My present system for measuring THD is rather limited and the circuit was at the measurement limit of 0.1 percent, indicating that it is in reality much lower. Listening tests confirm this.

CONSTRUCTION

A foil side artwork pattern is shown in *Figure 2*. If desired, the unit may be hard wired using a prototype board. It is very important to orient certain components correctly. Make sure that the IC is pointing towards the input/output pads and that the polarized capacitors (C4, C5, C6) and diodes are oriented correctly. The polypropylene caps are not polarized and can be inserted either way, as can the resistors. *Figure 3* is a stuffing guide to assist you in component placement. Most of the components are standing up as seen in *Figure 4*. This was done to keep the overall circuit as small as possible. The unit is small enough in fact to fit inside most instruments! Use a small pencil iron (15-20 watt) and a good 60/40 multi-core solder when you assemble the unit. Also, it is a very good idea to use an IC socket for the chip.

There are a few items that you may wish to include that are not shown on the PC board. First, a ground lift switch is certainly desirable. This helps to

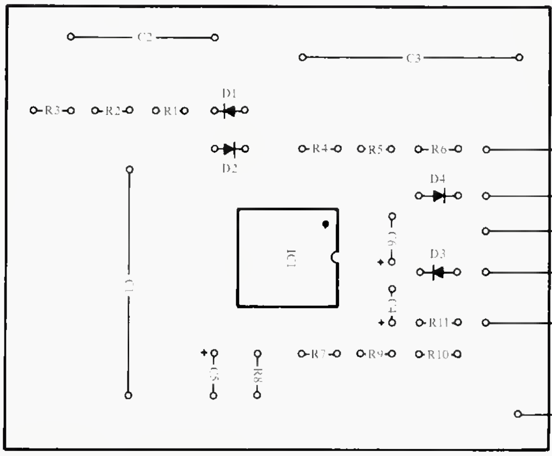


Figure 3. Stuffing guide for the circuit board.

prevent hum and noise caused by ground loops. Simply connect an SPST switch between the input ground and pin 1 of the XLR connector. One of the two grounds (usually the quarter-inch input jacks) needs to have a very good chassis connection to insure good shielding. Be sure that both the one quarter inch and the XLR ground connectors *do not* go to the chassis. Doing so will void the action of the switch by shorting it.

Another item of interest would be the inclusion of an input pad. This would enable you to take high level as well as instrument level signals into your console. For example, you may wish to tap off a guitar amp's loudspeaker in order to get the sound of the amp onto tape. To do this, simply add the pad shown in *Figure 1* to the front end of the direct box.

Finally, for the battery-hating diehards among us, a separate power supply is possible. Anything from nine to fifteen volts will work fine. A number of sources for either pre-made or kit supplies exist. I would suggest that the supply be very well shielded, or better yet, placed in a second chassis. Remember, working at high-impedance levels means that hum susceptibility is very great. Good layout and shielding is imperative for a low-noise unit. For the best performance, the device may be permanently mounted inside the instrument.

If you cannot purchase the components locally, there are several good mail order sources. **Digi-Key Corporation, PO Box 677, Thief River Falls, MN 56701** carries all of the components mentioned here. Other good sources include **JDR Microdevices, 1224 S. Bascam Ave, San Jose, CA 95128** and **Jameco Electronics, 1355 Shoreway Rd, Belmont, CA 94002.**

If you happen to have a few items laying around, be advised that the LF 353, TL072, TL082, and RC4558 op amps are all pin compatible with the 5532. Generally, they are all a bit noisier, and a little slower. Polyester (Mylar) caps can be substituted for the polypropylenes, with some sonic degradation. Do not use ten-percent resistors unless they have been hand matched, as they can seriously degrade the common mode rejection of the circuit. Total parts cost can vary quite a bit depending on where you buy. I recently saw a chassis selling for six dollars in one store while a competitor charged only $2.20. Shop around and you're guaranteed to save some cash. The completed unit shown in *Figure 5* was under $19 total. Over half of that was for the chassis, switches, jacks, and the like.

CONCLUSION

Using the direct box is a very straight-forward affair. Just plug an instrument into one of the two quarter-inch jacks and use the other quarter-inch jack as a return to the musician's amplifier. You may insert effects and the like between the instrument and the direct box. Connect an XLR cable between the direct box output and the console input. The unit may produce a small pop when first turned on or off, so make sure that the console's input channel is muted. Now simply throw the ground switch between positions, and leave it at the quieter of the two. That's all there is to it!

Hopefully this article will now allow many of you the luxury of a high-quality direct box(es) without the high price tag. Happy soldering! ■

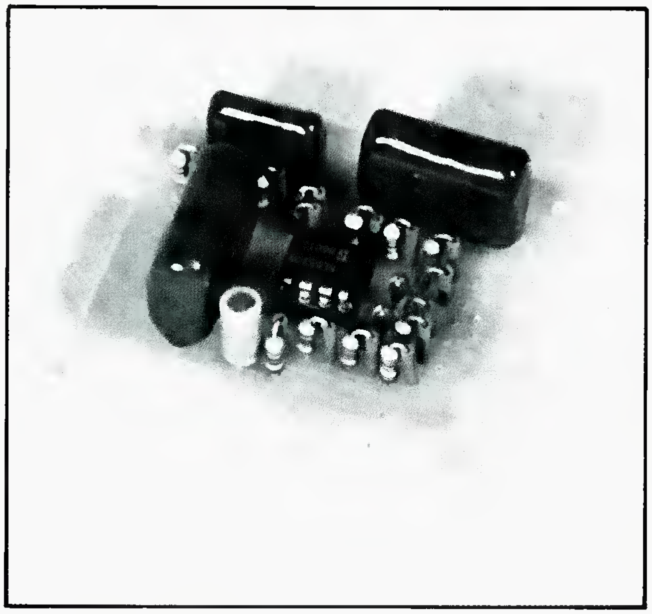


Figure 4. The completed circuit board.

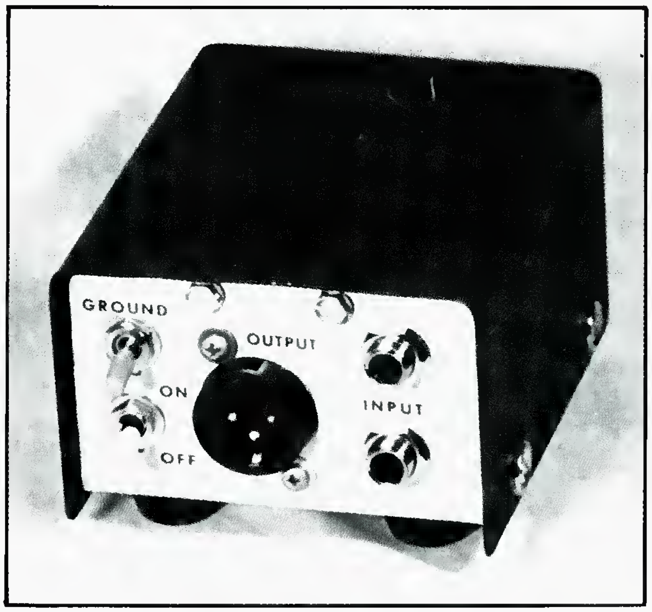


Figure 5. The completed direct box.

On Law

KENT J. KLAVENS

SAMPLING, PROCESSORS AND THE LAW: HOW FAR CAN YOU GO?

With the rapid technological advances of digital recording devices and keyboards in recent years, any financially able songwriter or producer can purchase or rent equipment that can "sample" and reproduce virtually any sound known to man, whether vocal, instrumental or otherwise. Names such as Emulator, Synclavier, Fairlight, Mirage and Kurzweil have become synonymous with these techniques, and the practice has become so widespread that "user groups" now trade disks containing a variety of sounds.

After hearing that disks duplicating the Fairlight sounds created by Peter Gabriel were being copied and circulated, I thought a legal article should be published to outline what limits, if any, might be imposed upon this new age technique of copying the sounds, voices and creations of others. Hopefully, this knowledge will save some of you the eventual cost and embarrassment of defending a major legal action.

First, here's the good news. In general, as a practical matter, any sound, instrument or voice you have sampled which has then been processed into another sound totally unrecognizable from the original sound or "nominal pitch" is probably fair game. As long as you do not reveal the source of your new sound to anyone, no one could ever prove you copied someone else's creation or performance. Don't smile smugly and stop reading this article here, however.

What if someone could prove you sampled a particular sound or musical or vocal performance? What if the final sound you use is clearly similar to the sound or performance you sampled? Who has legal rights against you and under what circumstances?

The discussion that follows will review some of the key laws and legal principles that may be applied to sampling in the future. It is not intended as either a complete or final treatment of the possibilities, however, because: (1) Many of these issues have not been resolved in the courts and are not covered with sufficient specificity in current laws; (2) Such a discussion would be far too complex and technical to be even remotely interesting; (3) This magazine would have to add a few hundred extra pages to this issue. So, when in doubt, always see your attorney!

Kent Klavens is a practicing attorney in Los Angeles, California and will now be contributing this column in each issue.

COPYRIGHT LAW

This is the major body of law that protects the unauthorized copying and use of the original creation of others. For the issue raised by sampling, two key questions arise with respect to copyright law:

(1) Whether the "work" being copied is original enough to be protected by copyright;

(2) If it is subject to copyright protection, whether there is "substantial similarity" between the protected work and the copier's work. As most sampling involves the use of sounds and performances from records and tapes widely sold to the public, I am ignoring the issue of "access" to the copyright work, another key element in proving infringement.

There are four categories of potential copyright owners whose rights you may be theoretically violating when you sample sounds from an existing recording:

(1) Songwriters and music publishers, who may own all or part of the copyright in the musical composition from which the sound was taken;

(2) Singers and instrumentalists, who some legal theorists believe may be entitled to claim copyright in any performance possessing a sufficient degree of originality, although these types of copyrights have not really been claimed or utilized in the past;

(3) Record producers (who are not mentioned specifically in the Copyright Law, but were identified in committee reports when Congress was deliberating over the Copyright Act of 1976) were viewed as potential copyright claimants based upon the originality and creativity inherent in the capturing and electronic processing of sounds and the compiling and editing of sounds to make the final sound recordings;

(4) Perhaps most significantly, record companies and others who own copyrights for the actual sound recordings from which the sampled sounds were originally recorded.

Regardless of the category of the potential copyright claimant, if a "work" is not deemed an "original work of authorship," there is no copyright protection whatsoever. Often, though, it can be unclear as to which elements of a work are sufficiently original for copyright purposes.

For example, the creation of one isolated sound would not likely merit copyright protection, although a collection of sounds recorded in a particular order might. Therefore, in a recording of a carpenter striking a nail, no original work of authorship has been created by the carpenter, but the sampling of that sound, and its processing into a variety of sounds involving some degree of originality as to melody, pitch or rhythm, would probably be deemed original enough for copyright protection. In one 1983 case, a federal court ruled that the extensive editing and compilation of sounds of a hockey game (for purposes of a video game) were sufficiently original for copyright protection, although the mere recording of such sounds, without more, would not have been.

Generally, for most courts, copyrights in musical compositions have been held to cover only the originality in creation of the melody lines and lyrics. As to the rights of either the songwriter or music publisher of a particular song, there would be no copyright infringement in the copying of a particular synthesizer sound or other instrument or voice, so long as there is no "substantial" copying or use of the melody line or lyrics (the issue of substantial similarity applies to all forms of copying and is discussed later in this article).

As to the singer or other performer of a musical composition that has been embodied in tangible form, the late Melville Nimmer, perhaps our county's foremost copyright authority, noted in his treatise, *Nimmer on Copyright*, that "there is little question that a performer's rendition of a work written by another by itself constitutes an original work." No cases have yet attempted to apply this theory to copyright law, however, and as a practical matter, two major factors make it an unlikely area for such an application.

First, any such copyright would "derive" from the copyright in the musical composition, requiring the permission of the songwriter or music publisher for recognition of the performance as a separately copyrightable, "derivative work." Owners of major song copyrights are extremely careful in allowing others to win rights in "derivative works," usually giving permission only for major projects and substantial payments. Second, most record company agreements with performers, whether vocal or musical artists, specify that all of the performers' efforts will be "works made for hire," meaning that the employer will own any copyright that has been created by the performance.

The theoretical copyright envisioned by congress for record producers, those responsible for "capturing and electronically processing the sounds, and compiling and editing them to make the final sound recording," is limited by exactly the same two practical restrictions discussed in the paragraph above for performers. For this reason, as well as the lack of any major legal precedent on this issue, copyrights for record producers are seldom claimed and protected.

What about record companies or others who own master recordings? Without regard to the originality of the music or sounds recorded, and aside from the rights they possess through the services of others involved in either the performances or the capturing, processing, compiling and editing of sounds, virtually any sound recording is legally protected under a separate "sound recording" copyright. This copyright protects the actual sounds as recorded on the record. In the previously mentioned legal decision involving the video hockey game, the court also stated that a computer chip containing sounds also constitutes a sound recording protected by copyright.

As a matter of common practice in the music industry thus far, these separate sound recording rights have not been enforced to punish those who sample isolated sounds. The major reason for this is that record companies and recording artists are generally concerned only with record pirates who copy entire recordings and manufacture counterfeit or "bootleg" products, competing in the marketplace with the artist's legitimate products and reducing sales of the original recording. As a practical matter, then, it is unlikely that Atlantic Records will initiate legal actions against those who have sampled and processed the sound of Phil Collins' snare drum. The second reason is the copyright law requirement that "substantial similarity" exist between the original work and the copy in order to support a legal finding of infringement.

Assuming you are copying something that is protected under copyright law by either a songwriter, music publisher, performer, producer or record company, what is "substantial" copying? Despite what someone may try to tell you about a certain absolute number of notes, bars or words that may be legally copied, there is no such finite, quantitative standard.

Clearly, the most minimal copying of one, isolated sound of an instrument or voice from a sound recording would not constitute copyright infringement. Just as obvious, the sampling of an entire melody line, such as Madonna singing "Crazy for You," would be a copyright infringement. Where does one draw the line?

In general, courts use an "ordinary observer" test. So regardless of the testimony of legal experts as to whether the similarity between two works is sufficient to infer that one was copied from the other, it is still up to a jury (or judge without a jury) to decide whether such copying took too much of the original work. Also remember, as briefly mentioned at the beginning of this article, that in the case of original works that have not been widely sold or distributed, the creator might have difficulty proving that the copier had "access" to the original work. In a major copyright infringement case against the Bee Gees, this defective element of proof caused the judge to overrule a verdict in favor of the songwriter making the claim, despite the fact that the jury had found the two songs to be substantially similar.

RIGHT OF PUBLICITY

The sampling of voices is a far more likely area to be used by the law to enforce the legal rights of performers. In general, the legal principle involved prevents the use of one's name, picture, voice or "likeness" for commercial purposes.

California law, for example, specifically forbids the use of another's voice in any manner "on or in products, merchandise, or goods, or for purposes of advertising or selling, or soliciting purchases of, products, merchandise, goods or services..." The sampling of even one note of a singer's performance would appear to be covered, even though no legal cases involving sampling appear to have been initiated yet under this law.

One significant complexity in this area is that the "right of publicity" has not been established as a matter of federal law, but rather is given individual treatment in a number of states. In fact, one important aspect of these laws, the right of heirs of a deceased entertainer to enforce a right of publicity after death, has been handled by different courts and state legislatures inconsistently. It might be legal to sample the voice of John Lennon in some states, but be subject to legal actions in others.

The sampling of one note of sound created by a musician or programmer, even if performed in a distinctive style unmistakably associated with that person, does not appear to be covered under any right of publicity laws now written. This is because the term "likeness" (the only term in such laws potentially capable of such broad interpretation) has generally referred only to the "image" of the person. That doesn't mean that an attorney won't try to expand the coverage of such a law to bring you to justice!

CONCLUSION

If you are going to sample anything, the safest bet is to process the sound beyond recognition and tell no one the source you sampled! Otherwise, someday when you least expect it, an attorney who has creatively interpreted laws involving copyright or rights of publicity will make your life miserable. ■

Editorial

The beginnings of what became the article about Neil Diamond's concert sound systems began about six months ago while Neil Diamond was performing at New York City's mammoth Madison Square Garden.

I had been invited by Stan Miller, Neil's sound man, to come to the Garden and see and listen. This is not a music criticism column, so I'll dismiss the musical aspects of the Diamond show with only:"I was splendidly entertained."

I was sitting at the console right alongside Stan, about a dozen rows back from the stage. Still, there was opportunity for me to walk around the Garden's tiers during the performance.

I've been to concerts at the Garden in the past. Each time, coming away with the impression that it is one of the world's largest boom boxes. Intelligibility always suffered, let alone high quality. So, it was with some sceptism at least, that I came this time. Stan Miller told me that he mixes for vocal clarity. That is, the audience comes to hear Neil, and that is exactly what they should get. All through the hall, even at extremely high sonic levels, Neil's vocals were crystal clear and clean. The music behind him, was, well, behind him, loud and clean too, but it was the clarity of the star's performance that was Stan's goal; and it was achieved.

So, hi-fi is possible at the garden. Even impressively good hi-fi. To learn more of how this was made possible, I asked Drew Daniels to follow up, since I knew that the show was heading for Los Angeles, where Drew is based. What I didn't know, was that Drew and Stan Miller already knew each other. All this is by way of saying that the sound I heard in New York is a trademark of Stan Miller's Stanal Sound. And the story of it begins on the next page...

At a recent NAMM show, a new company had come up with a fairly simple computer method to automate the console of a small studio. Intrigued, we had a system sent to Drew Daniels, since he does have a small studio setup at home. The unit is, of course, the A-KIA system described so well by Drew on page 27. Again, I'll let him tell it.

This issue introduces a new regular feature called appropriately "On Law." This first column takes on the subject of sampling and the legal aspects of using someone else's work as your own. Kent Klavens is the author and he has a Los Angeles law practice, specializing in recording studio and music-related matters, but as he points out, what is true in some legal matters in California may not be so in another state. Still, I'm sure you will find his column informative and valuable reading.

If you are picking this issue up for the first time at the winter NAMM show, welcome to the fast-growing family of **db Magazine** readers. L.Z.

Drew Daniels

Neil Diamond at the Greek Theater

Super Star Neil Diamond culminated a major tour at the Los Angeles Greek Theater where our author interviewed his long-time chief sound man.

NEIL DIAMOND IS USUALLY THOUGHT OF AS A POP ACT and not compared with big-draw rockers like Bruce Springsteen, but to sound engineers, comparisons of this sort are useless because the underlying principles of operating sound systems for big pop or rock acts is the same.

