including
CONCERT SOUND REINFORCEMENT

RECORDING
ENGINEER/PRODUCER

— *Urgent News* —
NEW TAX THREATENS INDUSTRY
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February 1982
Volume 13 — Number 1

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"The performance and technology sold me. The sound of the '90 sold my clients."

Guy Costa, Vice President and Managing Director, Motown/Hitsville Studios

"When we first got our hands on the Otari MTR-90 we were impressed. The tape handling is superb. The production features are all there. The electronics and logic are to the highest standard we've come across in all the years of Motown's recording history.

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February 1982
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RECORDING ENGINEER/PRODUCER

— the magazine to exclusively serve the **RECORDING STUDIO** and **CONCERT SOUND** industries . . . those whose work involves the **engineering** and **production** of commercially marketable product for:

- **Records and Tape**
- **Film**
- **Live Performance**
- **Video and Broadcast**

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Another original illustration by Trici Venola, whose work illustrating the creative links between music and recording technology has appeared on numerous front covers of *R-e/p.*

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SERIOUS NEW TAX THREAT TO RECORDING INDUSTRY

The California State Board of Equalization has imposed a retroactive sales tax on all independent producers, independent engineers, production companies and recording studios. All monies received prior to 1976, including royalties, will be subjected to a retroactive 6% sales tax, 10% penalty for failure to file, and 1% per month interest. All production expenditures between 1976 and the present, that were in any way connected with the "fabricating" of a tape (i.e., hotel expenses, rental cars, airplane fares, AFTRA scale, lunches and dinners), are subject to the same 6% sales tax, penalty and interest, retroactively. Independent engineers who charge for their services by the hour, are now being told that they should have been charging sales tax on their time since 1976, and are subject to the same sales tax, penalty and interest, retroactively.

. . . continued on next page —

DIGITAL OR ANALOG? IT'S YOUR CHOICE

The First All Digital Audio Console

The Digital Audio Console is here. Only the Neve Group has the financial resources, the advanced technology and the commitment to the industry to be first with digital audio console technology. After more than two years of research and development, Neve has already delivered a complete digital audio console system to a major user for evaluation, with a first production system now being assembled for 1982 delivery.

DSP/CCR - Digital Signal Processing/Complete Control Reset

All program path processes are handled in the digital PCM form, including filtering, equalizing, limiting, compressing, fading, mixing and assignments. Inputs and outputs are in multi-standard 16 bit formats, with greatly enhanced internal dynamic range and processing speed by applying up to 28 bit formats within the system. All program path controls may be memorized, with instantaneous set/reset giving precise repeatability. The time code synchronized computer control provides the ultimate in post production and mix-down facilities. Simply stated: everything is automated.

Superior Analog Consoles with Guaranteed Trade-In

Digital audio is certain to become the primary recording and processing medium. Neve recognizes that only the very top studios can justify the purchase of a Neve DSP/CCR system today. Therefore, for those quality conscious studios wishing to remain analog for the time being, Neve is introducing a Guaranteed Trade-In Plan applicable to 1982 customers of the Neve Advanced Analog Range of Model 8128 Music Recording and Mixing Consoles. This plan allows you to purchase a superior Neve analog console now, and not have to worry about its market value when the time comes to go digital. That's digital insurance!

If you need to buy analog, buy Neve. And get Digital Insurance!

Model 8128 Analog Music Recording Console 32, 48 or 56 Inputs - 24, 32 or 48 Tracks

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views letters news

CALIFORNIA TAX THREAT . . . continued —

Hundreds of industry professionals attended meetings in San Francisco and Los Angeles over the Presidents' Day weekend, to band together in an effort to fight this new taxation. As a result of these meetings, the California Entertainment Organization, a nonprofit corporation, was formed. CEO intends to take action against these tax statute interpretations of the State Board of Equalization, by bringing the enormity of the problem to industry members.

Capitol Records, Warner Brothers, Chrysalis and A&M, as well as individuals and major production companies, have already been served with sales-tax bills. Many have had to pay, and are now in court. By law, the State has the power to impose taxes and collect them by any means it deems suitable. The law also states that in order to fight the tax, you must pay it first. An ever-growing number of industry members are now enduring the threat and/or reality of personal liens against their homes or personal property.

The State accumulates a major portion of its information through audits of major record companies. Every invoice, royalty statement and contract is thoroughly examined, and the originators of these receipts are listed for future audits. "This means," a CEO spokesman explained to *R-e/p*, "that any independent producer, independent engineer, production company or recording studio who has done business with any of these companies during the Seventies, can expect a call from the California State Board of Equalization and an eventual audit.

A NATIONAL ISSUE

Non-residents of California are not exempt from the tax, making this a national issue. Any independent producer or engineer who has done business in California since 1974 is liable. Likewise, if production was conducted in California but sold elsewhere, payment of the tax is enforceable.

"If the smallest recording engineer goes down, the major record companies will go down," the CEO spoksman continued. "We need help — we need membership in the CEO, we need financial support to pay attorneys' fees to sort out this catastrophic mess, and bring the necessary information to our industry and to the public. We have found that ignorance is no defense and failure to file is a punishable of This is no joke — they're going to get you, they're going to get all of us if we don't do something and fast. Otherwise, anyone who is in the record business should go out and get a day job."