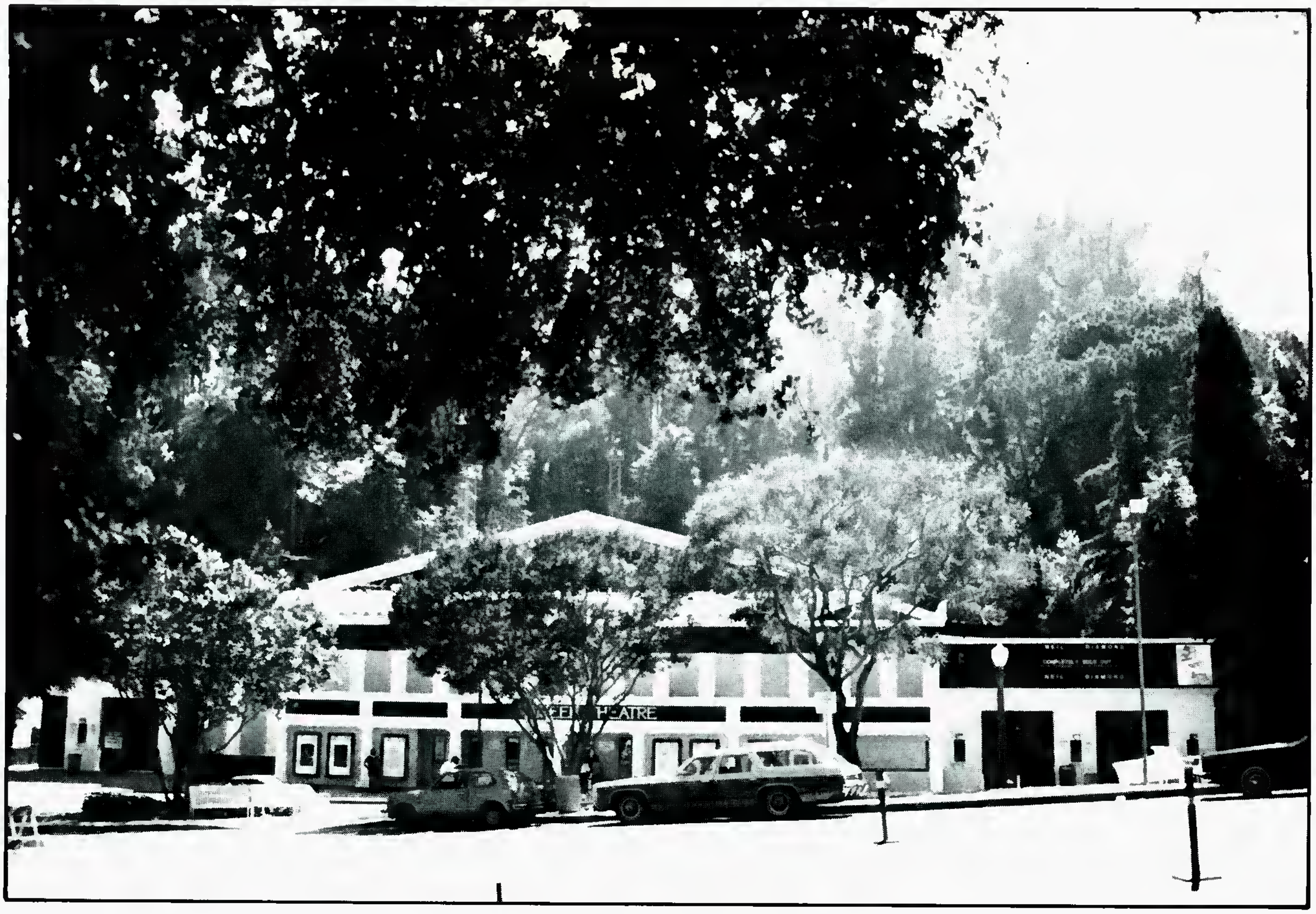
Neil Diamond fans are just as demanding as fans of The Boss or any other rock act, though I suspect their ears might be a bit more sophisticated. Diamond's fans are absolutely rabid when it comes to camping out for tickets, and that's because of his legendary showmanship as much as for his music. I said Neil Diamond fans have sophisticated ears and I think this is so because of the longevity of Diamond's career and because of the great recordings he has made over the years with recording engineer wizards like Armin Steiner and Alan Sides, who have the ability to present what their artists imagine. The great recordings, not to mention the style of those recordings, lend themselves to instruct listeners in what to expect when sound quality is of utmost importance. Springsteen's Nebraska album was recorded on a portable cassette recorder and then transferred to other media, including compact discs. This, in and of itself, belies the apparent regard The Boss has for audio quality; but at a Springsteen concert, volume and bottom end seem to take some precedence over the style of absolute, pristine, hi-fi-like clarity and precision of Stan Miller's treatment of Neil Diamond's show. This is in no way a knock on Springsteen's sound. I attended the L.A. Coliseum concert where Clair Brothers used two hundred S4 boxes and 400,000 watts of power to pound 100,000 satisfied fans. I never forget that it's the fans who make the judgement of whether the sound is good or bad. From my seat at the mixing booth in the Greek Theater , there was plenty of chest-thumping bottom. The kick drum in "Beautiful Noise" shook the concrete floor enough for any rock fan, which is interesting in terms of the neighborhood sound level problems at the Greek.(More about that later.)

NOT HEAVY METAL

Neil Diamond sings in perfect tune and is a consummate stage pro, a veteran of long years of perfecting his music and its presentation. It so happens that Diamond's style of presentation is much more "show" oriented than "rock concert" oriented, more as Las Vegas than the Grateful Dead–if that's any kind of reasonable analogy. Some of the numbers in the show are downright slick–such as the ballads using a dozen live string players mic'ed from the basement and piped through special side surround speakers. But other tunes have the rock impact of raw heavy metal. Tunes like "Brother Love's" get the audience up on their feet for twenty minutes at a time, singing along in a frenzy of excitement.

From the days when Stan carried a pair of small two-way speakers with internal passive crossovers, some 80-watt tube amps and a 4-channel mixer in a station wagon, there has been a long series of experiments, adjustments and tweeks to culminate in the equipment and speaker systems used by Stanal Sound today. "When we started–I mean everybody was doing concert sound–manufacturers just didn't make what we needed" said Stan. "We had to improvise on almost everything we used. The design of the speakers, which are the most customized part of thetouring sound business, is the result of the job that has to be done. Where you need to cover audience areas, how loud it needs to be, how much bandwidth and so on, you build a tool to do the job you have to do. The microphones, mixing consoles,

Drew Daniels is the Applications Engineer for JBL Professional and a frequent contributer to these pages.



The Greek Theater.

amplifiers, and loudspeakers, are all tools, such as a doctor's scalpel or a mechanic's wrenches. If you put the wrong wrench on a nut, you're going to damage it, so you've got to have the right tools. The relationship with the artist is one of the very most important things as far as I'm concerned. All of the best equipment in the world, doesn't necessarily produce good sound. In this business, we've been forced, over a period of years, to design those tools that didn't exist. It was a delight the first night I sat behind the Yamaha PM3000 mixing console at the Greek. It's fun to have those kinds of tools. I did this theater the first time with Neil Diamond with a home-built console, and I look back now and I wonder how I even made the thing work! Less exotic tools force you to be more creative, but if you learn on them, you're better able to use and appreciate the newer things that come along."

I reminded Stan about how promoters used to put on rock shows with maybe half a dozen acts. The prevailing attitude back in the late 60s and early 70s was "let's put on a show, we'll hire a sound company and six groups and blitz the local radio stations, it'll be just like Woodstock, wow man...." Now touring sound is part of specific shows the way Stanal is with Neil Diamond, John Denver, the Pointer Sisters, Dolly Parton, Mac Davis and Ricky Lee Jones. Stan points out, because of the sophistication of many of today's big acts, "the sound system has become an extension of the musical performance. The sound system is the bridge that brings the musical performance from the stage over to the audience, so it is part of the show."

IS THE SOUND DIFFERENT FOR DIFFERENT ARTISTS?

I asked if that implied that certain kinds of acts or certain types of music were more appropriately served by different sound companies because of a particular company's operating style or equipment. Stan's reply was: "I suppose there's a certain amount of truth in that–yes. If a sound system is well designed it ought to be able to handle lots of different kinds of things. But that is true because certain companies have expertise with certain kinds of acts and maybe not so much expertise with other acts. I'm sure there are some acts that our company doesn't have much expertise with. Part of it is because I don't particularly want that expertise.

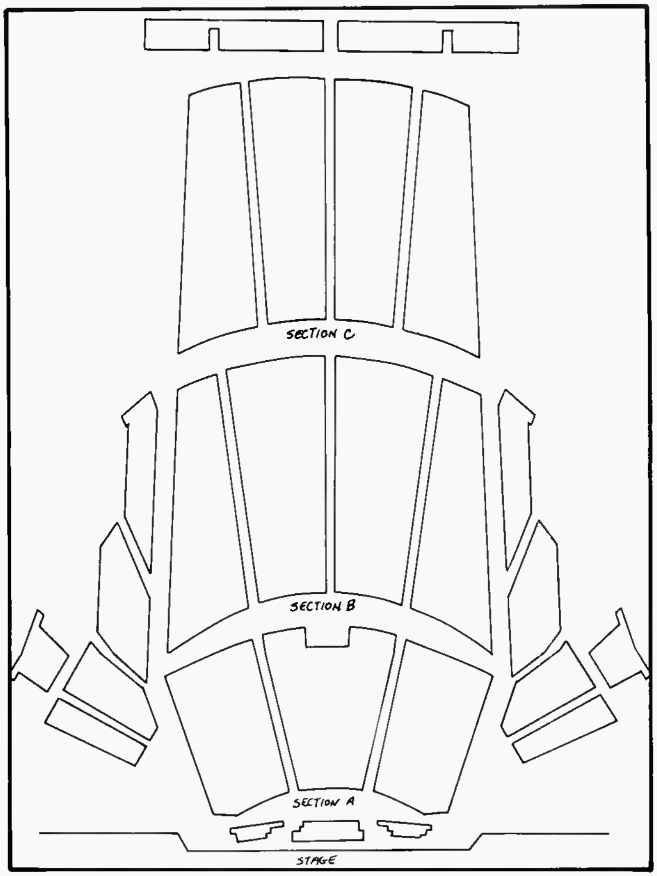
I asked what Stan thought of the many and varied loudspeaker design and construction techniques. I know a lot of people I talk to seem to think that because speaker box styles, layouts, componentry and so on are so different looking, that the speaker itself is a different kind of tool, suited for a different purpose. The readers will want to know if all the touring company's tools all over the country are pretty much the same or if they are different types of tools. Stan's reply is obvious. "Well, they try to do the same kind of a job, and there's more than one way to skin a cat."

The subject of concert venues is a barrel of laughs for those who tour with equipment and have a streak of masochism and a healthy penchant for the humor to be found in retrospect. Those who don't laugh, don't last either. Long ago I had talked with Ken Fause of Smith, Fause and McDonald, a noted sound, power and architectural acoustics firm, about how concert venues made it tough on touring acts. I'd make sure if I ever designed a concert hall or amphitheater that there would be a big PVC conduit pipe running from the house mix position to the stage under the floor. It turns out that Stan came up with that idea when the Universal Amphitheater was designed at Universal Studios in Universal City, California, the place people love to get screen tested, zapped, dunked and have King Kong terrorize them. Stan came to the Universal Amphitheater with John Denver just after it opened, and has been associated with the Amphitheater ever since. "In the early days I'd come into L.A. and stay at the Sheraton Universal Hotel because I didn't live here in those days, and I worked at the Amphitheater myself. One whole summer, in fact two summers I was out here. Of course they didn't have winter shows (until they put the roof on)." It takes a great sound system and a truly excellent sound engineer to accomodate the demanding instrumentation and varied repertoire of Diamond's show. Stan Miller has been personally mixing for Neil Diamond for 18 years, without missing a single show. In fact, after Stan had a heart attack in 1978, Neil had cancelled his tour to give Stan time to recover. He called Stan toward the end of the year and kept asking if Stan could go on the fall tour. Neil didn't want to trust Stan's assurances that he would be ready to tour in November, so Neil called Stan's personal doctor and asked him to go on the tour with them for two weeks to keep an eye on Stan. Stan says it's this kind of professional committment that has enabled him to grow with Neil Diamond and for Neil to grow into a

Stan Miller in front of the console.

technically sophisticated artist. "Neil understands the capabilities and limitations of sound systems and acoustics and which make it much easier to concentrate on the business at hand, "commented Stan. I asked if his professional relationship with Neil wasn't just about the best of all possible worlds for a concert sound man.

"Neil has been very good to me over those years, and allowed me to do a lot of things creatively. He has allowed me and my company to grow. That's one of the reasons that I still go out on the road with him. I really don't have too much desire to travel with many artists. There are a lot of engineers who have been around with certain acts for a long time, but not many owners. If Neil toured constantly, then I wouldn't do it. I'm an old man in

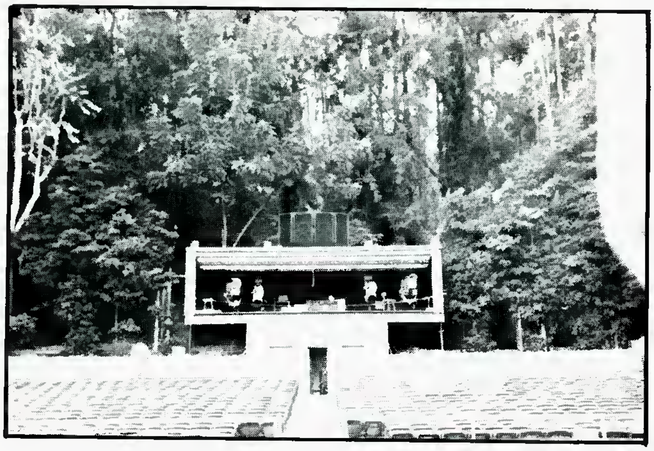


The seating arrangement at the Greek Theater. The capacity is 6,166.

this business. It's a young kid's business, being out on the road beating around from place to place and being on a bus every night. I don't have to do that with Neil. I used to drive the truck, in the early days when there wasn't anybody else. I have a photograph of Neil on stage in Moorehead, Minnesota at the University of Minnesota, where I had two little square boxes that each had a 515 Altec in them and a 288 on a 1505B multicell in a two-way system with a N500C passive crossover. I had two 80-watt Altec tube power amplifiers and a 1567 tube mixer, four channel mixer, which had only bass and treble on the overall thing, that was it! We had no stage monitors and only four microphones. I was one of the first people to use the Shure SM-58 microphone when everyone else was using the chrome-plated version, because it was "professional."

THE EARLY DAYS

In the early days, Neil went out primarily on the weekend, so we flew a lot. We literally went all over the country, I mean one weekend we would be at one end of the country and the next weekend at the other end of the country. That's kind of the way he toured. He did play some clubs during that time period, but not very many. There are a lot of funny stories from the early days, and one of the great things about Neil's organization is that there are still several people who have been with him for twelve or fourteen years. Ritchie Bennet, who plays guitar, was 18 years old when he came on the road with us, and Neil used to say on stage, that he had to ask his mother for permission to let him come on the road. Alan Lindgren, the keyboard and synthesizer guy was 18 or 19 when he started with Neil. There



The rear string stack has four 4870s.

are still a number of people with him well over ten years; Rhine Press and his wife Linda who sings backup, and Tom Hinsley who plays keyboards. It's kind of a family and that's one of the reasons that I enjoy going out with them."

I told Stan how people call me at work for design help and so on, and every last one of them comments about how they have little control of the artists they get to work with and how so many artists are unreasonable about monitors. Stan has probably had less trouble in this regard. I suspect his diplomacy and patience at educating his clients is responsible. "Well, the business is very different from what it was in the early days" said Stan. First of all, I suppose if I had it to do over again, I probably wouldn't be in this business. I'd think of some other way to make a living. Simply because it's not very easy to make a profit any more. We used to be able to make more money than we can make now from the touring part of our business. It's real tough to make a profit and part of it is because we are "equipmented" to death. It's a vicious circle, you have

One half of the main speaker section.

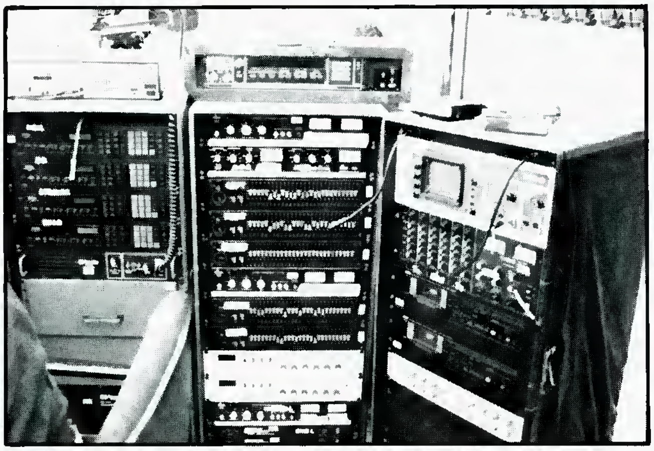
engineers out mixing and they hire a sound company and the only thing they can talk about is the equipment and what they had and we didn't have. Second, the equipment didn't exist when we started, we had to build a lot of it, and it has degenerated in a battle of engineers trying to out do each other to see who'll have the newest, latest thing. I don't think most artists are aware or particularly care about the equipment, the only thing they care about is that the sound works. They don't know a thing about the equipment–they really shouldn't have to. I changed the subject to the acoustics of show venues and asked Stan if any venue problems were unique. "I think the problems are pretty universal, we still have shows in venues that shows shouldn't be had in from an acoustical point of view"said Stan. With Neil, we have 44,000 pounds of equipment hanging in the air over the stage. That includes sound,lighting, rigging and some things that have to do with production parts of the show. I tell people we bring everything including the kitchen sink along. In the early days we used to use a Hammond organ and a grand piano. We rented those in every city wherever we went, it was part of the promoter's requirement to get the piano and the organ but we got to the point where we had so many problems that we started taking the piano and the organ with us. We still carry a grand, six-foot-four C7 Yamaha with us. It's in remarkably good condition considering that it has been around the world a few times. Most people are surprised to see it looking as good as it looks, but we have a guy whose job

The side strings are two 4870s.

is taking care of the piano and the other keyboard instruments– and a full-time piano tuner with us all the time. Generally, most of the venues that we're playing with an act like Neil Diamond are major cities, major arenas, ice hockey and basketball arenas, Most of them are newer buildings built within the last twenty years, so acoustically they are better than they use to be. These buildings have some thought behind their acoustical treatment or design. Occasionally you do run into a brand new building where they haven't spent ten cents on any acoustical treatment or even a consultant to deal with it, but generally those building are pretty good–much better than those college gymnasiums we used to play as a rule."

OTHER VENUES

I asked if he ever got stuck in an acoustical barn such as Boston Garden or the Chicago Bulls' arena. "Yes, you try to put the system in creatively, cover the environment and get maximum directivity out of the loudspeakers and so on, because those are highly reverberant spaces. In Neil's case, he has approval of venues to play and sometimes he has turned down a particular arena because of the acoustics. There are



The Greek Theater's signal processing system.

These Yamaha REV-7 units are part of the signal processing system.

even cases where we've spent show budget money to do temporary acoustical treatment to the room to help it out. As an example, every time we play San Diego Sports Arena, we hang several thousand feet of drapes up in the ceiling to try to stop some of the bad sound from bouncing around, and it does help. Neil will spend money for those kinds of things. Some artists won't. He is a knowledgeable individual, he is not off in the clouds somewhere by any stretch of the imagination. He's very aware of his entire organization,and everything that everybody does. He doesn't tell people how to do their job. Now that's unusual, because sometimes artists get involved in telling people that they've hired–whose judgement they've hired–how to do the job, and that's where some of the problems occur. Neil has allowed me to screw up. That's how I learn."

Here at the Greek Theater, there is a permanently installed Stanal system that Stan leases to the City of Los Angeles. Stanal also has systems installed in the Universal Amphitheater, in the Wilshire Theater, the Pantages Theater in Hollywood, the Golden Gate Theater in San Francisco and the Poplar Creek Music Theater in Chicago. Stanal also provided sound for the opening and closing ceremonies at the 1984 Olympic Games in Los Angeles. There are venues where Stan goes that quickly see the benefit of good sound, and so they ask Stan to provide a system. Stan says of course, not every house needs an installed system. There are places where it makes no economic sense at all, such as the places where touring companies always set up sound just for music acts and paging systems owned by the house suffice the rest of the time. One of the reasons that a venue like Universal Amphitheater doesn't own a sound system is because everybody has equipment and anybody can buy equipment; but what they really need is a sound service to make the equipment do something creative. That's a whole different story. Stan says, "Universal could afford buy a sound system. Money is not a problem, as evidenced by the fact that they spent $23 million on the building. What is the problem is that they have to hire somebody to run it. Now, here's what happens: This guy comes and he works there for a year or two or three and then he gets another gig and goes on to another job. Now he has been their guru and he has told them 'this is the equipment I need and this is how I'm gonna do the job' and he leaves and the next guy comes and he says ' you gotta have this, this, and this.' In this case, we try to offer consistency from venue to venue and it happens that we have a very good man who has been at the amphitheater for several years. But over the period of years that we've operated, there has been more than one operator there. We still operate with the same kind of controls from up above that we did from day one. In other words, things haven't changed. We've learned what works, and the next guy that comes, if he's gonna be there, he is now working for us and he does it our way. So there is consistency. Our deal says that if the system breaks down, we fix it at our expense, it's what we're being paid for. They know how much it's gonna cost every month." ■

Sammy Caine

Alan Brewer

Our multi-talented subject is a record producer, movie producer, song writer, performer, film score composer and movie music director

MANY PEOPLE HAVE SPENT THEIR ENTIRE LIFE striving to be at the top of their profession. Trying to acquire the respect and admiration of their peers. In our industry, one wants to be "in demand." But usually, those that become successful are successful in only one aspect of their field.

This is not so with Alan Brewer. Brewer not only is a record producer, with credits including Rick Derringer, OMD, Rick Wakeman and Clarence Clemons, he is also a movie producer, songwriter, performer, film score composer and movie music director.

In the past, he has worked in many capacities on film and television specials. In 1980, Alan was the assistant to the producer on the horror film, *The Burning*. He has also done technical supervision and sound consulting on many projects.

His latest project is *Playing For Keeps*, which he co-produced. In addition, Alan served as Musical Director and co-producer of the soundtrack as well as taking the time to write and produce several of the musical tracks. We spoke to him during a brief work break at his home on the north shore of Long Island.

db: How did you get involved in so many aspects of music and film production?

AB: In various situations, people have spoken to me and discovered that my interests and in some cases my abilities are very varied. They realized that they could take advantage of my abilities. Somehow, I just seemed to be utilized in more than one area,

Producer Alan Brewer recording the vocals of Karen Sledge of the group Sister Sledge.

which seems to help create some consistency in the project.

db: What exactly were your jobs and responsibilities on the film *Playing For Keeps*?

AB: I co-produced the film itself, so I was involved right in the beginning from the point where it was being developed from an idea into a treatment, from a treatment into a screenplay and then into production. I was also musical director and co-producer of the entire soundtrack for the film and soundtrack album, and also writer and/or producer of several of the musical tracks.

db: What were your responsibilities as Musical Director?

AB: As MD, one of my responsibilities was to communicate to the musicians what it was that we wanted, and what kind of moods and concepts we were looking for in the scenes accompanied with music. I had a direct creative involvement in most of the music created for the film from the point before it was written through the recording and mixing of it, and all the way 'til it was mixed into the soundtrack of the movie itself.

I'll use Pete Townshend for example, who recorded and wrote "Life to Life," the main title song for the movie. I was with Pete in the studio at all times when the recording was being done so I could interact with him about whether the recording was going in a direction that I felt was appropriate for the movie. And, of course, I was there for the mixing of the film version so that I could have some technical input. Knowing that I would be at the final film mix, I knew how I wanted my materials laid out–how the tracks should be laid out before they were transferred to 35 mag stock and then ultimately mixed into the music pre-mixes of the movie. It went similarly for many of the artists. In some cases I had less involvement because of my schedule conflicting with theirs, and it would be impossible to be in the studio at all times with them. Sometimes I had a great deal of involvement, especially the songs that I wrote and produced, and in several cases performed on the track. But as MD, I was there to be an active participant in the creation of the music so that what we wound up with was music that was capable of being integrated into the film.

db: How about as co-producer?

AB: I was one of the producers of the film and actually functioned as the line producer on the set during shooting. And that had virtually nothing to do with the music or anything technical sound-wise. I was responsible for the organization of the shooting and crew, involved in every step of the writing process, which included interacting with the screenwriters and I was involved in the casting sessions. I was also there to be the central point of communications for organizational purposes in terms of casting, location selection, crew selection, budget approvals and so on.

db: To what extent were you involved in the mixing of the tracks for the movie?

AB: I was there to supervise all the film mixes and the album mixes, and in every case there were separate mixes done. That was another function that I performed as MD, even on tracks that I didn't produce. I still wanted to layout my material the way I felt more comfortable working with them once they got to the film mixing stage.

db: What was involved in the pre-production for the music tracks?

AB: I was very much in a "fly by the seat of your pants" situation most of the time. I didn't have the luxury of working with one act for ten or twelve tracks in the same studio, or even the same city, day after day. I'd say just about every one of the tracks were recorded in a different studio–and many in different cities around the world. I spent several days working with programmers and musicians in London for the Sister Sledge track.

db: So it sounds like there could have been some problems?

AB: In some cases I had to travel to wherever the producer and artist had decided to record. Several were recorded in and around New York. There was a song called "Make A Wish" that needed to be recorded properly so it could be lip-synched by the performers in the film. It had to be written and recorded at the correct tempo with the correct lyrics and as close to the final vocal as possible before we actually shot the scene. Naturally, since we were still in production on the movie, nothing had really been done in terms of writing and recording the music. And that was one where I had to sneak off the set where I was functioning as producer of the film, grab my guitar and hide in my hotel room for a few hours to write the song. I had to get the song written, and then sneak away for a couple of days to New York from Pennsylvania. Then I recorded the tracks, brought them back and got them approved by the director of the movie–since it would be performed on screen for 3 1/2 minutes. I then went back to New York to record and mix it well enough to be used as a playback tape on the set. It was all done with SMPTE so we could keep going back and forth from the master tape and wouldn't loose the tempo or lip-synch. Of course, it got to be a nightmare at times when you keep going through a variety of formats, hoping all the transfers and the time code would be done properly. But it seems to have worked out.

db: What considerations do you have when you are recording songs for different mediums, i.e. records, movie and television?

AB: One consideration is that you are not going to get the same high fidelity response in the movie theaters that you will get on a home system. You just don't have extreme high and low ends. It's a lot better in a Dolby Stereo format theater, but about half are not Dolby. To take an album mix that you feel is perfect for the record and use that same mix in a movie situation means that you are not maximizing the possibilities for the sound from the movie. Another consideration is equalization and trying to do what you can to get the right effect with the best clarity by equalizing the mix differently for the movie than you would for the record. It's important not to fool yourself as to how the music is going to sound. If you are in the studio just mixing for a record, you are going to be surprised at the way it sounds when you hear it playing on a movie screen. Dolby recommends that you monitor through their simulation monitoring box. You are not actually encoding the tape the way the soundtrack of the movie would be encoded and you are not using any special system. You are actually monitoring the music while mixing *as if* it were being encoded and decoded again; therefore, you can hear a close representation of what the song is going to sound like once it ends up on the optical soundtrack of the film.

db: How does it differ?

AB: You hear what happens in terms of your separation and how it is going to be interfered with, and it always is. You hear what is going to happen with your frequency response and you adjust accordingly. There is no point in creating a really hot sounding track that's just never going to be heard that way once people are sitting in the theaters. You just might as well accept what it is going to sound like and make it sound hot within that system. You have to try and get the track to cut without the extreme high ends, and get the track to feel strong and full without the extreme low end. You have to create as much separation as you can given the restrictions that you have. If you just do a stereo left/right mix the way most people would like to hear it on a record, it's going to sound a lot more mono once it gets to a Dolby Stereo optical soundtrack format.

db: So how do you handle differences in placement?

AB: All the songs were at least 3-track mixes and some were actually 6-track mixes. When there were 6-track mixes they were usually left/center/right for instrumental and left/center/right for vocal. That way when we were doing the final mix, we would have control over vocals vs. instrumental, which is required in scenes where there are lyrics that have to be woven in between dialogue or vice versa. It is also very hard to tell what will happen to your left/right separation until you are in that final mix stage, and it's more comfortable for me to be in control of the elements by having a left, a discrete center, and a right track. This allows us to retain as much separation as possible. What happens is that information that is on both the right and left tracks end up being summed and placed in the center. So you wind up with a true discrete playback in the center of the screen of what would normally be considered a phantom center on a regular stereo mix. Because of that, everything comes out sounding almost mono. So an instrument you might mix as left-center on regular stereo record mix, you would often choose to place extreme left. Or what you might place right-center or right-right-center, you would place extreme right, so that you can sense more of a separation. Everything on left and right on the final Dolby system winds up being partially in the center anyway.

db: So that is why you need separate mixes?

AB: Since you can't use this 3-track mix on a stereo record, you need a separate mix. And again to have control over vocals separately from the instrumental tracks, that creates another three tracks. What I will often do is I will put a totally dry lead vocal track in the center and then background vocals with effects in left and right, and sometimes the effects only from the lead vocal in left and right. That way I can have more influence over how much pure, dry lead vocals will be used in the final mix, and how it is placed against the background vocals. That is the kind of thing that really changes once it goes through the Dolby matrix. And certainly, it's important while doing those mixes to keep checking your mono mix because you have another somewhat uncontrollable situation where you might do a mix that you think is absolutely perfect for the Dolby stereo theaters, and then find that something really strange is happening in mono.

db: Does this introduce any special problems?

AB: You have to be very careful with any phase related effects because they get treated in a very funny way by the optical soundtrack and the encoding process.

db: When you mix do you consider that movies are ending up on home VCRs very quickly these days? That becomes a fourth playback medium to consider when mixing.

AB: If you have created a high quality stereo mix for theaters, then chances are you have a pretty good mix for home stereo video cassette reproduction as well. In fact, you can also get a 4-track decoder unit to simulate the 4-track playback you get in the theaters, even off your 2-track video cassette, and wind up with a discrete left/center/right and surround with decoding. Although you are in a smaller room, and your system might be capable of better frequency response, you will get a very similar effect in terms of balance and separation.

The other consideration, as it always has been for movie mixes, is what happens when it goes on TV through one little speaker. But that's something that you always have to contend with when you are doing movie mixes and music. What I've been describing is more specific to doing stereo mixes, but there are similar considerations when you are doing a mono mix for a movie soundtrack. But again, even when working in mono, I prefer to do a separate mix and maximize the tracks and get the hottest sounding mono mix, rather than to just take a stereo record mix and let it playback in mono. I feel you can always get a far superior result when you do a separate mono mix for that purpose.

db: I would like to talk about the process used to make the two mixes. Did you go in and record the cuts in a studio with the artists and then use that to make the movie mix?

AB: I would go back to the multitrack tape each time to create an entirely different mix for the movie soundtrack each time. The mixes were often 24-track analog and several were 48-track analog. All the mixing was done in one place. In most cases, the tracks were mixed in the same studio as they were recorded and around the same time the original recording was done. In most cases both mixes were done about the same time.

db: What was the order of operation?

AB: First, we did the movie mix, and then we moved on and did the album mix. Sometimes it would save time to move right ahead and go from one to the other without having to set up again at another time and another place. Sometimes that wasn't possible so I was most concerned with coming away with the movie mix first, because the deadlines for getting the movie mixes were earlier. They had to be transferred to 35 mag recording stock and cut into the edit of the movie by the music editor, then worked with by the picture editors. While that was all being done, the album mixes could be done. In a few cases additional overdubs were done and a few other touchups for the record version, either because the record version was going to be longer than the version that was being used in the movie, or because a slightly different effect was desired in the movie. So for a typical song, I recorded the tracks in a studio, then I went ahead and did my 6-track film mix and then went right on to do the 2-track record mix in the same studio.

db: How were the 6-track mixes done?

AB: Sometimes the 6-track mixes were done straight to an 8-track machine by regenerating the original time code that was laid down onto track 8 of the master . We then mixed left/center/right instrumental on to tracks 1, 2 and 3 and left/center/right vocal on to tracks 4, 5 and 6, using track 7 as a guard track. Then we transferred from the 8-track to two different 3-track 35 mm magnetic rolls. When we were in the final film mix stage or at least doing our music premix, we could be running music and vocals separately on two different dubbers. Often we would create changes for ourselves in the editing room to make the mix easier. For example, if we knew that the first verse would be vocal, then we would have an instrumental verse (because there was dialogue or sound effects that had to be in the clear) then back to vocals in the chorus. We would sometimes chop out that second verse on the vocal reel so that as it was running in the mix, it would be less work for the mixer.

Sometimes instead of recording straight to 8-track, we would get our mix set the way we wanted it and then run a 4-track–this was a decision based mostly on availability of equipment–regenerate the time code onto track 4 and mix left/center/right instrumental onto tracks 1, 2 and 3 and have all the vocal tracks muted. Then we would lay down the same time code and just run the vocal tracks onto another part of the tape. We then transferred from those two pieces of 4-track to the 35 mm mag.

db: What problems did you have when using different studios?

AB: It gets very difficult sometimes when you are trying to record every piece of music for the movie in a different studio. And some studios just aren't technically prepared for the time code arrangements that you need or the multitrack final mixes that you need. No studio that I know of is set up to monitor through the Dolby 424 monitor box, so usually it would be a very extensive period of preparation to set up the studio for time code and monitoring systems. I had to monitor the movie mixes through a 4-speaker system, left, center, right and rear. Those would be four separate tracks with four separate and distinct signals coming off the monitor box. Another complication is that very few music studios are set up properly to view video with time code while running the multitrack tape machine in synch. In some cases, I had to visit the studio days in advance in order to make sure we would be able to record to picture. In some cases, we were recording music for dance scenes, and even if you can get away with recording some songs without actually watching the scene, you can't do that in the dance scenes. You have to make sure that the tempo is locked in and the movements are correct with the accents in the music, and in a few cases, we shot dances scenes with a song that we later changed, and we had to try to retain the precise tempo.

db: What about processing?

AB: We didn't use the same processing on all the tracks. We laid down the tracks clean. As far as equipment, it would have been impossible to use the same pieces of equipment on each track. If I decided that there was an effect that I wanted to use all the time, it wouldn't have always been available. There really is no one consistent process or piece of equipment that you would find on all the tracks.

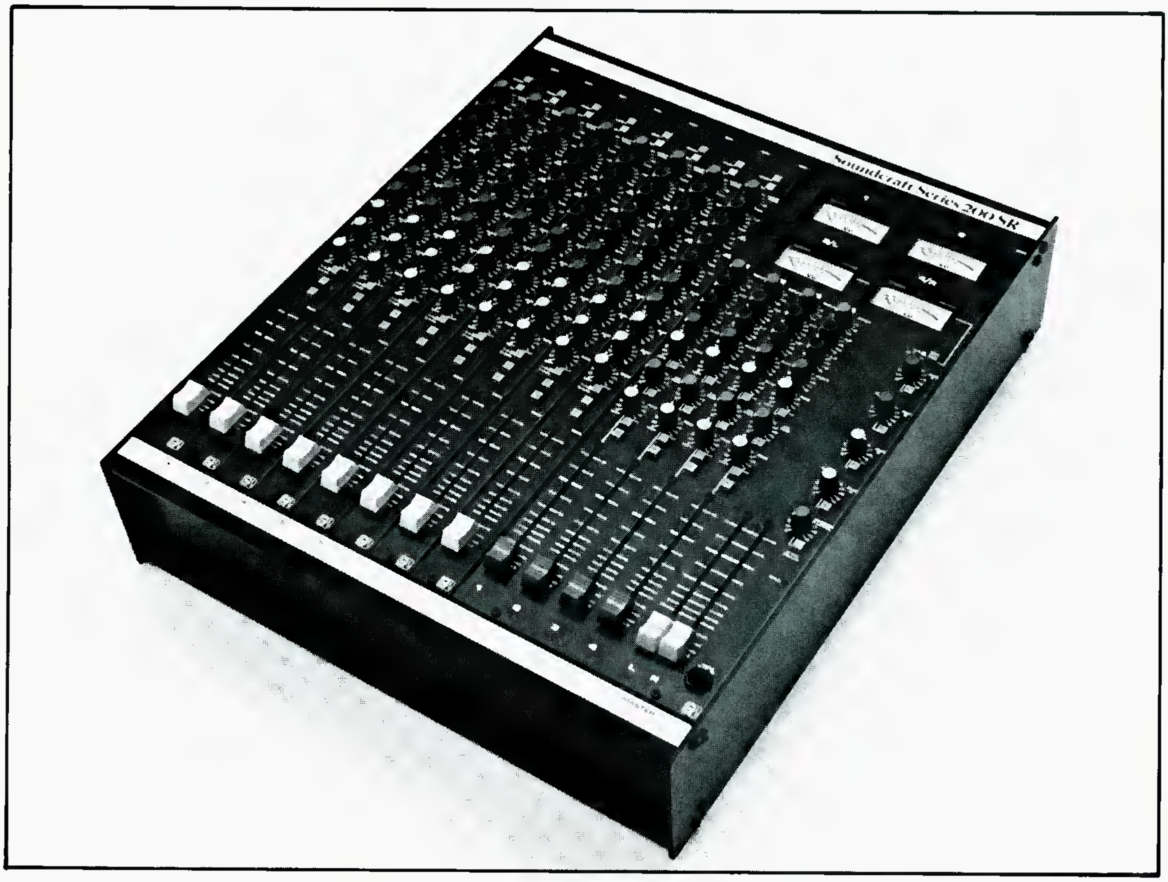
db: Were there any specific techniques that were used while working on the film?

AB: There were situations where I normally would choose a real drummer for the basic tracks, but I chose a drum program because it was easier to control the tempo to what was actually going on on-screen. Even in a case where there may not have been dancing going on, you might need an exact length of a piece of music. It is more easily achieved by using electronics than by using a real player. In that way, my production might be different because I'm recording for a movie. ■

Len Feldman

Lab Report

Soundcraft Series 200SR Mixing Console



General Information

The Soundcraft Series 200SR, designed and built by the British firm of Soundcraft Electronics Ltd. and distributed in the USA by JBL Professional in Northridge, California, was designed primarily for public address and monitoring applications. The console has four main buses, four auxiliary outputs and dedicated mix outputs. All inputs are electronically balanced rather than transformer coupled. Soundcraft maintains that this approach offers improved transient response and minimal phase shift. Outputs are "ground compensated" to minimize hum or external interference.

Microphone input impedances are approximately 2000 ohms, while line level inputs have input impedances greater than 10,000 ohms. There are four VU meters which read the "Group Outputs" 1 through 4. However, the third and fourth meters can be switched as a pair to read the monitor source which may be either the stereo mix bus, the 2-track return signals or any pre-fade listen (PFL) or after-fade listen (AFL) signal.

Unlike most small consoles we have tested recently, this one is powered by a completely separate external power supply unit which is connected to the console by a long multi-conductor cable. The power supply provides the required positive and negative 17 volts of DC as well as a +48 volt phantom power supply for use with mics that requires such a phantom supply. The console is designed to work at a level of +4 dBu and is available with 8, 16, or 24 input modules. There is also an optional rack

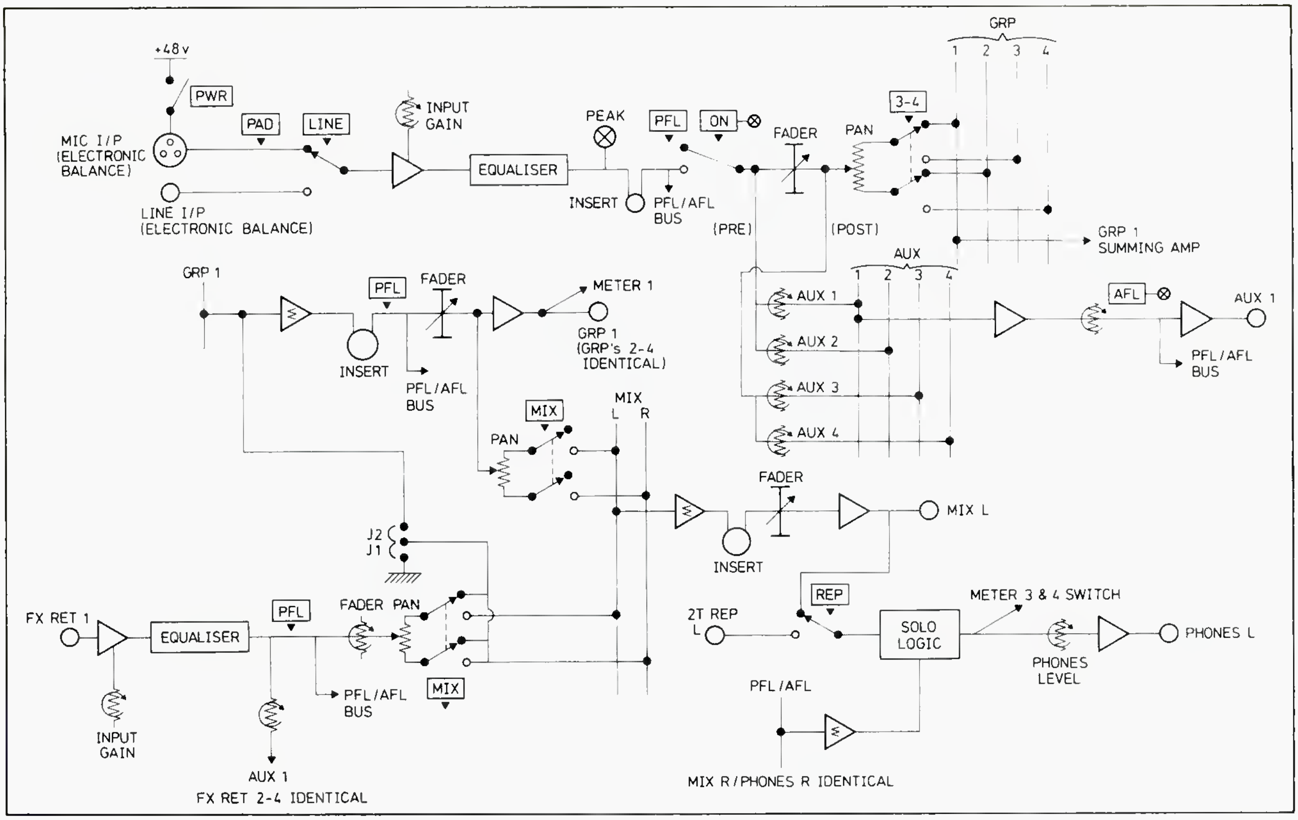


Figure 1. Signal flow diagram.

mounting version with a maximum of 8 input modules which is the version we tested for this report. A complete signal-flow diagram for the 200SR Mixer Console is shown in *Figure 1.*

Control Layout

Each of the eight input modules supplied in our version of the 200SR is divided into five logical sections: A channel input section, an equalizer section, the auxiliary section, a routing section and a channel status section. In the channel input section a "Line" pushbutton selects high level line input, attenuating nominal mic input gain by 30 dB. When not in the line level position, there's another attenuator button for reducing mic inputs by 20 dB. A variable control allows input gain to be varied over a total range of 40 dB, (from 20 dB to 60 dB of gain).