For further information on membership and financial support contact: ***CEO, P.O. Box 512, Van Nuys, CA 91408. (213) 906-2080.***

MCI, Inc. ACQUIRED by SONY CORPORATION of AMERICA

As announced by Kenji Tamiya, president of Sony Corporation of America, MCI, Inc., the privately held Florida firm, founded in 1955, will henceforth be an independent division within Sony Corporation of America. Mr. Tamiya stated that the daily operations of MCI will continue unchanged. MCI founder, G.C. (Jeep) Harned will remain as president and chief executive officer of the MCI Division, and Michael Schulhof, a director of Sony Corporation of America, has been appointed chairman. "MCI has a worldwide reputation for quality," said Mr. Tamiya. "Sony will now be able to offer the full range of state-of-the-art recording studio equipment, ranging from microphones to digital audio processors and recorders."

At the signing: Joyce and "Jeep" Harned, and Michael P. Schulhof of Sony Corp of America

VIEW: AUDIO/VIDEO PERSPECTIVES

STEREO TELEVISION SOUND

— The Coming Audio Revolution —
by Martin Polon

The advent of stereo television will provide a much needed growth for all levels of the audio business. At the professional level, the sales of consoles, multitrack recording equipment, and audio processing peripherals will bring television production facilities into the audio Twentieth Century. Skill levels and the sophistication of audio staff will increase, as will the acoustical design and treatment of television production studios. Cable television and local television operations will need to upgrade their audio chain, both for production and distribution. Home consumers will be buying adapters and receivers configured to provide reception of stereo television sound, and probably upgrade or augment the home hi-fi to take full advantage of the improved quality.

Does this sound like another futuristic pipedream? But for the slow grindings of the bureaucratic Federal Communications Commission, the United States could be less than a year away from stereo television. However, the history of stereo television has been marked with many false starts and setbacks.

The quest for stereophonic television entertainment in the living room began before World War II. At that time, the early pioneers of low-band VHF television and FM broadcasting, such as Major Armstrong, envisioned an entertainment service that would

"EDDIE, I LOVE THE SOUND OF THIS BOARD."

Roberta Flack

High praise indeed from a performer of Roberta Flack's stature. But not unexpected by veteran producer and studio owner, Eddie Germano. Ed Germano got to know the JH-600 console series when he bought one for his Hit Factory Studios in New York City. And he was impressed, so impressed that he has already taken delivery of another of MCI's latest, most popular line of consoles.

Professionals like Eddie Germano and Roberta Flack demand the best. That's why they demand MCI.

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TWA6001-0001-82

include multiple channels of audio. The rapid changes in TV broadcasting and radio-frequency spectrum use that the War brought, put an end to serious planning for stereo TV.

The final format used for television transmission throughout the world "acquiesced" to a single channel monaural FM audio capable of 50 Hz to 15 kHz bandwidth. Status quo prevailed through the Fifties. The Sixties became the decade of experimental stereo concerts, using television audio for one channel and monaural FM radio for the other. As stereophonic FM radio broadcasting took hold during the Sixties, the non-availability of suitable long-distance high-fidelity stereo audio transmission channels remained a major stumbling block to television networks.

The relatively poor quality of television sound during the Fifties and Sixties had a "Chicken or Egg — which came first?" quality. American manufacturers of television sets blamed the networks and television station operators . . . the stations and networks blamed the telephone company . . . the telephone company suggested that the burden was on the users of long-distance audio circuits to pay for adequate full-fidelity service. The high cost of transmission relegated most television stations to using 8 kHz bandwidth circuits, or less, with all of the attendant distortions of long-distance telephone transmission. Television receivers of that period had relatively poor audio, with some exceptions in expensive console models. But, nowhere in the sound chain was the full fidelity or channel capacity of TV audio allowed to reach its potential for quality; television was considered to be simply "Radio with Pictures."

In 1958, Motorola offered one of the first systems for stereo TV broadcast, using a narrow-band FM subcarrier of 23.629 kHz above the video-scanning frequency. Traps were used to minimize interference, but frequency deviation was limited to 5 kHz. The system was restricted therefore to a frequency response of 100 Hz to 4 kHz for the second channel, with 25 to 30 dB of stereo separation. The system proved feasible, but little else. Perhaps the major stumbling block to stereophonic audio in the United States still remained the limitations of telephone company audio transmission.

The arrival of the Seventies heralded a change from domestic television set manufacture to Japanese export dominance. The engineering skill and solid-state technology interest brought by the Japanese to TV set manufacture began to affect the limitations on television receiver audio, while other events in Japan made television stereo in the U.S. much more likely to happen.

American television developed sophisticated audio capabilities, with Neve and Ward-Beck boards, for example, replacing workhorse RCA, Gates, and GE consoles at television stations and production centers.

The telephone company problem was finally resolved by combining TV video and audio by means of diplexing. Two full-frequency audio channels are now provided on video circuits: the primary one at 5.8 MHz above the video signal; and a second channel at 6.4 MHz.

Parallel to all of these changes in the U.S., the rest of the world used the late Sixties and Seventies as a stepping-off point for the development of stereophonic and multi-channel/bilingual television services, especially in England, Uganda and, most significantly, Japan and Germany.