The equalizer section on each module is a 4-band fixed frequency design. The controls, each of which provide approximately 15 dB of boost or cut, are designated as HF, HI MID, LO MID, and LF. The Low Frequency and High Frequency controls provide a shelving type of boost or cut, while the two mid-frequency controls offer their maximum boost and cut at around 300 Hz and 5 kHz.

The Auxiliary section of each input module consists of four auxiliary "send" controls available for use as echo, foldback or other auxiliary effects units. The routing section allows channel inputs to be routed to either Groups 1-2 or Groups 3-4. The pan pot in this section is center-detented and introduces a loss of 4.5 dB when at its center point. This is a compromise between the 3 dB loss needed for constant power panning and the 6 dB loss that would be required for constant voltage level panning. Pressing the ROUTING button in this section routes the channel input signal to Groups 3 AND 4. By panning left, the signal can be routed to odd numbered groups while panning right routes the signal to even numbered groups.

Finally, the "Channel Status" section of each input module includes a channel "ON" button with indicator light, a PFL (pre-fade listen) button which "solos" the signal from that module on the monitor/headphone outputs along with any other modules that have the PFL button depressed, a PEAK LED indicator which illuminates at 4 dB below clipping and a long-throw linear fader. I

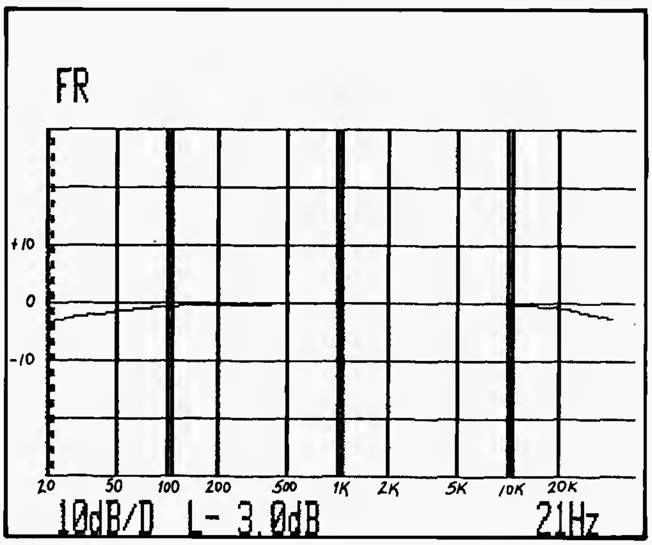


Figure 2.(A) Frequency response to 21 kHz.

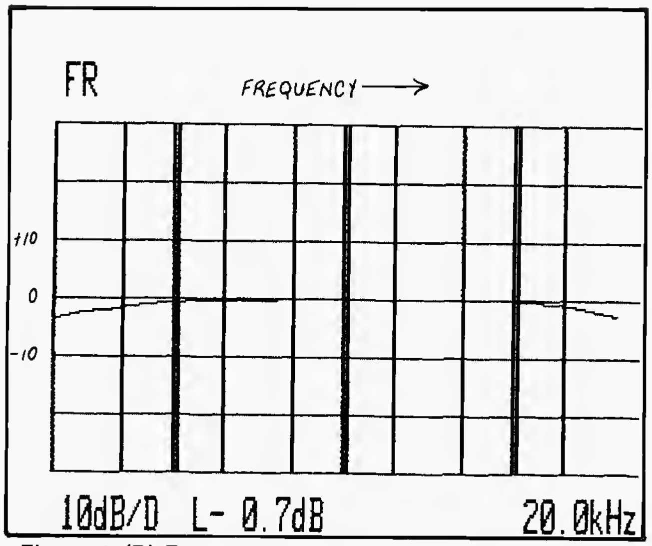


Figure 2.(B) Frequency response to 20 kHz.

might mention that this fader is one of the smoothest acting that I have encountered. Furthermore, its infinity cutoff is greater than 90 dB.

To the right of the eight input modules is the Master module which contains the four VU meters, the Group/Returns sections, the Auxiliary masters and the headphone monitor section. The four VU meters normally monitor the outputs from the four groups or the FX returns, but pressing a button labeled MNTR changes the functions of the third and fourth meters which can then be used to

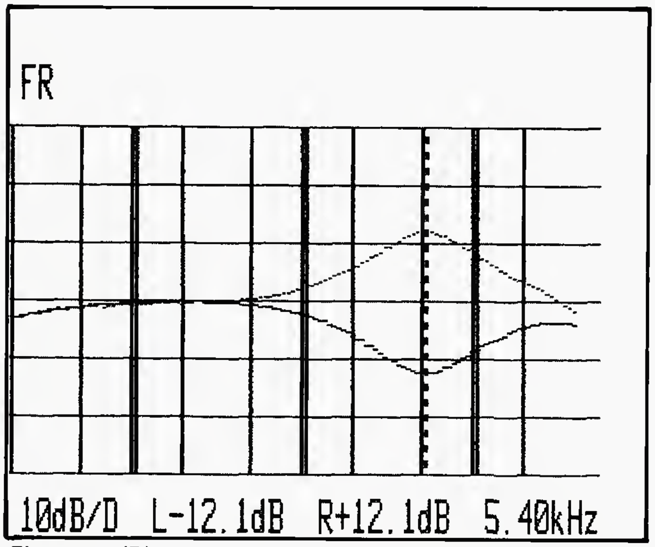


Figure 3.(B) Boost and cut range of the high mid control for the equalizer section of the input modules.

monitor the main mix bus or the Auxiliaries.

The Group/FX returns section consists of an input gain control, a 2-band equalizer section with turnover frequencies at 10 kHz and 60 Hz, an auxiliary send, a pan control which allows the FX return signal to be panned to the correct position in the stereo mix when the MIX button is pressed, a MIX button, a rotary fader control for the FX signal and a PFL button which solos the FX return signal. Four long-throw linear faders control the overall group output levels.

At the extreme right of the console is the auxiliary master section. Each of the four auxiliary buses has

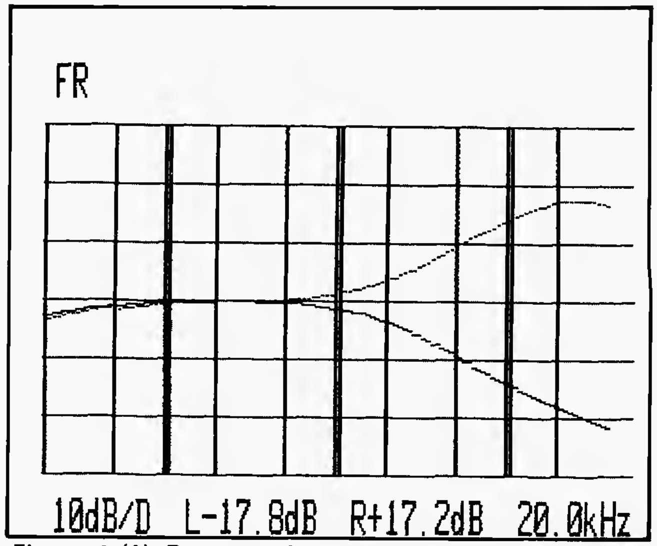


Figure 3.(A) Boost and cut range of high frequency control for the equalizer section of the input modules.

its own master level control and an AFL select. The AFL works much the same way as the PFL bus, i.e. AFL solos the signal from the auxiliary master on the headphone output along with any other PFL or AFL selected by the operator. A headphone jack and a phones level control are also located at this end of the console. The phones are normally fed from the main mix bus, but may be fed from the output of a 2-track tape machine by pressing the 2-track replay button. One limitation imposed by the phone amplifier is that it must drive phones having

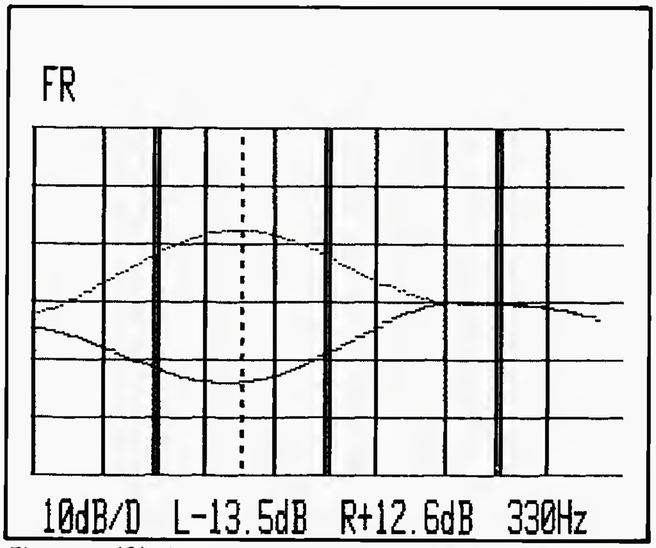


Figure 3(C) Boost and cut range of the low mid control for the equalizer section of the input modules.

an impedance of between 200 and 400 ohms. The output devices used for this circuit are not suitable for driving low-impedance phones or loudspeakers.

Input and output connection panels are found on the rear of the unit. In our version of the console, the entire console must be mounted vertically in a rack. If it were to be oriented horizontally, the unit would lean on the input and output cables and connectors. Some users who are used to working in a horizontally oriented console may find this objectionable, but we found that we could prop up the back top end of the rear of the console in such a way that it could also be placed on a flat table or desk

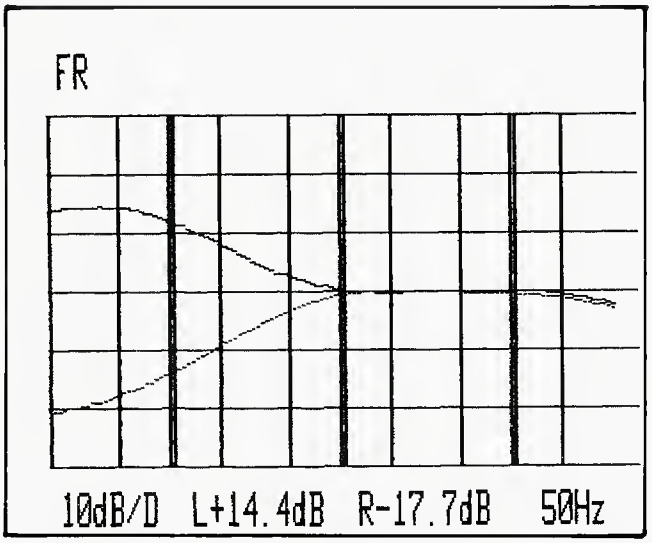


Figure 3.(D) Boost and cut range of the low frequency control for the equalizer section of the input modules.

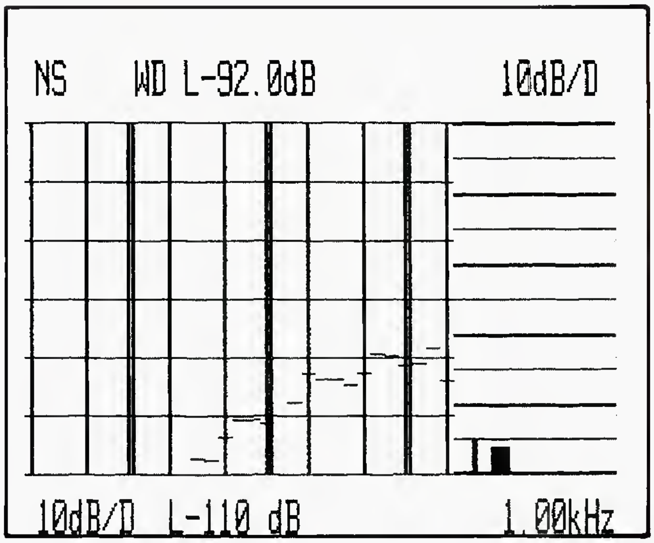


Figure 4. A-weighted signal-to-noise ratio analysis.

and have a slope upwards so that the cables and connectors would not be under strain. The input panel contains eight XLR mic connectors, eight ring-tip-sleeve phone jacks for high level line inputs, and an equal number of phone jacks for insert send/receive signal connection. Associated with each mic connector is a tiny switch which, when activated, provides the +48 volt phantom power. Incidentally, the XLR wiring puts Pin 1 at Ground, Pin 2 at "Hot" and Pin 3 at "Cold". Since this may not always correspond with your mic cable wiring, you'd best check before connecting any mics to these inputs, especially if you've got the phantom power voltage turned on! As for the insert jacks, each carries both the insert send and the insert return signal. The former is connected to the "ring" terminal of the jack while the latter is connected to the "tip" terminal, with the "sleeve" used as a common ground.

The output panel consists of the main mix outputs, which are in the form of XLR connectors, the four group outputs (also XLRs), the four auxiliary output jacks (ring/tip/sleeve phone jacks), four "Insert" jacks, four "FX Return" jacks, a pair of Mix Insert jacks and a pair of 2-track output jacks. The connector for the power supply cable is also found on the output panel.

Lab Measurements

A complete table of VITAL STATISTICS covering Soundcraft's published performance specifications as well as our own lab test results will be found at the end of this report. Frequency response for the mixer console, measured from a high-level input to either the auxiliary or main mix outputs was down 3 dB at 21 Hz and was down 0.7 dB at 20 kHz, as illustrated in the response plots of *Figures 2A* and *2B*. *Figures 3A, 3B, 3C* and *3D* show the maximum boost and cut range for the 4-band equalizer section of each input module. Signal-to-noise ratios, analyzed in *Figure 4*, measured an impressively high 92 dB below a nominal 0 dB output level as indicated on the system's own VU meters. With input faders set to their minimum positions, noise and hum was an incredibly high 110 dB below 0 dB reference level, A-weighted. Even with the A-weighting filter removed, S/N under these test conditions remained a very high 96 dB. Harmonic distortion at mid-frequencies for 0 dB reference level was only 0.007 percent, remaining well below 0.1 percent even at the frequency extremes of 20 Hz and 20 kHz. SMPTE-IM distortion was also quite low, measuring only 0.07 percent.

Comments

After examining its features and layout, we found the Series 200SR to be an ideal small console for live sound reinforcement work, either for providing stage foldback for the performing musicians or as a "front of house" mixing console in a small setup. When using the Series 200SR as a monitor console, all instruments would be connected to the console via microphones in the usual way. Each output can then be assigned to one or more monitor loudspeakers on stage. For example, the Mix L & R outputs might be used to feed the side fills while the Groups 1-4 outputs could be used to feed individual musician's monitors. The Auxiliary outputs could also be used if more than six monitor outputs are needed. Signals from any Effects devices are routed back into the console via the "FX" returns sockets on the rear panel, while the operator's monitor is provided by the separate "Phones" output.

If you want to use the system in a small PA setup, the main Mix outputs can provide the "sends" for the main PA. Effects Returns can then be routed to the mix bus by using the FX returns. If you wanted to use the Series 200SR to provide both the main PA sends and foldback sends, the main Mix outputs can provide the "front of house" PA while the Auxiliary outputs can be used to provide the foldback sends, taking the signal for the foldback from the auxiliary outputs. Using the Auxiliaries for the foldback sends in this way allows the Returns to be used for Effects returns on the main PA.

In short, the Soundcraft Series 200SR, despite its rather compact size as configured in our sampled version, offers a great deal of flexibility, extremely quiet and distortion free performance, and durable con-struction. We opened up the unit after completing our tests and found that much of the console is of modular construction. If servicing is ever needed, the layout of this unit would make it extremely easy to troubleshoot and repair.

VITAL STATISTICS

MAKE & MODEL: Soundcraft Series 200SR

SPECIFICATION	**MFR'S CLAIM**	**db MEASURED**
Frequency Response		
Measured at +10 dBu	-0.5 dB @ 20 Hz; -1.0 dB @ 20 kHz	-3.0 dB @ 20 Hz -0.7 dB @ 20 kHz
Harmonic Distortion		
Line Input to Group Out	0.01% @ 1 kHz	0.007%
Output Noise		
Group Output Bus Noise	-90 dBu	-92 dBu
Input Equalizer Range	15 dB boost/cut	See Fig. 2(A-D)
Peak Warning Level	4 dB below clipping	Confirmed
Max. Gain		
Mic In to Group Out	90 dB	92 dB
Line In to Group Out	60 dB	62 dB
Max. Output (all outputs)	21 dBu/600 ohms	Confirmed
Crosstalk (1 kHz/10 kHz)		
Between Group Outputs	-70 dB/-68 dB	-73 dB/-68 dB
Between L & R Mix Outputs	-65 dB/-60 dB	-65 dB/-62 dB
Dimensions (W x D x H, ")	19 x 20.9 x 5.0	Confirmed
Shipping Weight		
Console	combined	34lbs.
Power Supply	52 lbs.	12 lbs.
Price	$2295.00	

Circle 51 on Reader Service Card

Ed Learned

Jay Barth—Taming the American Storm

IN THE PAST 15 YEARS, Jay Barth has mixed for artists such as Kiss, Cheap Trick, Ted Nugent, and Rod Stewart. He is probably best known for his work with Bob Seger and the Silver Bullet Band, with whom he's been house engineer for over ten years. This interview with Jay was conducted in two parts: We first spoke at his home in Ann Arbor, Michigan in mid-April, before the release of Bob Seger's smash LP *American Storm*. We hooked up again at Merriweather Post Pavilion outside Baltimore, Maryland as the tour was in its very early stages.

db: There aren't a whole lot of guys, with the scene the way it is today, who can say they've been with one artist for ten years. How did you first hook up with Bob Seger?

Barth: It seems like I've done gigs with Bob all my life. When I was 12 years old, I worked with a band that opened for him at Notre Dame High School on the east side of Detroit. Later, I worked for the Catfish band in Detroit, and we played a lot of shows with Bob, Ted Nugent, Savage Grace, and all those other area groups. I got close to the Bullet band a few years later while I was working for Fanfare, a sound company out of Ann Arbor, Michigan.

db: What were you doing for Fanfare at that point?

Barth: I was house engineer, and at that point I was mixing Kiss. It was part of one of the worst atrocities in the history of booking rock and roll tours: Bob Seger opening a whole tour for Kiss! With them being the opening act, I got to know Bob and the guys real well. They had an English fellow as engineer, a guy who had worked with Bob for a number of years. He had some immigration problems, so in the middle of the tour they needed someone to mix. That was late 1975; I've been with them ever since.

db: You are in the process of planning for Bob's upcoming tour. What are some of your concerns when you first start planning for a tour like this?

Ed Learned is a mixing engineer for Aerial Enterprises. Recent acts he has worked with include Ashford and Simpson, and John Cougar Mellencamp.

Barth: The first thing I try to do is determine what the band will consist of and what we're going to have to deal with on stage. Another important consideration for me, being house engineer, is what type of sound system we will use, and whether it will perform to my liking.

db: Do you concern yourself with the type of speaker array you will use during the planning phases?

Barth: Yes, because one of the biggest problems I see in today's shows is that the PA is way up high and way out wide. People in the cheap seats are getting blasted, pulverized by horns, and people in the good seats are hearing mud: It's loud, they feel the bass, but they can't understand anything. I try to impress on everyone involved with the production the importance of point source. I stress the need to keep the stage as narrow as possible so the PA doesn't have to be out so wide.

db: Do you fly everything, or is there some ground support as well?

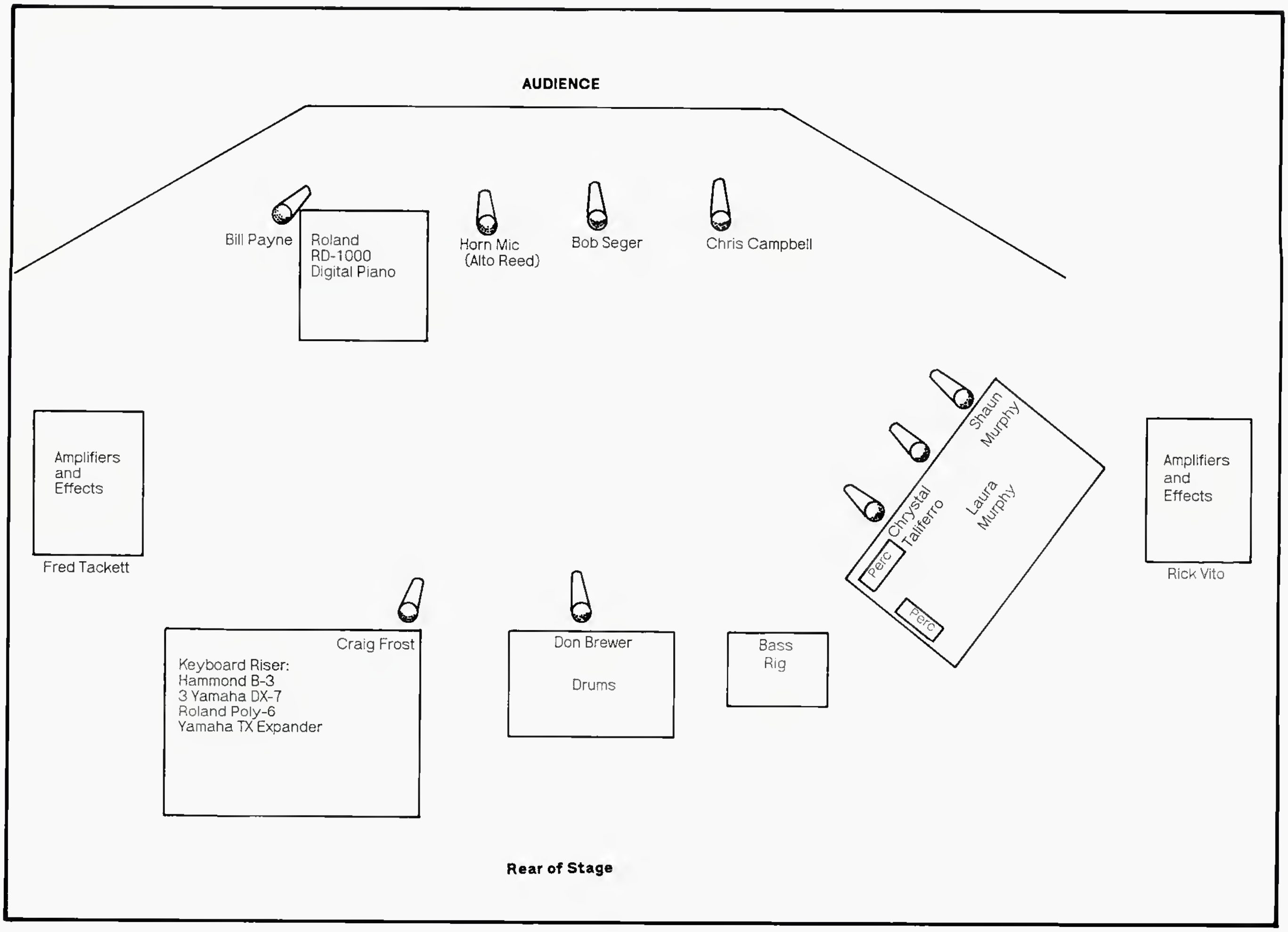
Barth: I'm totally against flying everything. I think it takes away from the sound on the floor. I'm into massive subwoofers, with massive stuff stacked on that, and massive stuff flown above that. I try to get the ground support stacks and flying system aligned vertically as closely as possible, to prevent phasing problems. I also try to keep the floor array curved, to prevent beaming in the various frequency bands. That can be a real problem with square boxes in square arrays.

db: Where do you request your mixing location?

Barth: I want to be no more than 90 feet from the stage and slightly off to one side, almost in front of one of the stacks. This allows me to hear the room ambience, but still have my direct source from the system and the stage.

db: What is your most important criteria for sound systems in a touring situation?

Barth: Even coverage. For every show that I do where I'm house engineer, I take the time to walk the seats. I want to insure even coverage everywhere I'm walking. That means using crossover levels or am-



The stage layout.

plifier gain controls to balance the system out. Often, we have the amp gains marked for rooms of different sizes–we might have to adjust structure for near-field coverage. I can tell the guys on the crew to take the third row down to the second mark, and they know right where that is. I do that every day.

db: Many engineers wouldn't bother.

Barth: That's right. How many times have you gone to a show and walked over to the side, where a horn aimed to the side is just–WOW!

db: Parting your hair.

Barth: (covers ears with hands) Owwww!

db: What area are you asking the ground support system to cover, and where does the flying system fill in?

Barth: For big arenas, 90 feet is about where you try to go with the floor stuff, although a lot of times some of it might be stacked sideways and angled up for side reaches that might be further than that. The flying stuff starts kicking in at 90 feet for sure. Ideally, if it is set up perfectly vertical to the floor stuff, it should couple together acoustically for the whole place. For really big places, I sometimes use a delay stack.

db: Where is that placed?

Barth: It depends on where the scoreboard is. I hang from that point, or I put it where I can. I set it with a click through both systems, and then I walk to the back to double-check the delay time. Back when I was with Fanfare, we did a computer study of how many tweeters it would take to get real highs to the back of the hall in a typical big arena. We came up with something like 24,000 tweeters (laughter). Using the delay stack makes a huge difference in getting coherent highs to the back of the hall.

db: Musically, what do you do to prepare for a tour that will support new material?

Barth: First, I listen to the tracks; I get a copy of the record sent to me. I try and make notes as to what's important as far as the live sound is concerned. For instance, a flanged bass part might not be as important to the song live as is a big drum echo. I try to determine what's *really* there, and what's sort of there.

db: Frequently, the live arrangement of the song is greatly different from the recorded version. How is that handled?

Barth: You listen to what the band's playing, and try to figure out what you need to use to capture the feel of the record. I sit down with Bob and the band during rehearsals to work out this stuff. There is a lot of communication involved. As far as recreating effects go, working for Bob Seger makes it fairly easy. If I need a certain digital reverberator, I make a call and I get that digital reverberator. Many bands don't have that luxury.

db: Any advice to those who do not have many effects devices?

Barth: Use your creativity and imagination. If you've only got one effect, think about it–plan it out. Where are you going to use it? Do I have to have the big vocal sound in this song, or is the exploring snare drum more important? During the chorus, could I switch the exploding snare drum and let the

Jay Barth setting levels during a soundcheck.

vocals go dry, then switch back for the verse? These are the types of questions you should ask yourself. Also, think about routing: is there a send for it, or will it have to be patched through a channel or a subgroup?

db: What are some of your favorite big arenas to play in as far as sound quality goes?

Barth: Rupp Arena in Lexington, Kentucky is a great sounding big hall. Capitol Center in Largo, Maryland, and Cobo Hall in Detroit are also really good. Poplar Creek Music Theater outside of Chicago is the best of the nederlander outdoor-type theaters. Some real thought went into that one. The fence on the back of the lawn is angled so there is no slapback into the hall. The roof tapers gradually up and out in a horn-type of shape, so the sound is evenly distributed throughout the pavilion and isn't constricted by a narrow opening at the back. Narrowing the opening traps the sound inside in what I call a clamshell effect, similar to Pine Knob Music Theater in Clarkson, Michigan.

THE SECOND PART OF THE INTERVIEW

db: Let's start today by talking about your mic'ing technique. What are you using on the drums?

Barth: We've got Don Brewer playing drums for us, someone you should remember from your Grand Funk days. He plays a straight-ahead Ludwig kit. For the kick, I use a two-mic system, utilizing a Beyer 88 and a Crown PZM. His front head is off, so I put the PZM inside the drum, sitting on a piece of padding. The 88 is positioned halfway in, midway between the center of the rear head and the shell of the drum. The PZM gives an awesome low bottom end, while the 88 provides the punch and top end. For snare, I've got two Shure SM-57 mics, one on top and one underneath. I make sure that the over and under mics are exactly equidistant from the drum, a mirror image of themselves, and put the bottom one out of phase. Without the phase reversal, you destroy the whole idea.

db: You have some type of miniature on the rest of the kit. What model are you using?

Barth: They're Crown GLM microphones. Those are condensers, with the wide frequency range you'd expect, but they can also handle 150 dB SPL with no distortion, so they're perfect drum mics. In sonic quality, they are very comparable to AKG 451s. Because they are smaller, it visually makes for a "clean" look on the drums without a maze of stands. I use cardioid elements on the hi-hat, rack tom, and one each on the two floor toms. I've got two omni-elements on the last tour; They sounded good but they were kind of cheesy, we were always sending stuff back for repair. The GLM seems to have the Crown quality that I'm used to, and so far they're holding up great.

db: Chris Campbell has been Bob's bassist for over 13 years. Do you take him strictly direct, or do you use a mic too?

Barth: I'm into using both. Chris has fairly healthy bi-amped bass rig, so what I do is use a pair of E-V RE-20s on the bass speakers, and "Y" them together into a single channel. That way I get both the highs and lows of his high rig. I've also got an active direct box as well, which sounds fantastic: It's what I predominantly use in the house. He's got a wireless system for his bass, which I like because it adds some compression that makes the bass sound even tighter.

db: You have two new guitar players on this tour. Tell us a bit about them.

Barth: Rick Vito and Fred Tackett are on guitar this time around. Both of them play guitar on the new record, and each plays electric and acoustic guitar live. Fred even plays trumpet on several numbers, so we have a horn section we've could never get before. I take the acoustic guitars direct with active direct boxes, and mic their amps with SM-57s. The amps are located on the side of the stage area, blowing across so everyone can hear them. They can also use wireless systems for both their acoustic and electric guitars.

db: I couldn't help notice there are more keyboards

ShowCo engineer Jeff McGinnis tunes the P.A. System.

than any previous Seger tour–no doubt adapting to keyboard-oriented 80s. I see that you no longer have the acoustic grand piano along. You are probably happy about that.

Barth: That's the hardest thing in the world to do: Get a grand piano heard over rock and roll band. Of course, I've got a lot more keyboard channels in the house now, because we have two keyboard players on this tour. *American Storm* is one of Bob's most keyboard-oriented records.

db: I assume Craig Frost is still around. Who is the new addition?

Barth: We've added Bill Payne, who is probably best known for his many years of work with Little Feat. He's a fantastic musician, and a real addition to the band. Mostly, he plays the Roland digital piano, which is MIDI'd to a Kurzweil. I use all three in the house via a direct box; I get the Kurzweil in stereo, and a single line on the Roland. Mostly I use the Kurzweil, with just a touch of the Roland added. The Roland is eq'd to try and boost the middle ranges of the piano that get kind of lost. The Kurzweil sounds very real...

db: I heard it while it was being checked, and it was pretty amazing–you would be hard pressed to tell the difference between a real grand and the Kurzweil.

Barth: It is real. But, strangely enough, that's the problem, because the musicians don't want it to sound real. They'll tell you they do, but by the time it's sounding like they want it on the stage, it's eq heaven. It's back to sounding electronic, different from a grand. I get my signal before any of that happens to it.

db: Does Craig predominantly play the keys on the upstage-left keyboard riser?

Barth: Yes, although he does switch with Bill on certain songs. There is a B-3 organ, Roland Poly-6, three DX-7s and a TX expander up there. That stuff is all direct, some passive and some active. There is a Leslie for the B-3 which I mic with a pair of Sennheiser 441s.

db: Alto Reed and his saxophones: What are you using on him?

Barth: His saxes are all mic'ed with AKG D58E mics. We tried all types of little mics: Iso-max, Audio-Technica, small PZM, the whole bit. We always seem to have problems with feedback. Some girl who played horn in a local Ann Arbor band told me about this AKG mic; she said she never had trouble with feedback, and it seemed to work well with her horn. We tried it, and it works great. It had the most feedback rejection, and being a dynamic, it didn't have the brash, steely quality to the highs that we got with the condensers we tried. The mic really rolls off around 150 Hz, so it requires a lot of low-end tailoring, more eq than I'm used to using, but for close-mic'ing a sax it does great job. Just another case of the cheaper mic doing a better job; everyone seems to use them as talkback mics.

db: You've got three background singers and some percussion gear on the upstage-right riser. What do you have up there?

Barth: There are three singers: Shaun Murphy and Laura Creamer, who were with us last year, and Crystal Taliferro, who joined last year. Crystal is not only a great singer but also plays percussion and is an excellent saxaphonist. I use Beyer 88s for the three vocals, and for Crystal's congas and mounted toms. I have an SM-57 up there for her saxes; she plays alto, tenor and baritone. Together with Alto and Fred Tackett on trumpet, we can have a three-piece horn section, which gives us a lot of flexibility. All three girls also play miscellaneous hand percussion from time to time.

db: What about Bob?

Barth: Bob has both acoustic and electric guitars, which I handle like Rick and Fred. Vocally, I use a Beyer M-600 which is a dynamic mic. It's my favorite mic for vocals: It's got the best feedback rejection and best overall sound of any vocal mic I've ever used. Bob uses a wired mic–he doesn't care for the compression that goes on in the transmitter of a wireless rig. It doesn't really bother me, because I put the squash on the vocal in the house anyway; I'd just use less. But he could feel the difference in the monitors, so we elected to go with the wire.

db: Are you using M-600s on the other vocals as well?

Barth: Yes, except for the girls. There are vocal mics for bass, drums, both keyboard positions, and a spare for Bob. By the way, there is also a downstage horn mic, which is an SM-57. That's used by Fred Tackett when he plays trumpet, although it's also used by Alto and Crystal on certain songs as well.

db: The effects rack you are using is extensive, from a Lexicon 224XL to four dbx 900 racks. I'd like to pick your brain a bit about what you use where. Start with the 224XL.

Barth: I'm bringing that back in stereo, and it is used predominantly for vocals, although I occasionally dump solos into it to wet things up a bit, depending on where we're playing. Tonight, when the people get in here it will dry up considerably, so I will utilize it for solos.

The stage system at Merriweather Hall showing flying and stage support systems.

db: Can you be more specific about settings and songs without giving away your secrets?

Barth: We don't want to be too educational (both laugh). Generally, I use hall programs, and I try to tailor them to the hall that we're playing in. A lot depends on how much reverberation is going on in the hall itself. It usually ends up being about half as much as you would use if it was going straight to tape–with the venue reverb added, it ends up being about the same as the record. It's a matter of day-to-day experimentation. I also use a plate program for rock and roll numbers like "Making Thunderbirds" or "Old Time Rock And Roll". I give those a 50's slap with a plate reverb. On ballads, I use a hall with a slightly longer decay time.

db: What about the Yamaha SPX-90?

Barth: That is also brought back in stereo. I use it as an effect on the snare, and when I can get away with it I use the auto pan for certain solos. I use a long hall setting on the snare during "Like a Rock" and "Her Strut". There were a couple more, but we stopped doing those songs.

db: How about the Yamaha REV-7?

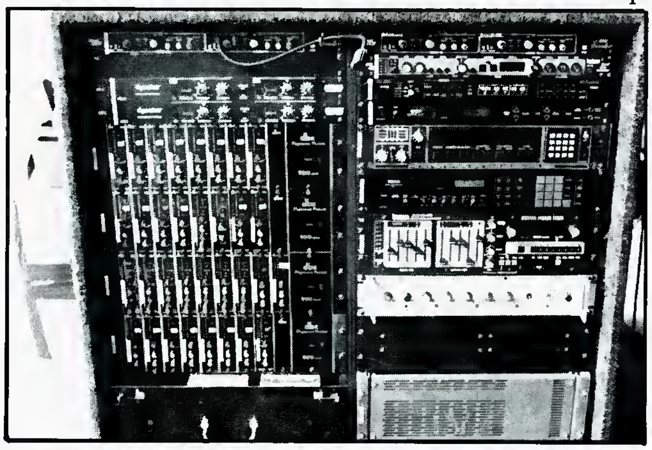
Barth: Again, that is brought back in stereo. I use the percussion plate program, and it is always on the drums and Crystal's percussion to wet things up.

db: On your console layout, I noticed that you have your AMS reverb coming back, in stereo, right in the middle of the drum inputs immediately after the snare mics. What is that for?

Barth: The AMS is always on the snare, using the non-linear program. You could also use the reverse gate, but I like the non-linear better. It has a much denser sound than the SPX-90, which I use only as an effect. The Yamaha units are excellent, but when you a-b them next to an AMS, you can see why one costs more money. The AMS is a much thicker and richer sounding reverb.

db: For delay lines, you've got a Roland 3000 and a Lexicon Super Prime Time. Where are they used?

Barth: I use both sparingly. To tell the truth, I could probably get by with only one. I've got a couple of programs in each one that I've saved, for vocal effects mostly. This isn't a heavy effects type of band. Take Cheap Trick: If I'd had this rig with them, I'd have had this stuff going all the time, and really lived it up. For Bob, there's just a little echo here or there. At the end of "Main Street" I dump Bob's voice into it and have a fall-off echo, that sort of stuff.



The house effects rack features Yamaha REV-7 and SPX-90 units.

db: What type of effect is this GELF that I see on the layout?

Barth: It's an antique phasing unit made in England. I haven't figured out where to use it yet, but I saw it in the shop and told 'em to throw it in my rack. I love 'em! It's got the deepest phasing sound you can imagine, a real "Itchypoo Park" kind of deep phasing effect.

db: Tell us a bit about your use of the limiter and gates in the dbx 900 racks.

Barth: All the individual vocal mics have their own limiter, channel patched into the respective console input. I also limit Fred and Rick's electric guitars, Alto's mixer send, and the bass mic and direct channels. Each drum channel is patched through a gate and then into a limiter.

db: I like your idea of gating and then limiting the drums. Do you feel this is an advantage to your mixing?

Barth: With those two units I can control the dynamic range of each drum. I can let the mic open or close as fast as I want, and when it does open up, I can control the ceiling that it goes to. As a result, I can fit the drums into any space on the dynamic scale of the band. You can always get a big, rich drum sound without it over-riding the rest of the band. The gates are set real tight, so that the drum that is on will be the only thing that will open it. It helps a lot with noise rejection and preventing over-rings on other drums.

db: You mentioned limiting Alto's mixer send to the house. His rack of stuff is offstage left-how does that work?

Barth: Freddy can tell you. (Enter Fred Christiani, sax technician.)

Christiani: Alto uses a Yamaha wireless system on his horns. These signals feed into the mixer that is part of his off-stage rack. We use the effects bus on the mixer to feed his effects–we've got two SPX-90s and a REV-7. Depending on which horn he's playing, I turn up the appropriate channel to feed the effects. We have programs entered that we just step through in order during the show. Mostly we use the SPX-90s, although the REV-7 is also used, sometimes as a reverb effect and sometimes as a chorus effect. His board sends a mix of dry and wet saxes to the house. The downstage horn mic also runs into this mixer, although it also goes straight to the house before the sax mixer.

db: Jay, how do you like ShowCo's Harrison house console?

Barth: It's a pretty decent desk. With the expander, it ends up being a 48 x 16 x 2 configuration. The eq is fully parametric, and I really like the fact that it's got sixteen effects sends on each channel. There are eight VCA groups as well as eight stereo submasters, and there are eight programmable mutes that can be assigned to both inputs and submasters. I wish the submasters had effects sends- that way, I could send all the drums or all the vocals to an effect instead of having to do it with all those separate knobs on inputs.

db: What do you have assigned to the submasters and VCA groups?