The European Experience

The United Kingdom in the Seventies provided a testing ground for the concept of producing high-quality in-studio television audio, and transmitting it around the country for ultimate re-broadcast. The British Broadcasting Corporation has developed several methods for transmitting stereo signals. One system utilizes a nationwide PCM audio feed on coaxial cable for FM radio broadcasting. This multi-channel service has had application during TV/FM simulcasts from the larger BBC regional production centers

. . . continued on page 12 —

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For additional information circle #4

Since its introducti
Synclavier® II has outsold all ot

Synclavier II creates sounds never before possible from any synthesizer.

In April of this year New England Digital Corporation introduced a stereo LP demo record to illustrate some of Synclavier II's incredible sounds. After hearing this record, many people called to say they couldn't believe all the sounds on the demo could possibly have been created by any synthesizer. However, after seeing and hearing Synclavier II for themselves, they were amazed at more than just the absolute realism of its instrumental sounds. They were awed by the infinite variety of tonal colors, unique sounds, and special effects so easily created by this incredible instrument. We might add, many of these people now own a Synclavier II.

Synclavier II not only produces sounds no other synthesizer can produce, it also offers more live performance control than any other synthesizer.

Synclavier II gives you an extraordinary ability to change sounds as you play them. Using Synclavier II's real-time controllers you can accurately recreate many of the subtle changes real instruments make during a live performance.

Here are some of the real-time controls that have made Synclavier II famous: Attacks can be individually altered both in length and brightness for each note. Vibratos can be brought in at different times. Vibrato depths can be changed at will. Individual notes and entire chords can be made to crescendo and decrescendo smoothly and naturally. Final decays of percussive sounds can be made to ring out longer for low notes than for high notes. In strummed chords, some notes can ring out longer than others to compensate for the differences between open strings and stopped strings. Individual notes and entire chords can be pitch bent up or down. The overtone content of any sound can be completely varied from one note to the next. Up to four different rates of portamento can be performed on the keyboard at one time. Some of the harmonics of a sound can remain stationary while other harmonics of the same sound slide against them. And the list goes on.

The possibilities for programming new sounds with Synclavier II are limitless.

Although Synclavier II comes preprogrammed with over 128 preset sounds, it does not lock you into these preset sounds. All of these presets can be modified any way you wish. The possibilities for creating sounds from scratch are limited only by your own skill and imagination.

Synclavier II can store an unlimited number of sounds.

Any sound created on Synclavier II can be permanently stored on a floppy disc with just the touch of a button. From 64 to 256 separate sounds can be stored on a single mini-diskette. The number of mini-diskettes you can use with Synclavier II is unlimited.

All of this is just a glimpse of Synclavier II's enormous potential. The real potential of Synclavier II can be more completely understood by taking a close look at Synclavier II's super advanced hardware and software. The capabilities of Synclavier II's hardware and software extend far beyond any demands currently being made on them.

Synclavier II is controlled by the most powerful computer available in any synthesizer made today.

New England Digital Corporation leads the field in the development and use of hardware applications for music synthesis.

New England Digital uses a powerful 16 bit computer that addresses up to 128k bytes of memory. Other digital manufacturers design their systems around microcomputers. Microcomputers are simply not powerful enough to control large numbers of voices on the keyboard at one time. Most current digital systems are limited to 8 usable voices. When these systems try to control more than 8 voices at once, the speed at which these voices can be played on the keyboard slows down considerably. So, for musical applications, more than 8 voices can not be played on the keyboard at one time.

These microcomputers are also not fast enough to permit extensive real-time control of a sound while it is being played on the keyboard. A few real-time features are available while other important features are deleted because of speed limitations of the microcomputers.

New England Digital Corporation designs and builds its own 16 bit computer, as well as the Synclavier II synthesizer.

New England Digital's 16 bit computer and Synclavier II synthesizer are so unique, New England Digital has been awarded three basic patents on their design, and has several others pending.

The speed of Synclavier II's computer is unmatched by any other digital synthesizer system on the market today. Synclavier II's computer can easily control up to 32 voices on the keyboard at one time without slowing down. No other digital system in the world comes close to this kind of control.

While some synthesizer manufacturers consider a "voice" to be one separately controlled sine wave, one voice of a Synclavier II synthesizer consists of the following: (1) 24 sine waves, (2) a volume envelope generator, (3) a harmonic envelope generator, (4) very sophisticated digital FM controls, (5) an extensive vibrato control, featuring up to 10 different low frequency wave forms, (6) a portamento control that can be either logarithmic or linear, (7) a decay adjust feature, permitting lower notes to have longer decays than higher notes.

on just one year ago, her digital systems combined.

Synclavier II's 16 track digital memory recorder is more sophisticated and has more features than any other synthesizer recorder or sequencer in the world.

Synclavier II's digital memory recorder has enormous capabilities because its computer is fast enough to perform the millions of math computations necessary to make all these features operational at one time.

For example, Synclavier II's digital memory recorder enables you to set independent loop points for each of its 16 tracks. So, you could have 8 notes repeating on track #1, with 64 notes repeating on track #3, and 2 notes repeating on track #7, and so on. All 16 tracks can be looping independently at the same time but still be in perfect sync.

In addition, you can transpose each separate track individually. Track #6 could be transposed up a 4th, while track #8 was transposed down a 5th, and so on.

Other recording features made possible by Synclavier II's ultra fast computer.

Sounds can be bounced from one track to another. You can overdub on just one track, without losing the material already recorded on that track. You can change the volume of individual tracks. You can change the speed of the recorder without changing the pitch. You can punch in and out instantaneously. You can fast forward or rewind just as you would on a 16 track tape machine. You can instantly erase any number of tracks in the recorder.

You can change the scale of a piece of music already recorded in the recorder. For example, if you had a piece recorded in the key of C, you could change it to the key of B flat minor without rerecording a single note in the recorder. Or you could change a piece of music already recorded in the recorder from a tempered scale to a microtonal scale, without recording a single note over again.

You could keep the notes of an instrument that was recorded on one of the tracks in the recorder, and assign a new instrument to play the previous instrument's notes. For example, if a flute were playing on track #5, you could assign a guitar to track #5 and have it play the flute's notes automatically.

Synclavier II's computer is not only the fastest and most powerful computer available on any synthesizer today, it's also enormously expandable, with A to D converters, D to A converters, real time clocks, printers, modems, and alphanumeric and graphic CRT's.

The New England Digital Computer has had 5 years of proven production and successful sales to scientific end users for real-time applications. This history of steadfast reliability has been a major part of Synclavier II's unparalleled success in a market place choked with new products.

Synclavier II has the fastest and most accessible software available in any synthesizer today.

Synclavier II uses an extremely high level structured language called XPL. XPL has proven to be an extremely fast language which has continually provided the means to add new features to Synclavier II on a regular basis.

Other synthesizers are still using languages too limited for our purposes. Assembler is a good example. It is by far a much slower and more difficult programming process to use than XPL. Software improvements made by Assembler language could take months. But with XPL we've been able to add totally new features to Synclavier II in a few days.

New England Digital can add new features to your Synclavier II synthesizer through the mail.

During the 10 months since the introduction of Synclavier II, New England Digital has issued four software updates to the owners of Synclavier II synthesizers. Those updates were mailed out to Synclavier II owners automatically. They included new software that customers had asked for. The updates also included new features and improvements that New England Digital felt were a strong enhancement to the operation of Synclavier II.

Software updates ensure the Synclavier II customer that his system will always be state-of-the-art.

When you buy a Synclavier II, you will automatically be sent new features as they are developed this year, next year, and for years to come.

The Synclavier II synthesizer is not a temporary answer in a technological world moving at warp speed. It is the answer. When you buy a Synclavier II, the instrument improves as fast as our technology improves. Since we're already leading the field of digital synthesis, we feel you are comfortably safe in assuming Synclavier II will be your companion for a long time to come.

When you own Synclavier II, you will never need to sell your "old" system in order to buy a better one. Your Synclavier II system becomes better automatically.

For further information and a copy of Synclavier II's stereo LP demo record send your address plus $2.50 (outside USA $6.00) to either of the following:

Dept. 10, New England Digital Corp.
P.O. Box 546, White River Junction, VT 05001
(802) 295-5800

Denny Jaeger, Western U.S.
N.E.D. Rep.
6120 Valley View Rd.,
Oakland, CA 95611
(415) 339-2111

new england digital

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—AUDIO/VIDEO—
RECORDING

STEREO TELEVISION SOUND

— The Coming Audio Revolution —

to the Corporation's nationwide matrix of television transmitters. A diplex format, with the audio riding on the video signal, has also been utilized by the BBC, and is known as "Sound-in-Sync."

The production in BBC studios of stereophonic television operas such as *The Love Of Three Oranges*, and *MacBeth*, has provided impetus to the search for a suitable system for in-home stereo television broadcasting. There are plans to field test a DPSK digital system for TV broadcast use. Satellite direct broadcast tests have also taken place, and Britain is observing the results of the Japanese and German stereo TV experience.

Britain's Independent Broadcasting Authority (IBA) is studying compatible 2½- or 3-channel Ambisonics/UHJ "surround-sound" broadcast techniques, to provide for the home.

It seems that the potentials for UK or EBU (European Broadcasting Union) direct-broadcast satellite may prove the most feasible way of providing stereo TV sound, since a satellite down-converter in the home could easily take care of the stereo decoding. The existence of hundreds of millions of non-stereo TV sets around the world, means that acceptance of stereo TV will be dependent upon *compatible* technology that allows for inexpensive out-of-set decoding.

Surprisingly, Uganda provided a unique testing ground for multi-channel television bilingual service during the mid-Sixties. The use of four FM subcarriers to carry the four major tribal languages proved that it was feasible to reduce the main carrier modulation, and thereby allow the use of subcarriers. The bilingual service was utilized on inter-region hops, with all station-to-station linkage being provided by re-broadcast. On-air signals received within the country were monophonic, the correct language for a region being selected from the main transmission. The validity of this service was enhanced by the fact that all of the equipment used was of off-the-shelf American or English manufacture.

Germany has conducted extensive laboratory tests and propagation studies on a compatible double carrier system, located above the first 5.5 MHz sound carrier, and below the upper adjacent channel. The two-carrier system has also been tested in East Germany, Finland, Italy, Switzerland and Yugoslavia, as part of on-going EBU research projects in stereo TV which have been underway since 1966. The West German studies, operated by the Institute fur Rundfunktechnik (IRT) in Munich, have culminated with official adoption and on-air tests that began in September 1981.