Barth: I use two subs for bass and drums, and one each for keys, guitars, piano, alto, effects and voice. I break these down further with the VCA groups. For example, I use one for Bob, one for girls, one for guys, so I have better control of the vocals within the vocal sub. And I might mix them up on "Old Time Rock And Roll," and I put Chris' vocal into the girls VCA because he's the only guy that sings. That way, I've got all the background vocals on one fader. I'm also using VCAs for kick, snare, and cymbals, basically because they're at the far end of the board which I can't reach that easily. The subs and VCA masters are in the middle of the console. That's pretty much where I do most of the mixing during the show.

db: What is your usual soundcheck procedure? That is, where do you start with your balancing, and what order do you go in?

Barth: I always start with the drums and then the bass, to get the bottom happening. I believe you start from the bottom and work up. After that, you could go to just about any of the "middle" instruments. I play guitar, so I usually go to those next–they're much easier than keyboards. Keys and horns fall in after that, and I do the vocals last. Once the PA gets tuned I find I don't have to alter eqs drastically at all.

db: In the Silver Bullet Band, you've got two keyboard rigs and two guitars going almost all the time. Keyboards and guitars together can pose a mixing dilemma: How do you deal with reproducing each part without it masking another? Do you use any special techniques to deal with this problem of separation?

Barth: First, I think it's a question of level, as far as what's a piano song and what's a guitar song. If one stands out, the other will be somewhat masked, yet you do have to get it to the point where you can sort of hear it. It's a matter of really fine-tuning levels, although you can help matters with a very slight amount of panning. You can't get too drastic in these situations, but you'll find going from 12 o'clock to 1 o'clock or 11 o'clock on the pan pot will separate things immensely.

db: You did something during sound check which I thought was pretty smart; you turned down the master volume of the PA. Could you explain why you did that?

Barth: In an arena with a hanging PA, you've got seats well below the speakers, with nothing but air in front of the band for quite a distance. In this place, the seats start right at their feet and rise immediately. So what you've got is a parabolic reflector, composed of hard surfaces. The PA speakers are only 30-40 feet away, blowing into seats and bouncing back onto the stage. It freaks the band out–They go "Oh, it's so loud–I can't hear anything." And it's all garbled: the delay from the seats is like reverb, because each row of seats will slap it back at a slightly different time. I have to keep the PA low so the stage volume can overpower the slapback. Otherwise, the stage and monitor volume gets turned up, which only makes things worse. Tonight, we'll have a full house, so it'll dry right up, and the PA will be working hard again.

The sound system for Bob Seger's American Storm tour is supplied by ShowCo, Inc. of Dallas, Texas. I sought out both house engineer Jeff McGinnis and monitor engineer Peter Buess, and spoke with them about the enclosures and electronics provided.

db: After looking on the loading dock, I could see that you left some of the PA on the truck for this venue. Could you describe what your total house speaker compliment is?

Jeff McGinnis: I've have twenty-eight high-packs, sixteen low-packs, and eighteen sub-woofers. The extra high-packs are going to be used as delay stacks in the larger halls; it's something that Jay likes to do. I'm also carrying special cabinets for things like floor fill or special ground-support use.

db: You do have a lot of stuff down on the deck here. The coverage demanded by this venue doesn't really require much in the air.

McGinnis: That's right. This isn't the greatest place to showcase the PA because I'm not using all of it; it's primarily designed to be a flying system. In a situation like this, where you have to stack more PA, it really doesn't sound as good as it could. Most of the venues we play on this tour won't be like this one; the majority of the cabinets will be in the air.

db: Can you describe the components in your cabinets, starting with the flying system?

McGinnis: It is a two box system, with added sub-bass cabinets that stack on the floor. The sub-bass cabinet contains two JBL 18-inch woofers in a front-loaded, ported cabinet. The flying low-pack consists of three JBL 18-inch woofers, also in front-loaded, ported enclosure. The high-pack contains four JBL 12-inch woofers, two JBL 2445 compression drivers mounted to two JBL Bi-radial horns, and four Yamaha compression tweeters. The horns are mounted vertically in the enclosure. If the cabinet stands upright, the horn flare is really 60 degrees horizontal by 90 degrees vertical, instead of the other way around.

db: In flying design, turning the horns over is a great idea. It insures good high-end coverage of the floor areas from the air. It would seem to require very careful focusing of numerous cabinets to prevent dead spots horizontally.

McGinnis: In our standard array of four across, we seem to get very even horizontal coverage in most situations. We've gone a step further towards controlling dispersion with these particular flying cabinets. The eight flying high-packs have separate inputs to each midrange horn; I can turn down the lower horn without affecting the upper horn within the same cabinet.

db: When I first came out front today, I noticed that you had the crew turn down the outside lower horns in the flying arrays. What was your logic behind that?

McGinnis: The flying high-packs will do a very effective job of covering the pavilion area all the way to the back if I run at a certain level, which would probably kill the people in the near seats. With the separate horn drive capability, I can turn down the lower horns close to the crowd and still get the level I need to carry to the far seats. We feel it's an effective answer to the question of needing different levels between front and back. Eventually, all our flying cabinets will have this capability.

db: You have another type of enclosure for your center fill here. What do you have in those?

McGinnis: Again, we utilize a two-box system. The low-end cabinet is a single 18-inch JBL woofer in a front-loaded, ported cabinet. The high-end cabinet contains two 12-inch JBL woofers, a single 2445 on a JBL Bi-Radial horn, and a single tweeter.

db: What type of power amplifiers are you using, and which components do they power?

McGinnis: ShowCo has been a Crown company for years. I've got PSA-2 and Micro-Tech 1200 amps. The PSA's power all the cone speakers, and the 1200s power all the 2-inch drivers and tweeters. Our total system power is around 40 kilowatts.

db: What do you have in the way of system drive electronics?

McGinnis: We run in stereo, using Klark-Teknik stereo third-octave equalizers and ShowCo custom crossovers. The crossovers have separate level and mute controls for each frequency band, each with individual metering and clip lights. The system is a real four-way electronic, with separate sub-woofer capability. I also use a dbx 900 rack for system limiting; the 903 limiters are channel-patched at the individual crossover sections, so I have separate limiting control over each frequency band. We have a Crown BDP-2 analyzer that is used for system tuning.

db: How about the console and electronics used for the monitor system?

Peter Buess: It's a Harrison monitor console, 32 x 32. It has sixteen discrete mix pots that are switchable between busses to get the 32 out. For instance, a knob could be assigned to #1 or #17. I have Klark-Teknik third-octave equalizers on every mix, and I also use ShowCo crossovers, three or four, depending on use.

db: I counted at least four different types of monitors on stage, from standard ShowCo wedges to side fills. Could you briefly describe each?

Buess: The basic wedge contains a 15-inch woofer, 2-inch JBL compression driver on a custom ShowCo horn, and a tweeter. I have a big wedge with two 12-inch woofers instead of the single 15. I have two fill cabinets: one has an 18-inch woofer, two 12-inch woofers, a 2-inch on a ShowCo horn, and a tweeter; the other contains two 18-inch woofers, two 15-inch woofers, a 2-inch on a ShowCo horn, and two bullet tweeters.

db: What power amps do you use on the monitors?

Buess: I've got the same ones; the PSA-2s run the 18-inch and 15-inch woofers, and the Micro-Tech 1200s run the 12-inch woofers and high-end stuff, although I sometimes use them to run the 15-inch speakers in certain mixes.

db: I'd like to get a general idea of where each of your mixes are going on-stage, and what type of cabinet you use on that mix. Start with the vocals.

Buess: First, we have Bob Seger's vocal mix, which is downstage center. I use one of the dual 12-inch wedges on him, because it is a much smoother sounding monitor for his voice. Another mix is downstage right; that is a standard wedge for Chris Campbell's voice and a bit of guitar fill. There are standard wedges, each on a separate mix, by the downstage piano for voice, the upstage keyboard riser for vocal, and the drum riser for vocal. There are three more standard wedges on the upstage right riser: each of the girls has a separate mix. That's eight mixes for vocals, although depending on the person, there might be some other stuff added.

db: You have two sets of side fills out there. Why?

Buess: They are actually two discrete systems. I use the downstage pair as stereo side fills, and the upstage pair as stereo piano fills. I've also got a sax fill, which is offstage left, and an upstage keyboard fill, which is just behind the keyboard riser. I usually use one of the larger cabinets for drum fill in big halls, but since Merriweather is small, I've only got a wedge back there today. ■

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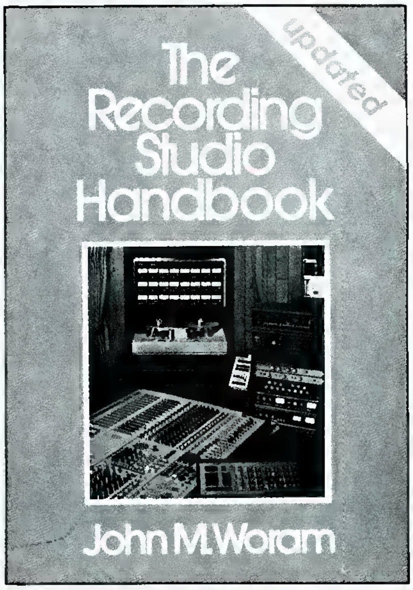
The Recording Studio Handbook



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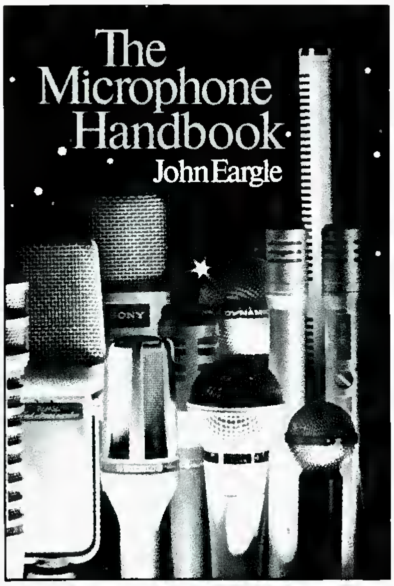

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Buyer's Guide:
Studio & Performance Speakers

This issue's Buying Guide covers professional monitor- and performance-type speaker systems. Our charts give a comparison of the most salient features that should be examined in regard to these products.

All the data comes from forms which we had sent to the respective manufacturers. Obviously, if a particular manufacturer seems to be absent, it probably means that we could not get the information by our necessary deadlines.

Finally, the Features column is where we invited the respective manufacturers to state what they felt was of particular prominence with their product.

Model	Dimensions, H,W,D	Cabinet Finish	Front Screen Finish	Impedance, ohms	Frequency Response +/- dBs	Bass Speaker Size, in.	Bass Speaker Type	Midrange Speaker Size, in.	Midrange Speaker Type	High-Frequency Driver, Size	High-Frequency Driver, Type	Crssover(s)	Weight, lbs.	Price	Features
ALTEC LANSING CORP.															
937	24x18 x16.3	black paint	black knit	8	70-15	12	cone			1	cone	3k	46	$603.00.	Vented 2-way monitor with 150 watt power rating.
AMR															
PRM-312	16x25 x13	walnut veneer	black	8	42-18 3	12	cone	6.5	cone	1	cone	300 3000	58	$499.00.	Built-in time correction, acoustic foam and uniform response.
PRM-310	14x21 x11.3	walnut veneer	black	8	44-18 3	10	cone	6.5	cone	1	cone	300 3000	40	$399.00.	Has time correction, acoustic foam, and uniform response.
PRM-308	12x18 x11.5	black vinyl	black	4	45-18 3	8	cone	6.5	cone	1	cone	300 3000	26	$299.00.	Same as above.
208	15x10 x8.3	black vinyl	black	8	68-18 3	8	cone			1	cone	2500	16	$199.00.	Same as above.
205	12x7 x8.25	black vinyl	black	8	79-18 3	5	cone			1	cone	2500	7	$99.00.	Same as above.
APOGEE SOUND INC.															
AE-5	23x14 x16	black polane	metal	8	53-17 4	12	cone	13x 5.5	horn			1320	76	$2,045.00	Front end processors used with all speakers.
AE-6	23x14 x15.5	black polane	foam covered	8 /4	53-17	12	cone	13x	horn 5.5			1320	73	$2,065.00	Includes full rigging hardware.
AE-2	32x10 x10.5	black polane		16	65-19 4	8 (2)	cone			3x5 (2)	horn	5000	38	$1,025.00	
AE-3	16x13 x10.5	black polane		8	70-19 4	10	cone			3x5	horn	5000	25	$620.00.	Processor circuitry with eq, protection circuitry, time alignment.
CARVIN															
3000E	24x42 x24	gray ozite	black metal	8	50-500Hz	18	cone						130	$399.00.	Folded-horn sub-woofer is highly efficient.
1200E	16x28 x22	gray ozite	black metal	8	160-4 kHz			12	cone				78	$269.00.	Acoustically tuned horn-loaded mid-cabinet.
1330H	32x28 x24	gray ozite	black metal	8	55-4	15	cone	15					108	$259.00.	Mid/bass bin with very efficient woofer design.
R-540 H	12x28 x24	gray ozite	black metal	16	1.2k-18		horn		horn				55	$299.00.	
820H	22x12 x17.5	gray ozite	black metal	8	70-20	12	cone			3x10	horn	1.4k	43	$199.00.	Compact and powerful full range speaker system.

Model	Dimensions, H,W,D	Cabinet Finish	Front Screen Finish	Impedance, ohms	Frequency Response +/- dBs	Bass Speaker Size, in.	Bass Speaker Type	Midrange Speaker Size, in.	Midrange Speaker Type	High-Frequency Driver, Size	High-Frequency Driver, Type	Crssover(s)	Weight, lbs.	Price	Features
850H	26x20 x14	gray ozite	black metal	8	50-20	15	cone			6x 16.5	horn	1.4k	56	$249.00.	Offers increased dispersion and projection.
960H	31x24 x15	gray ozite	black metal	8	55-20	15	cone			6x 16.5	horn	1.4k	80	$299.00.	Horn-loaded high efficiency design.
980H	32x24 x18	gray ozite	black metal	8	50-20	15	cone	6x 16.5	horn	3	twetr	1.4k 5000	100	$379.00.	3-way, full-range system.
750H	22x18 x13	gray ozite	black metal	8	70-20	12	cone			3.5x 10	horn	1.4k	50	$199.00.	Accurate monitor handles 200-300 watts with smooth response and controlled dispersion.
790H	26x20 x14	gray ozite	black metal	8	60-20	15	cone			6x 16.5	horn	1.4k	64	$249.00.	Has tuned port enclosure, high output and high power handling.
CETEC GAUSS															
7258	29x24 x19.5			8	40-18	15	coax cone					1.2k	116	$1,585.00.	
7228	22x18 x16.5			8	40-18	12	coax cone					1.6k	73		
EASTERN ACOUSTIC WORKS															
FR102	18x12 x8	poly	perf steel	8	70-16 3	10	cone			rad-ial	horn	3.5k	38	$340.00.	
FR122	24x15 x12.5	poly	perf steel	8	50-20 3	12	cone			2	hard dome	2.8k	53	$455.00.	
FR153	18x20 x25	poly	perf steel	8	40-19 3	15	cone	6.5	cone	2	hard dome	450 3.5k	75	$700.00.	
FR222	35x19 x19	poly	perf steel	4	40-16 3	12 (2)	cone			1	horn	1.5k	90	$780.00.	
FR253	41x25 x20	poly	perf steel	4	38-17 3	15 (2)	cone	6.5 (2)	cone	1	horn	450 3k	163	$1,280.00.	
JF500	50x25 x30	poly	perf steel	8	60-18 3	15	cone	10	cone	1	horn	250 2k	232	$1,750.00.	
KF550	33x54 x30	poly	perf steel	4	48-20 3	15 (2)	cone	12	cone	2	horn	250 1.6k	310	$3,690.00.	
KF850 T	42x27 x30	poly	perf steel	8	80-20 2	15	cone	10	cone	2	horn	250 2.2k	198	$3,300.00	
stage SM122 M		poly	perf steel	8	65-18 3	12	cone			rad-ial	horn	3000		$455.00.	
SM202		poly	perf steel	4	70-17 3	10 (2)	cone			1	horn	1.5k		$655.00.	
SM155 C		poly	perf steel	8	48-17 3	15	cone			1	horn	1k		$685.00.	
SM202 P		poly	perf steel	4	45-20 3	12	cone			2	horn	1.2k		$1,835.00.	

Model	Dimensions, H,W,D	Cabinet Finish	Front Screen Finish	Impedance, ohms	Frequency Response +/- dBs	Bass Speaker Size, in.	Bass Speaker Type	Midrange Speaker Size, in.	Midrange Speaker Type	High-Frequency Driver, Size	High-Frequency Driver, Type	Crssover(s)	Weight, lbs.	Price	Features
ELECTRO-VOICE															
FM-1502	22x23 x27.9	black carpet	steel	8	65-20	15	cone			1	CD horn	1.5k	72	$726.00.	300-watt, 2-way constant-directivity monitor with aligned, vented enclosure and durable construction.
FM-1202	20x20 x24.4	black carpet	steel	8	75-20	12	cone			1	CD horn	1.5k	65	$576.00.	Similar to 1502.
MTH-4	36x36 x30	black carpet	nylon cloth	2x4 8/8	160-20	10 (4)	CD horn	2 (4)	CD horn	1.25 (4)	CD horn	160 1.6k	367	$4,500.	3-way, active, horn loaded Manifold technology for high-level concert sound reinforcement.
MTL-4	36x36 x30	black carpet	nylon cloth	2x4	40-200	18 (4)	cone						263	$2825.00.	Low-frequency system for high-level concert sound reinforcement.
S-1803	36x28 x19.4	black carpet	steel	8	50-16	18	cone	6	vent	1	CD horn	600 4k	134	$1,194.00.	200-watt 3-way keyboard system with high acoustic output and efficiency.
S-1503	29x25 x13.8	black carpet	steel	8	65-16	15	cone	6	vent	1	CD horn	600 4k	105	$954.00.	Similar to above.
SH-1810	48x25 x24	black carpet	steel	8	35-20	18	cone	10	CD horn	1	CD horn	250 2.5k	153	$1,974.00.	300-watt, 3-way full range all-horn loaded frontal system for high SPLs biamp-able with tweeter protection.
SH-1810S	33x25 x24	black carpet	steel	8	35-20	18	cone	10	CD horn	1	CD horn	250 2.5k	112	$2,190.00.	Similar to above with mounting hardware.
Sentry 100A	17x12 x11.2	black vinyl	cloth coverd	6	45-18	8	radi-ator			1.5	dome tweeter	2k	28	$255.00.	High efficiency model with extended low range.
100EL	17x12 x11.5	black vinyl	cloth coverd	6	45-18	8	radi-ator			1.5	dome tweeter	2k	33	$524.00.	Powered system with 100A features.
500	24x27 x13	black vinyl	cloth coverd	8	40-18	12	radi-ator			1.5	dome tweeter	1.5k	70	$499.00.	For broadcast/recording uses. Has vented design and 50-watt amp.
505	20x26 x19	black vinyl	cloth coverd	8	40-18	12	radi-ator			1.5	dome tweeter	1.5	6	$499.00.	Similar to 500.
FOSTEX															
RM765	15x9 x8.7	matte		6.5	70-20 3	6.5	coax cone			2	printd ribbon	5k	14	$250.00.	Coaxial design for time and phase coherency and excellent imaging.
RM780	17x10 x10.7	matte		6.5	58-20 3	8	coax cone			2	printd ribbon	5k	19	$330.00.	Near field monitors that handle high power levels.
RM865	24x13 xs13			6	50-20 3	10	cone	6.5	coax cone	2	printd ribbon 5k	250	45	$399.00.	
SP7	5x7 x4.9	black paint	black metal	7.5	120-20 3	4	cone						5.3	$99.00.	
SP11	7x14 x8.4	black plastic	black metal	8	60-18 3	4 (2)	cone						12	$240.00.	Built-in power amp.
SPA11	7x14 x8.4	black plastic	black metal	8	60-18 3	4							17	$349.00.	
LS2	29x26 x20	black		8	45-20 3	12	cone		wood horn		horn	1.2k 7k		$1,649.00.	Studio monitors.
LS3	36x31 x29.5	black		8	38-20 3	15	cone		wood horn		horn	1.2k 7k		$1,999.00.	Has high efficiency, low distortion and wide dispersion.
RP 2001	51x21 x11			6	30-45 3	12								$1,750.00.	Time coherent design.