The German IRT study for system selection established three parameters for the stereo dual-carrier system: the resultant stereo TV service is compatible with all existing receivers; it provides two isolated channels for bilingual service, as well as stereo; and does not degrade the transmitted television signal throughout the service area. Audio provided by the second channel has 40 Hz to 15 kHz frequency response, with 50 to 60 dB signal-to-noise ratio, and crosstalk rejection approximately 60 dB with independent channels.

The German system, keyed to 625-line PAL television broadcasts, has no current application for stereo television in countries where 525-line NTSC scanning is used, such as Japan or North America. But, within the short space of six months, 65% of all West German viewers (plus a small group of dedicated East German audio-video enthusiasts) are within the reach of the new West German stereo television transmissions. Currently, the state-run ZDF Network is offering stereophonic and bilingual programs through 29 stations in the Federal Republic of Germany. It is felt that neighboring East Germany, having little desire for the expense of such Western "decadence," will go along in a de facto way, as will bordering Switzerland and Belgium.

Stereo television receivers and adapters are currently available from such European manufacturers as AKG, Blaupunkt, Grundig, ITT Lorenz, Metz, Philips, Siemens, Sony Deutschland, and Telefunken, among others. Fifteen manufacturers are making available a range of over 50 different television sets suitable for the German stereo TV service. A whole range of accessory decoder/converters for existing television sets, outboard speaker and amplifier units, remote infrared headsets and other devices are also currently being manufactured. One of the big selling points expected to impact the affluent German consumer is the stereo TV systems bilingual characteristics. This provides ease of access to popular foreign (read: Hollywood) feature films without the need for subtitles.

STEREO TV IN JAPAN

The Japanese experience represents the most detailed examination of the development and true potentials of a stereophonic television audio service. As early as 1964, utilizing the Motorola system under license, Japanese broadcasters began experimenting with bilingual, multi-channel TV multiplexing. Experiments were also conducted with FM subcarriers, AM subcarriers with suppressed main carrier, and single sideband subcarrier (similar in concept to the proposed AM-stereo systems for radio broadcast).

Beginning in 1969 and running until 1973, the Technical Research laboratories of the Japan Broadcasting Corporation (NHK) finalized and tested the FM/FM subcarrier system, which was adopted as a national standard in Japan for stereophonic and bilingual (simultaneous Japanese and English) broadcasting. Beginning in 1978, a nationwide stereo television service has been available on-the-air in Japan.

The Japanese FM subcarrier system operates at a center carrier frequency of 31.47 kHz (Figure 1). This is twice the color-line frequency for horizontal television signals. This method minimizes audio interference from video signals (Figure 1). The L+R,L-R system provides 50 Hz to 15 kHz frequency response on the main channel, and 50 Hz to 12 kHz on the secondary channel. Compatible stereo is provided with existing TV sets receiving a monaural audio signal. Signal-to-noise ratio is in the range of 60 dB, with stereo reception even in degraded areas offering 55 to 58 dB.

Since the system first went on the air, NHK engineers have found ways to improve the secondary channel response to a full 15 kHz bandwidth, and to add a third audio channel of limited response without degrading the overall system performance.

During over three years of operational usage, the Japanese TV stereo system has reached more than 70% of all homes in Japan. Since the beginning of a regular service, over two million stereo/bilingual TV receivers, adapters for existing TV sets, and adapter/amplifier/speaker systems have been sold in Japan. Stereo television now accounts for more than 20% of all receiver sales in Japan, with over 100 models currently on the market. The service is offered by more than 30

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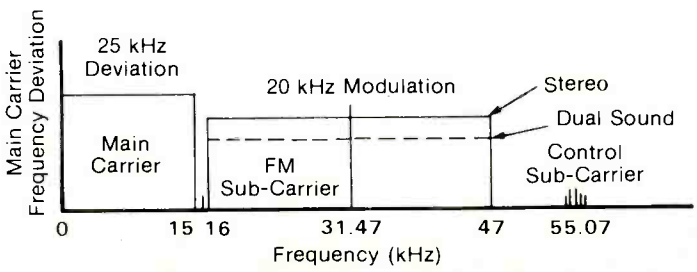
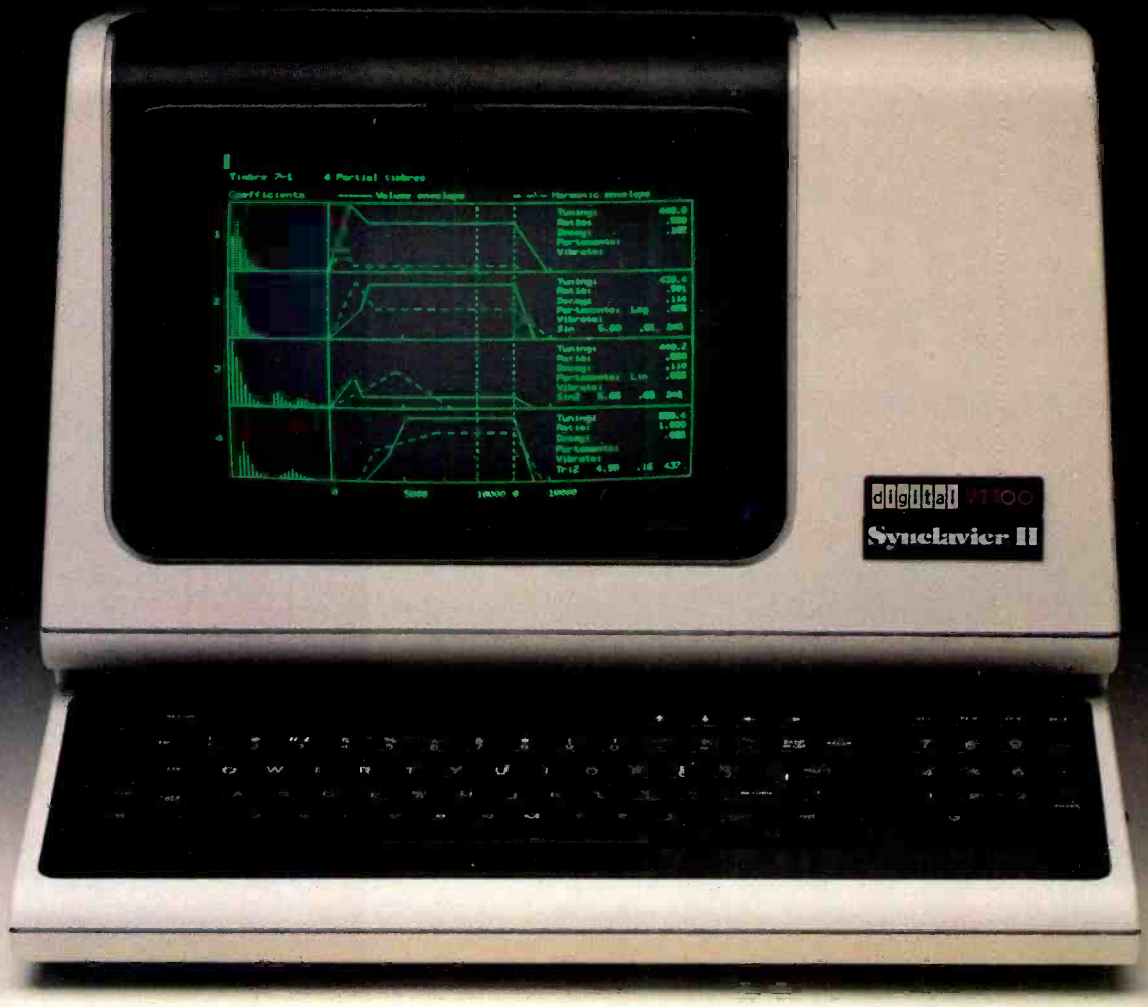


Figure 1

Introducing Synclavier II's Terminal Support Package.



The Terminal Support Package provides a completely new method to access Synclavier II's computer. The Terminal Support Package consists of three items: (1) Graphics, (2) Script, a music language, (3) Max, a programming language.

GRAPHICS

The Graphics Package allows the user of Synclavier II to have a readout of numerical data printed out on a computer terminal screen. With the depression of the return character on the terminal, the numerical data is changed into a graphic display. A clear depiction of the volume and harmonic envelopes are drawn out on the screen. The relative volumes of each sine wave, comprising the sound whose envelopes are currently on the screen, is also displayed.

The graphics display provides an extremely valuable visual tool for programming new sounds and for thoroughly analyzing sounds which have already been programmed for Synclavier II.

SCRIPT

Script is a music language. It can be used as a composing tool to write musical performances into Synclavier II's computer without playing anything on the keyboard. Precise polyrhythmic melodies can be developed which would be difficult or even impossible to play on a keyboard. Composing with Script gives you up to 16 tracks to record on.

All the real-time changes available with Synclavier II's digital memory recorder can also be programmed through a terminal with Script. This includes dynamics and other musical accents.

Any composition created with Script can be stored on a disk, and then loaded into Synclavier II's digital memory recorder. All compositions created with Script can be made to play back in perfect sync with a multi-track recorder.

Another feature which is extremely helpful for musicians is the editing feature of Script. This allows you to edit existing compositions through the terminal. You can cut apart, reassemble, or tailor in any manner a composition without ever risking a loss of any of the original elements.

MAX

Max is a complete music applications development system. It allows you to control all of New England Digital's special purpose hardware, i.e., the computer, analog-to-digital converters, digital-to-analog converters, and other devices like a scientific timer which can be programmed to be SMPTE compatible.

Max comes complete with documentation for the Synclavier II hardware interfaces to enable a programmer to design his own software program. This language is for people who possess a much more sophisticated knowledge of programming computers. Basically it is a superset of XPL, the software language New England Digital uses to program Synclavier II's computer.

Max is designed to permit the owner of Synclavier II to take greater advantage of New England Digital's powerful 16 bit computer. Up to now, all software had to be written by New England Digital. The Terminal Support Package with the Max language gives you the opportunity to explore new ground on your own. The ways in which Synclavier II's hardware can be used by Max is virtually limitless.

All of us at New England Digital feel we've only begun to explore and tap the awesome potential of the Synclavier II digital synthesizer. The Terminal Support Package is just one more step in an exciting journey toward this realization.

new england digital

For additional information circle #6

The world's most advanced digital music system has just become the world's most advanced sampling and printing system.