Model	Dimensions, H,W,D	Cabinet Finish	Front Screen Finish	Impedance, ohms	Frequency Response +/- dBs	Bass Speaker Size, in.	Bass Speaker Type	Midrange Speaker Size, in.	Midrange Speaker Type	High-Frequency Driver, Size	High-Frequency Driver, Type	Crssover(s)	Weight, lbs.	Price	Features
IMC (FANE)															
RE-112H	20x16 x16	Ross epoxy		8	60-20	12	cone					3.5k	32	$249.95.	Has stackable corners and horn protection network.
RE-112HC	24x18 x13	Ross epoxy		8	55-20	12	cone					2k	43	$349.95.	
RE-115HF	26x21 x15.5	Ross epoxy		8	55-20	15	cone					3.5k	75	$499.95.	High power handling capability.
RE-115FL	33x24 x24	Ross epoxy		8	60-18.5	15	cone					1.5k	90	$649.95.	Long-throw design.
RC-112HC	23x18 x12.5	carpet		8	55-22	12	cone					2k	49	$379.95.	Excellent efficiency to size ratio.
RC-115HF	26x21 x15.5	carpet		8	55-22	15	cone					3.5k	75	$529.95.	High power handling capability.
RC-115RF	28x21 x15.5	carpet		8	55-17.5	15	cone	6.5x 16.5	horn driver			1.5k	77	$649.95.	High performance.
RE-15HM	18x19 x27.2	Ross Durapoxy		8	85-20	15	cone					3.5k	54	$389.95.	Wedge-type stage monitor with volume control.
JBL															
Cont-rol 1	9.3x7 x5.7	black	metal	4	120-20	5	cone			1	dome tweeter	6k		$159.00.	2-way monitor with 90 dB SPL.
4312	24x14 x11.8	oiled walnut	black fabric	8	45-15	12	cone	5	cone	1	cone	1.5k 6k		$432.00.	3-way monitor with 91 dB SPL.
4406	16x10 x8.5	oiled walnut	black fabric	8	45-27	6.5	cone			1	dome tweeter	3k		$180.00.	2-way monitor with 87 dB SPL.
4408	17x12 x11.7	oiled walnut	black fabric	8	40-27	8	cone			1	dome tweeter	2.5k		$225.00.	2-way monitor with 89 dB SPL.
4410	24x14 x11.3	oiled walnut	black fabric	8	35-27	10	cone	5	cone	1	dome tweet	800	4.5k	$357.00.	3-way monitor with 91 dB SPL.
4412	14x24 x11.3	oiled walnut	black fabric	8	35-27	12	cone	5	cone	1	dome tweet	800	4.5k	$549.00.	3-way monitor with 90 dB SPL.
4430	36x22 x15.8	oiled walnut	blue fabric	8	35-16	15	cone			1	horn	1k		$1,260.00.	2-way monitor with 93 dB SPL.
4435	36x38 x20.4	oiled walnut	blue fabric	8	30-18	15 (2)	cone			1	horn	1k		$1,716.00.	2-way monitor with 96 dB SPL.
MARTIN AMERICA															
VRS-800	23x51 x26	matte black	steel mesh		35-20	18	cone	12	cone	1	horn	200 . 1.5k	229	$2,700.	Full-range horn-loaded sound reinforcement system.
RS-800	45x30 x26.5	matte black	steel mesh		35-20	18	cone	12	cone	1	horn	200 1.5k	269	$2,900.	Similar to above.
RS-1200	45x35 x26	black			35-20	15	cone	12	cone	2	horn		419	$4,600.	Similar to above.

Model	Dimensions, H,W,D	Cabinet Finish	Front Screen Finish	Impedance, ohms	Frequency Response +/- dBs	Bass Speaker Size, in.	Bass Speaker Type	Midrange Speaker Size, in.	Midrange Speaker Type	High-Frequency Driver, Size	High-Frequency Driver, Type	Crssover(s)	Weight, lbs.	Price	Features
MEYER SOUND															
MSL-3	21x57 x30	black gray	foam metal	4/8	75-20 4	12 (2)	cone	4 drvr (2)	horn 4 elect	pizo		800	265	$4,376.00	Proprietary drivers feature high efficiency.
UPA-1A	15x23 x13	black paint	foam metal	8/ 16	55-16 4	12	cone			3	horn	1.6k	66	$2,293.00	High SPLs with low distortion.
UM-1A	14x14 x22.5	black paint	foam metal	8/ 16	55-16 4	12	cone			3	horn	1.6k	66	$2,293.00	Similar to above.
UPM-1	7x18 7.2	black	foam metal	16	75-20 4	5	cone	5	cone	5 (2)	pizo elect	7		$900.00	Even dispersion with high SPLs.
500 Series x14	20x32 laquer	black		8 3	30-16	15	cone		(2)	10	horn	1k	110	$1,250.00	Amp included with system.
833	20x32 x14.7	satin black		8	35-18 3	15	cone			10 (2)	horn	700	115	$2,900.00	Wide dynamic range with damped cabinet.
PEAVEY															
3020 HT	38x36 x18	carpet	black metal	4	40-20	15 (2)	cone	10 (2)	horn cone	rad-ial	super tweet	400 1.2k 5k	167	$899.50.	4-way system.
SP-4	53x26 x20	carpet	black nylon	4	40-16	15 (2)	cone			rad-ial	horn	1.2k	120	$649.50.	
118	37x27 x20		black nylon	8	40-20	18	cone	horn	comp	rad-ial	super tweet	1.2k 5k	133	$599.50.	
1510 HT	29x25 x14	brown carpet	black metal	8	60-20	15	cone	10	cone		super tweet 5k	450	83	$499.50	
CL-2	21x23 x15	black tolex	grille cloth	8	60-16	6	cone			rad-ial	horn	800	56	$399.50.	
315S	26x22 x16	black tolex	nylon cloth	8	60-14	15	cone	rad-ial	comp	con- sup ical	1.2k twt	88		$399.50.	
1545M	20x20 x23	brown carpet	nylon cloth	8	60-16	15	cone			rad-ial	horn	800	58	$429.50.	
1245M	17x17 x21	brown carpet	nylon cloth	8	60-16	12	cone			rad-ial	horn	1k	52	$399.50.	
112HS BW	17x23 x13	black paint	nylon cloth	8	80-20	12	cone			dual	horn	5k	39	$289.50.	
115HS	19x24 x14	black paint	nylon cloth	8	60-20	15	cone			dual	horn	5k	39	$269.50.	
112HS	17x23 x13	black paint	nylon cloth	8	80-20	12	cone				horn	5k	32	$229.50.	
Flex Monitor	14x20 x6.3	Tolex	grille	8	80-14	8	cone				horn		16	$159.50.	
Mini Monitor	12x20 x16	black	grille cloth	16	100-20	10	cone				horn		14	$169.50.	
STUDER/REVOX															
2706	24 15.1	grey Nextel	black	4	42-20 3	12.5	cone	2	dome	1	dome	720 2.5	48	$670.00.	Hardwood enclosure, tweeter protection, threaded bracket for wall mount or floor stand optional.

Model	Dimensions, H,W,D	Cabinet Finish	Front Screen Finish	Impedance, ohms	Frequency Response +/- dBs	Bass Speaker Size, in.	Bass Speaker Type	Midrange Speaker Size, in.	Midrange Speaker Type	High-Frequency Driver, Size	High-Frequency Driver, Type	Crssover(s)	Weight, lbs.	Price	Features
TANNOY															
SGM 10B	21x15 x10	walnut veneer	brown fabric	8	55-20 3	10	cone			2	comp horn	1.2k	40	$724.00.	Dual concentric drive unit.
SGM 12B	23x16 x11	walnut veneer	brown fabric	8	55-20 3	12	cone			2	comp horn	1.4k	46	$849.00.	Patented sync-source time alignment.
SGM 1000	40x26 x18.5	walnut veneer	brown fabric	8	46-20 3	15	cone			2	comp horn	1k	128	$2,199.	Hard wired crossover with eq.
SGM 3000	40x26 x18.5	walnut veneer	brown fabric	8	38-20 3	15	cone			2	comp horn	1k	128	$2,199.	Phase coherent monitor.
FSM	41x28 x21	walnut veneer	brown fabric	8/4	40-20	15	cone	15	cone	2	comp horn	500 1k	198	$2,499.	
FSM-U	36x31 x23	black textre	black fabric	8/4	40-20	15	cone	15	cone	2	comp horn	500 1k	198	$2,499.	Dual concentric drive units.
Dread-nought	35x53 x23	black textre	black fabric	8	30-20	15 (2)	cone	15	cone	2	comp horn	250 2k	340	$5,499.	
NFM-8	18x12 x8	black textre		8	55-20	8	cone			1	comp horn	1.8k	25	$399.00.	Near-field monitor.
TOA															
SEB	22x28 x24	gray poly	black nylon	8	40-150	18	cone						104	$859.50.	Sub-woofer with internal crossover.
SDB	22x28 x20	gray poly	black nylon	8	40-150	15	cone						86	$679.50.	Similar to above.
SLB	20x26 x16	gray poly	black nylon	8	50-150	15	cone						51	$439.50.	Similar to above.
380-SE	30x20 x16	gray poly	black nylon	8	50-20	15	cone	3	horn	1	m/c	800 8k	80	$799.50.	Designed for electronic music.
480-SE	32x22 x17	gray poly	black nylon	8	45-20	18	cone	3	horn	1	m/c	600 8k	99	$1,259.	Similar to above.
38-SD	27x20 x18	gray poly	black nylon	8	50-20	15	cone	3	horn	3 (2)	m/c	1k 10k	60	$629.50.	
30-SD	23x17 x15	gray poly	black nylon	8	60-20	12	cone	3	horn	3	m/c	1k 10k	46	$579.50.	
SL-12	20x15 x12	gray poly	black nylon	8	70-20	12	cone		pizo horn				9	$279.50.	
SL-15	24x18 x12	gray poly	black nylon	8	70-20	15	cone		pizo horn				5	$329.50.	
312ME	23x13 x12	gray poly	black nylon	8	50-20	11	cone	5	cone	1.5	cone	1.5k 15k	36	$699.50.	Studio monitors.
280ME	16x9 x9	gray poly	black nylon	8	60-20	8	cone	1.5	cone	.75	cone	1.5k 14k	15	$399.50.	Flat response.
265ME	14x8 x10	gray poly	black nylon	8	60-20	6.5	cone			1.5	cone	3k	12	$299.50.	

Model	Dimensions,H,W,D	Cabinet Finish	Front Screen Finish	Impedance,ohms	Frequency Response +/- dBs	Bass Speaker Size,in.	Bass Speaker Type	Midrange Speaker Size,in.	Midrange Speaker Type	High-Frequency Driver,Size	High-Frequency Driver,Type	Crssover(s)	Weight,lbs.	Price	Features
UREI															
809	17x23 x13.5	black		8	50-18	12	cone			1	horn	1.5k		$699.00.	Coaxial monitor with 93 dB SPL.
811C	21x27 x22	black		8	70-18	15	cone			1	horn	1.2k		$1,496.00.	Coaxial monitor with 97 dB SPL.
813C	36x31 x26	black		8	50-18	15 (2)	cone			1	horn	1.2k		$1,996.00.	Coaxial monitor with 101 dB SPL.
815C	32x44 x24	black		8	40-18	15 (3)	cone			1	horn	1.2k		$2,496.	Coaxial monitor with 103 dB SPL.
VISONIK															
David 9000	15x9 X9.7	grey nextel		4	32-25	7	cone	1.5	cone	1				$1,100.00.	
YAMAHA															
SO 110T	29x13 x9.2	black vinyl	black nylon	8	70-19	10	cone				horn	6k	35	$245.00.	Portable system with 97 dB SPL.
S3208 H	22x18 x10.2	black paint	black nylon	8	65-17	8 (2)	cone				horn	2.5k	43	$575.00.	97 dB SPL.
S3115 H	26x19 x14.6	black vinyl	metal	8	65-14	15	cone				horn	2k	65	$565.00.	98 dB SPL.
S4115 HII	33x25 x18.2	black paint	metal	8	50-16	15	cone				horn	1.6k	97	$645.00.	103 dB SPL.
S300	29x22 x15.4	black paint	black nylon	8	50-20	15	cone	8	cone		horn	800 4k	65	$615.00.	3-way system with 97 dB SPL.
S2115 HII	22x27 x26.2	black paint	metal	8	50-16	15	cone				horn	1.6k	77	$615.00.	2-way stage monitor with 100 dB SPL.
S10X	10x6 x6.5	black	metal	6	65-20	4	cone						6.2	$130.00.	Compact monitors.
S20X	12x8 x7.8	black	metal	6	65-20	4 (2)	cone						11	$190.00.	
SW118	30x25 x18.2	black paint	metal	8	30-2	18	cone					120	90	$545.00.	Sub-woofer with 91 dB SPL.

Manufacturer Addresses

Altec Lansing
PO Box 26105
Oklahoma City, OK 73126

AMR
Route 2, Highway 503
Decatur, MS 39327

Apogee Sound Inc.
1150 Industrial Ave, Ste. C
Petaluma, CA 94952

Carvin Mfg. Co.
1155 Industrial Ave
Escondido, CA 92025

Cetec Gauss
9130 Glen Oaks Blvd
Sun Valley, CA 91352

Eastern Acoustic Works
PO Box 437
Jenkintown, PA 19046

Electro-Voice
600 Cecil St.
Buchanan, MI 49107

Fostex
15431 Blackburn Ave
Norwalk, CA 90650

Fane
1316 E. Lancaster
Ft. Worth, TX 76102

JBL Professional
8500 Balboa Blvd.
Northridge, CA 91329

Martin America
PO Box 5139
Chatsworth, CA 91313

Meyer Sound Labs
2832 San Pablo Ave
Berkeley, CA 94702

Peavey
711 "A" St.
Meridian, MS 39301

Studer/Revox
1425 Elm Hill Pike
Nashville, TN 37210

Tannoy North America, Inc
300 Gage Ave, Unit 1
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UREI
8500 Balboa Blvd
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Buena Park, CA 90622

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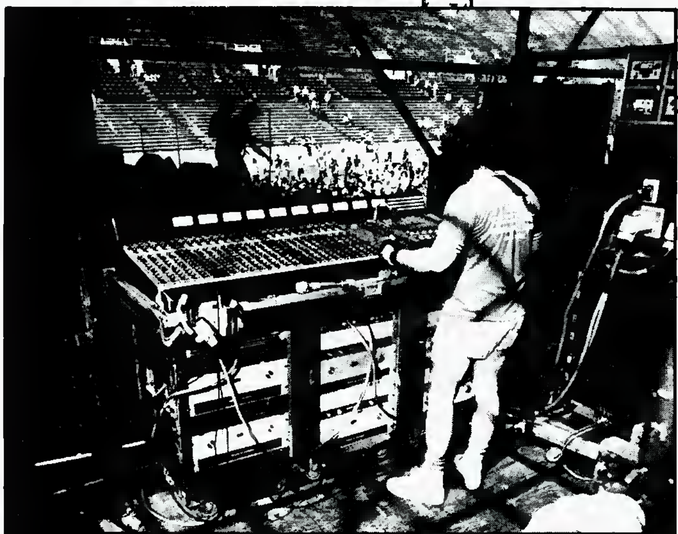

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New Products

VESTA FIRE CASSETTE MULTI

Vesta Fire's MR-30 is a 4-track, 4-channel multitrack cassette recorder (with mixer) that uses standard cassette tape speed to make it compatible with home audio equipment. It incorporates Dolby B noise reduction and 3-band equalizer for easy tone control in mixing down. Overdubbing feature is also included.
Mfr: Vesta Fire USA
Price: $299.00.

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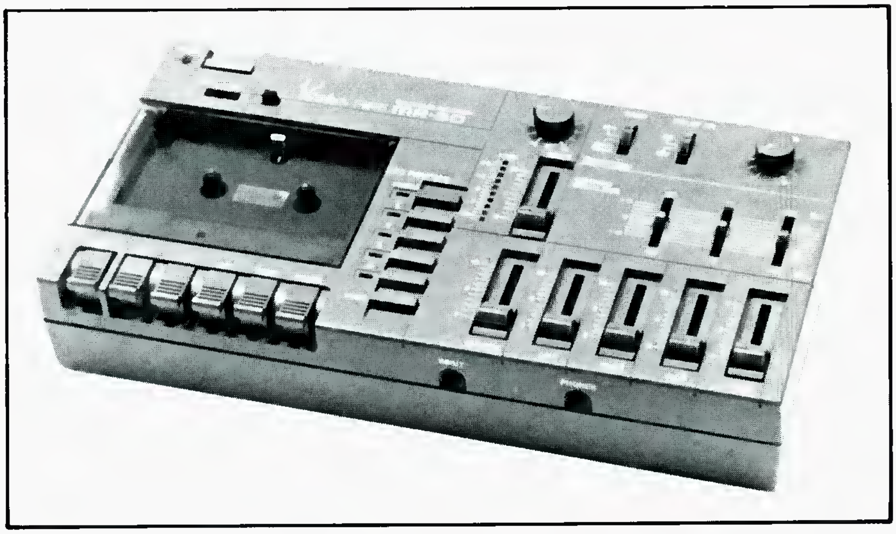
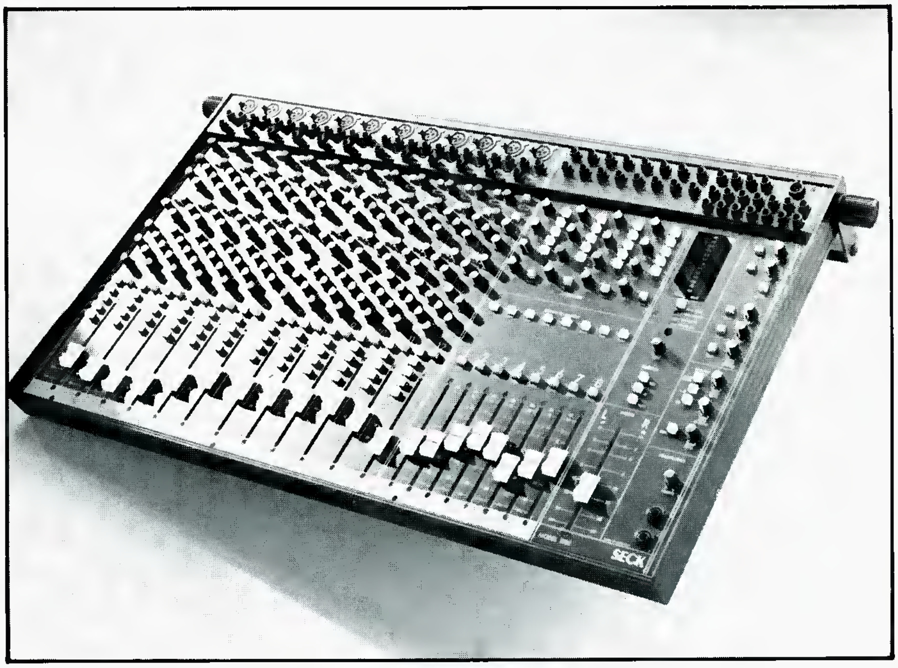


SECK MIXING CONSOLE



The SECK 1282 is a studio quality, portable audio mixing console featuring twelve balanced inputs, eight sub-group buses, four auxiliary returns, each with full equalization and routing and a stereo output. Each balanced mic/line input delivers superb transient response from the extended range gain input section. Instruments are connected directly without the need of a direct box. This optimizes the full benefit of the circuit design. Tip-ring-sleeve insert jacks are provided on each of the 12 inputs and each of the 8 sub-groups. The equalization is in three bands, with a sweepable mid- range. The monitor section is in-line with the inputs allowing for easy and instinctive use of all the facilities. During mixdown the monitor section can be switched to the channel, allowing up to six auxiliary sends to be used. Full-throw 100 mm faders give easy and smooth control over all the inputs, the eight sub-groups and the stereo outputs. A built-in talkback mic, with push-button access to tape and the fold-back buses, allows communication during recording sessions. Metering is via twin bargraph LED meters which can be switched from averaging to peak reading.
Mfr: Connectronics
Price: $3,450.00.