Introducing the Synclavier II music sampling and printing options.

Synclavier II's Digital Analysis/Synthesis Option (Sample-to-Disc)

Synclavier II's sample-to-disc option is the superior approach to music sampling. Its audio fidelity and its length of sample times far surpass anything on the market today.

Synclavier II's sample-to-disc option can be added to any size Synclavier II music system. It offers 14 or 16 bit resolution and a sample rate of 50K. You can select a range of sampling times from a minimum of up to 100 seconds to a maximum of 24 minutes. (The longest you can currently sample a sound with existing sampling synthesizers is under 10 seconds.)

Synclavier II's sample-to-disc option lets you have either one long continuous sample or any number of shorter samples you wish. This provides maximum flexibility for sampling sounds.

Here are just a few of the things you can do with Synclavier II's sample-to-disc option. You can record real instrumental sounds and sound effects into Synclavier II and then play them on the keyboard. After you've sampled a sound, you can use Synclavier II's terminal to do digital filtering and exhaustive analysis and modification of sampled sounds. In addition, you can mix the attack of a real instrument with a synthesized sustain to create one composite sound. For example, you could take just the attack of a real trumpet and use it to replace the attack of a synthesized trumpet.

Synclavier II's sample-to-disc option is so advanced, all of its capabilities are not yet fully understood. But one thing is sure, it's destined to permanently change the world of music synthesis.

Synclavier II's Music Printing Option

Synclavier II's music printing option is an extremely sophisticated computerized printing system, capable of printing entire musical compositions with tremendous accuracy. The rhythmic resolution of the printer is adjustable from 64th notes to any greater rhythmic value.

With this printer option, you can play any combination of notes on Synclavier II's keyboard, then automatically transfer those notes to Synclavier II's terminal. You can then edit the notes, modify the notes, transpose the notes, correct the meter, and even add crescendos and decrescendos. When you type "print," everything prints out in traditional music notation.

Synclavier II's music printer option can be added to any existing size Synclavier II. It can be used to print out lead sheets (complete with lyrics), music scores, and individual instrumental parts.

This amazing new development frees you from the drudgery of translating your musical ideas to paper. Now you can concentrate on your creativity and let Synclavier II take care of the paper work.

It's not by chance that Synclavier II is the number one selling digital system by a wide margin. New England Digital's technology is simply the most advanced in the world.

Synclavier II's 16 bit mini-computer is 10 times faster and more powerful than any microprocessor currently being used in other digital systems. Synclavier II's high level real-time language, XPL, gives Synclavier II the fastest and most flexible software available today. And here's what's going to keep Synclavier II number one. Synclavier II is the only system that has completely modular hardware and software. This unique combination gives New England Digital an extraordinary capability to quickly adapt new advancements and options to Synclavier II's existing programs.

The sample-to-disc option and the music printing option are an example of this, yet they are only a next step in the continuing evolution of the world's most advanced digital instrument.

When you buy a Synclavier II today, you won't be stuck with today's technology tomorrow. Synclavier II has the ability to be upgraded and enhanced year after year to remain state-of-the-art indefinitely.

Synclavier II digital music systems start at $13,750.00.

For further information and for a personal appointment to see Synclavier II, call or write:

Dept. 10, New England Digital Corp.
Main Street, Norwich, Vermont 05055
Attention: Brad Naples
(802) 649-5183

Denny Jaeger Creative Services, Inc.
6120 Valley View Road
Oakland, California 94611
(415) 339-2111

new england digital

For additional information circle #7

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AUDIO/VIDEO RECORDING

STEREO TELEVISION SOUND

— The Coming Audio Revolution —

broadcast stations in such cities as Tokyo, Osaka, Nagoya, and other large metropolitan areas.

The Japanese multi-channel service is being used to provide stereo sound for musical programs, feature films, sports such as baseball, and other live events where the stereo is considered to enhance the experience of "being there." News, American and other foreign television programs, feature films from abroad, etc., are presented in multi-channel format, with English or Japanese translations supplied as the second channel information. The presence of stereo sound has increased sales of large screen television systems in Japan, apparently to allow increased enjoyment of the stereo programming, for the stereo TV service has become very elaborate. For a 30-minute Sunday night program called *Sound Inn 'S,'* the Tokyo Broadcasting System utilizes 60 microphones for over 50 performers, mixed down to 15 tracks on a 16-track audio recorder; the last track is reserved for SMPTE time code. The 15-track output is also further reduced to two-channel stereo mix for inclusion on the one-inch C-Format VTR for monitoring. Post-production is used to conform a final stereo mix from the 15 tracks on audio tape, and dub it on to the videotape for eventual broadcast. The degree of acceptance of stereo TV services in Japan can be measured by the complexity of software production for the stereo audience.

Currently, the public and private networks in Japan (NHK and NTV) offer nearly 40 hours of stereo TV programming per week.

Meanwhile . . . in the U.S.

At the beginning of 1980, the U.S. broadcast television marketplace has seen the re-equipping of many television studios for stereo audio, the widespread acceptance and use of diplexers for telephone company and in-house microwave transmission of audio with video, and a real improvement in the audio quality of television sets from the same manufacturers building stereo televison receivers and adapters in Japan. But, as yet, no standard exists for stereo television in the United States. There is currently little in the way of detailed laboratory investigation of stereo TV systems, such as took place in Germany, Great Britain and Japan.