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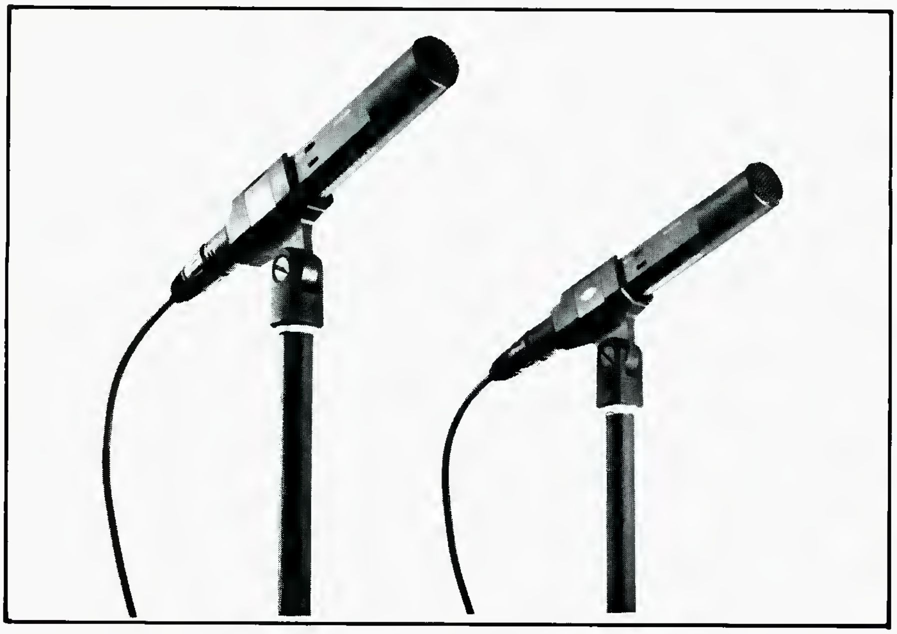
SENNHEISER OMNI MIC



Sennheiser's transformerless MKH 20 P48 omni-directional studio condenser microphone utilizes the same symmetrical capsule as its cardioid sister, the MKH 40. Optimal resistive loading of the microphone diaphragm results in a highly linear frequency response and an inherent noise level that is virtually imperceptible by modern digital recording equipment. The MKH 20 responds to both high and low sound pressure levels with ease and accuracy. It is ideally suited for the reproduction of acoustic string and wind instruments. The MKH 20 has a frequency response of 20 Hz-20 kHz, +/-2 dB; a sensitivity of 25 mV/Pa; nominal source impedance of 160 ohms (balanced); and a minimum load impedance of 1k ohms.
Mfr: Sennheiser Electronic Corp.
Price: $696.00.

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TAPE HOOK

Techworks has developed a new solution to the problem of where to put the reel of 1/4-inch leader tape while editing and assembling audio tape. The Hook is made of nickel-plated solid steel and comes with adhesive-backed fabric fasteners that attach to tape machines, effects racks, audio consoles, desks, or to the edge of any clean, smooth surface. It doesn't deface and devalue any equipment, as the hook-and-loop style fabric allow The Hook to be quickly and easily moved from one piece of equipment to another. It can also be rotated 90 degrees to serve as a spindle for positioning tape reels directly in the tape path, a feature that allows for rapid unreeling of large amounts of leader tape and also facilitates the handling of machine to machine tape loops. The Hook can also be used to support empty reel, flanges, hubs, headphones, patch cords, mic cables, and AC cords.
Mfr: Techworks
Price: $10.00 each.

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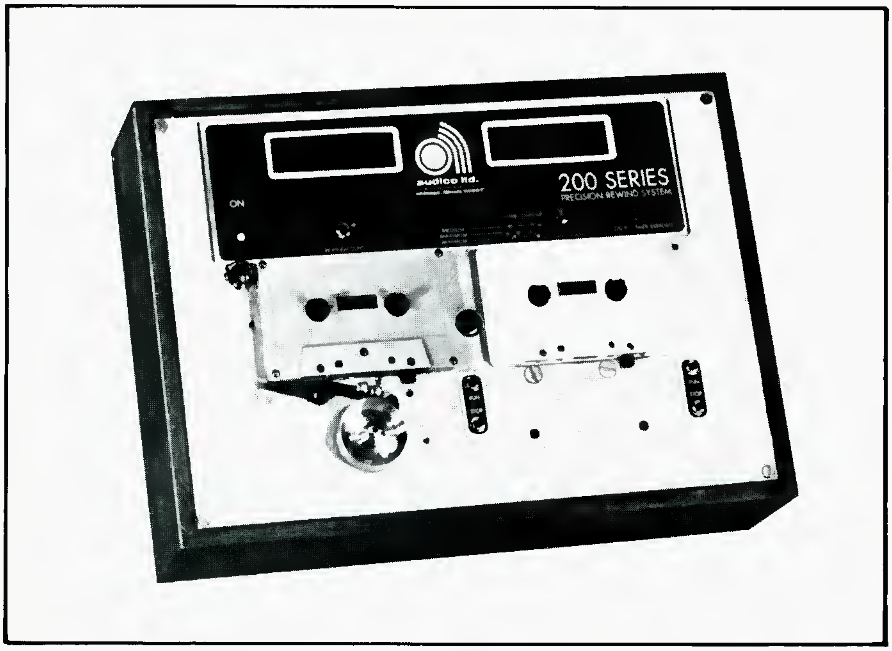
CASSETTE LENGTH VERIFIER



Audio cassette lengths can be easily and rapidly determined using Audico's Model 200-9 Timer/Rewinder/ Exerciser. A sensing device distinguishes between tape and leader, and provides an accurate read-out in minutes and second of playing time. A C-60 is timed in less than 20 seconds. Each of the unit's two independently operated stations can be used as a high speed rewinder. Their combined rewinding capacity is 400 C-60s per hour. One station doubles as the timer and the other as an exerciser. The exerciser life tests cassette shells to determine their quality.
Mfr: Audico, Inc.
Price: $895.00.

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JBL MONITORS

JBL's Control 1 Monitor is a small-sized monitor suitable for musicians working in home studios. Its compact configuration enables console-top mounting. The two-way system incorporates 3/4-inch tweeter, a 5-inch woofer and a high performance dividing network. It provides wide frequency response and has a 4 ohm impedance. It can also be used safely in close proximity to a video monitor because its high and low frequency transducers are magnetically shielded. This feature makes the system ideal for recording studios, mobile audio/video control rooms, and broadcast studios. The molded enclosure will accommodate an optional mounting adapter for permanent attachment to any rigid surface. A clamp system is available for semi-permanent mounting. Mounting on photographic tripods and microphone stands is also possible with the optional adapter.

Mfr: JBL Professional.
Price: $159.00 per pair.
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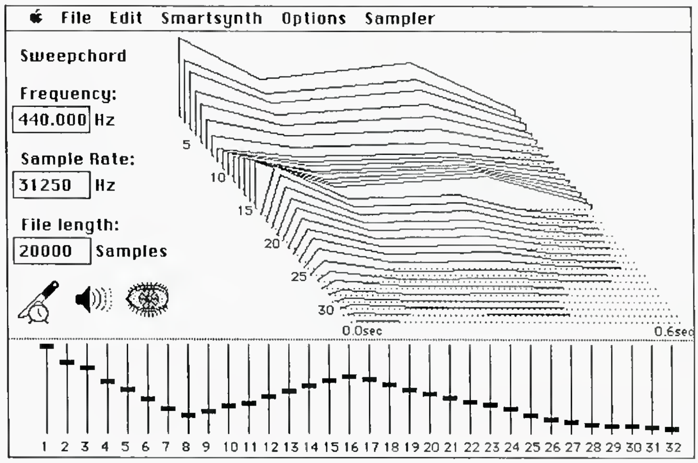

Main Programming Screen

DIGITAL SYNTHESIS SOFTWARE

The Softsynth Digital Synthesis Software for the Apple Macintosh computer and digital sampling instruments is a revolutionary new approach to synthesis: An advanced, 32-oscillator digital synthesizer that creates high quality digital sounds using software-based synthesis. These sounds can be "designed" using Softsynth's graphic programming screens, then synthesized by the Macintosh and transferred to a digital sampling keyboard for playback. It is a digital additive synthesizer capable of creating a wide range of sounds. It provides precise control over all additive synthesis parameters: Each of the program's 32 oscillators/harmonics has a 40-stage amplitude envelope, 15-stage pitch envelope, selectable wave-type and variable frequency. The Main Screen displays the envelopes of all 32 harmonics and provides 32 harmonic faders for adjusting the overall level of each harmonic. The sound length, sample rate, and frequency are all adjustable. In Single Partial mode, separate complex envelopes can be drawn using the Macintosh mouse for the amplitude and pitch of each partial. In Time Slice mode, a single master envelope can be created for the entire sound, and up to 40 different timbre events can be positioned in the sound. The Smartsynth function intelligently generates sounds based on the user's specifications. After the sound has been designed, Softsynth generates a 16-bit digital sample that can be transferred to any Softsynth compatible sampler using a Macintosh MIDI interface. Since they are computer generated, Softsynth's samples have less distortion and noise than acoustically generated samples. The current version of Softsynth works with the Emu-System's Emulator II and Emax, Sequential's Prophet 2000/2002, Ensoniq's Mirage/Multisampler, Akai's S900 and S612, and Korg's DSS-1.

Mfr: Digidesign, Inc.
Price: $295.00.

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ART PRO-VERB

Applied Research & Technology's Pro-Verb has 100 user presets in a 19-inch rack mountable metal package. All relevant program information is situated with controls on the front panel for ease of use in a rack mount. It also has 1/4 inch phone jacks on the rear panel alongside stereo in/outs.

Functions of the Pro-Verb 100 presets are 50 reverb, 10 gated, 10 reverse, 10 chorus, 10 delay and 10 echo effects. It is MIDI-based with built-in AC power supply and offers the latest CISC (Complex Instruction Set Computer) architecture enabling it to perform more powerful, more complex processes per instruction at a rate of 6 million calculations per second.

Mfr: Applied Research & Technology
Price: $399.00.

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CARVIN RECORDING CONSOLE

Carvin's MX-2488 recording console is a full-function recording console with 24 input channels, and 8 outputs. It will serve as a production control center for 8-track studios. It internally routes appropriate signals as required for mixdown and overdubbing, effectively eliminating unnecessary patching. The logical layout and well thought out design of the console will maximize the efficiency of any studio. The MX-2488's independent control room mixer with "bus/tape" source selection enables matrix mixing to the effects and monitor buses for quick overdubs and for live monitor mixing. Each of the 24 input channels feature 3-band, continuously variable parametric equalization with defeat switching, high-quality, low-noise bi-polar differential preamps with mic/line switching, four buses with pre/post switching, assignment switches with pan controls for each of the 8 sub outputs, solo, mute, and 100 mm, long-throw fader. The unit's output channels each feature 2-track interconnects and matrix bus interconnects with pre-posting switching for quick overdub and punch-in capability. It features two effects return "line level" inputs with full function assignments. A separate outboard power supply is rackmountable and easily connects to the MX-2488 via XLR-type connectors for greater flexibility.

Mfr: Carvin Corp.
Price: $3,995.00

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CD SOUND EFFECTS LIBRARY

The Sound Ideas Sound Effects Library is now available on compact disk digital audio. This newest format manufactured by Polygram, West Germany utilizes a specially designed programming system and mastering process to maximize disc capacity and reduce crucial access time. The entire Library of over 3,000 effects, originally offered on 125 reels, is now pressed on just 28 compact discs. The entire Library is fully catalogued using both alphabetical and disc track/index listing and is fully cross referenced. Each CD also has its own numbered jewel holder and every Library comes complete with its own carrying case.

Mfr: Sound Ideas
Price: $1,450.00 (Tape formats also available).

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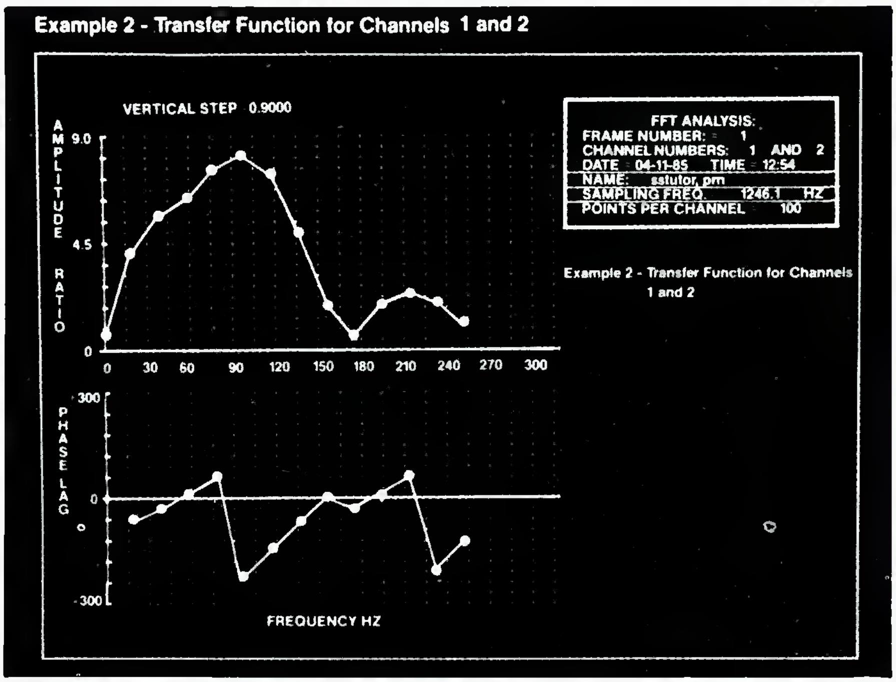
MENU-DRIVEN FFT SOFTWARE



Snap-FFT is a frequency spectrum analysis software package for the IBM PC. It converts time domain data acquired with HEM's Snapshot Storage Scope to the frequency domain using a Fast Fourier Transform (FFT) algorithm. It is menu-driven and requires no programming. It calculates amplitude and phase for four channels, takes the amplitude ratio of any two sets of channels, and takes the difference in phase angle to determine phase lag. The amplitude ratio and phase lag calculations provide the transfer function for control engineers, impedance value for electrical, mechanical and acoustical engineers, and frequency response for instrumentation engineers. The data output can be in either a tabular or graphic format. It can be displayed on the screen, recorded to disk, and/or sent to the printer. Graphs can be plotted on either a linear or log format. Advanced features such as windowing are available to modify data if necessary. The user also has the option of selecting a portion of the time data to be analyzed in the frequency domain. Menu selections are used to change the analysis parameters. The user can save menu settings and then recall them to quickly run various sets of data. Minimum system requirements are: An IBM PC/XT/AT or compatible computer, 256 kB RAM, IBM graphics adapter or compatible, monitor with monochrome graphics capability, and DSDD floppy diskette or hard disk drive.

Price: Snap-FFT $295.00; Snap Shot Storage Scope $495.00.

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1986 INDEX

INDEX BY TITLE

INDEX BY COLUMNIST

INDEX BY AUTHOR

Classified

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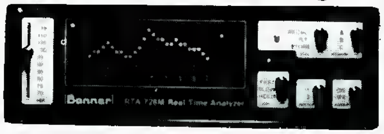

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People, Places...

CareerTrack, Inc. has appointed **Brian J. Battles** as Manager of the new Audio Production Department. Battles will direct and design all audio cassette programs and be responsible for recording, editing, and script supervision. Battles is an authority in the audio production industry with a decade of broadcast experience and is a regular columnist and contributing editor for **db Magazine**.

The Bose Corporation has appointed **David H. Bell** to the position of Manager, Professional Products. Bell has been with The Bose Corporation for more than five years as General Manager of the company's subsidiary in Australia, and previously as National Sales Manager there.

Agfa-Gavaert, Inc. has relocated their Dallas Marketing/Training Center and Regional Distribution Center from their previous location in Las Colinas, Texas. Both centers have been combined in a new, two-story building. They share a total of 41,280 square feet. The new address is 4251 West Highway 114, Irving, Texas, 75063. The phone number is (214) 258-1441.

The **National Academy of Recording Arts and Sciences** has appointed **Joe Smith**, one of the recording industry's most innovative leaders and spokesmen, as its first full-time president. Smith has worked in record promotion, and in the executive ranks as President and General Manager of Warner Bros./Reprise Records and as Chairman of Elektra/Asylum/Nonesuch Records. Smith's office is at the central headquarters of NARAS in Burbank, California.

JBL Professional has named **Hector Martinez** to the new position of market manager. In the position, he will interface with JBL dealers and sales reps and provide their input as new products are developed. Martinez will be focusing on the professional musician, audio, and recording markets. He will work with the JBL dealer network to develop products and programs that emphasize the packaging of JBL loudspeakers, UREI electronic products and Soundcraft mixing consoles and tape machines, as well as moving further toward merging audio with video.

Watch these pages in the next few months for the availability of **db Magazine** columnist and contributing editor **Bruce Bartlett's** new book, which has been based on his columns.

Cerwin-Vega has moved its corporate headquarters and manufacturing facilities to 555 East Easy Street, Simi Valley, CA, 93065. The 110,000-square foot facility will house administrative offices, research and development and manufacturing. Their new phone number is (805) 584-9332.

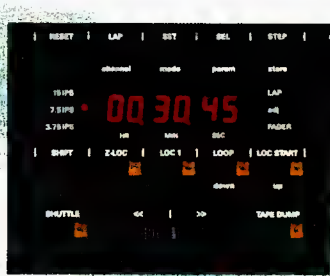
& Happenings

Bhaskar Menon, Chairman and Chief Executive, **EMI Music Worldwide**, announced the commissioning of **Capitol/EMI's** first Compact Disc manufacturing plant in the US located in Jacksonville, Illinois. The plant, a Capitol Records facility since 1965, has 180,000 square feet dedicated to compact disc manufacture and cassette tape replication. The plant has an annual capacity of 7 million compact discs and 25 million cassettes. With the addition of Jacksonville to EMI Music's worldwide manufacturing system, the company now operates three separate compact disc facilities on three continents.

Casio, Inc. announced the formation of a Professional Products Division to be managed by **Jerry Kovarsky** under the direction of **Robert Larsen**, VP of Casio's Electronic Musical Instrument Division.

Top Notch for Your Bottom Line

A top-notch performer, the new Studer A807 has the features and sound quality you need for demanding production applications. And, as the lowest-priced Studer, it also looks good on your bottom line.



The Fruits of "Trickle-Down" Technology. Over the past five years Studer has developed a whole new generation of microprocessor-controlled ATRs. Now, with the A807, budget-minded pros can find this new technology in a compact, ingeniously engineered package.

A Summary of Top-Notch Features:

- Digital setting and storage of audio alignment parameters
- Tape shuttle wheel
- Zero locate, autolocate, start locate, and loop functions
- Multiple timer modes
- Programmable keys for additional locator addresses, fader start ready, or lifter defeat
- Backspace (momentary rewind to play), library wind, varispeed, and reverse play
- Three tape speeds
- Microphone input with phantom powering
- Complete monitoring facilities
- RS 232 port for external computer control
- ...and the list goes on!

Sonically Superior. For superior high frequency dynamics the A807 audio electronics incorporate advanced phase compensation and Dolby HX Pro.™ You won't sacrifice the top end of your sound to enjoy a great bottom-line price.

Tough Stuff. The A807 is 100% Studer, with a die-cast chassis and headblock, rugged AC spooling motors, and a new brushless DC capstan motor. Manufactured with Swiss precision, this ATR is designed to get the job done faithfully—even in adverse outdoor remote assignments.

Suit Yourself. The rack-mountable A807 may be ordered with optional wooden side panels and handles, or in a roll-around console with padded armrest. A wide range of remote controls and options make it suitable for practically any application.

To find out more about this new top-notch performer, call or write for complete information and the name of your nearest Studer Professional Products dealer.

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The Shure system features our exclusive Diversiphase™ dual-antenna system designed to eliminate dropout and provide the strongest signal possible at all times. Unlike other systems, Diversiphase corrects reflected or direct (multipath) signals that are out of phase, so they won't cancel each other...and adds them. Result: more antenna gain.

The new Shure Wireless also prevents interference from TV stations and other radio signals. Each system features a computer-selected frequency best suited to your area or a special frequency for touring needs. Individually tuned linear phase filters also help screen out unwanted signals, without adding distortion.

Fits nearly any application.
Choose from either W25DR Diversiphase or W20R Single-Antenna Receiver with compact W10BT Transmitter. Either Shure system can be used with the specially designed WL83 Electret Condenser Lavalier or a variety of other Shure mics. For information, write or call Shure Brothers Inc., 222 Hartrey Ave., Evanston, IL 60202-3696 (312) 866-2553. G.S.A. approved.

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