In the United States, the investigation of compatibility with existing NTSC monaural television transmissions is entrusted to a subcommittee of the Electronic Industries Association (EIA). The Subcommittee on Multi-channel Sound is responsible for recommending a compatible system; the need for and type of companding of the stereo audio signal; and the method for matrixing the stereo to create compatible monaural signals. Committee recommendations will travel through the EIA to the FCC — the whole process could take several years, especially if any protests to the process occur.

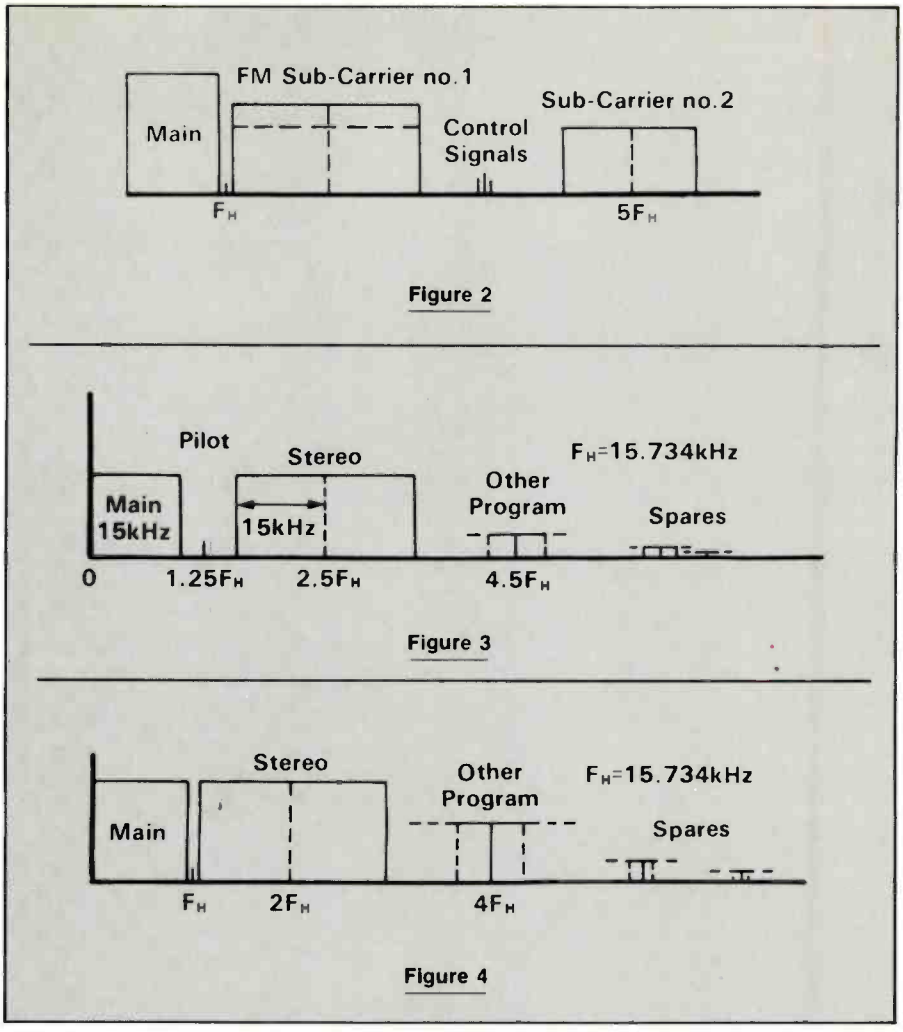

Figure 2

Figure 3

Figure 4

Currently, the EIA committee is considering three submitted systems for stereo transmission in the U.S. The first is the system from Japan, submitted by the Electronic Industries Association of Japan. The EIAJ system offers two stereo channels with 50 Hz to 15 kHz response, and a limited response channel of 100 Hz to 8 kHz (Figure 2). The second system to be offered is from a company called Telesonics, and utilizes a double sideband suppressed AM subcarrier. The Telesonics system provides for two stereo channels to 15 kHz, with an isolated channel to 5 Hz and provisions for carrying telemetry and remote feeds (Figure 3). The last system is from Zenith, and also uses an AM double sideband suppressed subcarrier (Figure 4). The Zenith system also provides for an isolated channel, remotes and telemetry.

Three noise-reduction systems will be tested to improve the signal-to-noise ratio for each transmission system: Dolby, dbx, and CBS' CX. Each noise-reduction system will be evaluated using average, TV-quality speakers, and with true high-fidelity monitoring systems. Also to be evaluated by the committee will be the matrix options of L+R/L-R, versus L+R monaural encoding.

The Japanese, in their extensive tests, rejected companding or noise reduction, since the FM/FM system provided the best signal-to-noise ratios as is. The Japanese experience also validated the full matrix of using L+R and L-R. Combining left and right alone produced an incorrect monaural signal, lacking some psycho-acoustic signal components.

On-the-air tests for the EIA stereo sub-committee have been conducted at several stations, including WTTW in Chicago during the latter part of 1981. Corollary cable system stereo tests utilized Continental Cablevision's system in Franklin Park, Illinois.

Will Cable be the Answer?

The U.S. cable industry is not waiting for either the EIA sub-committee, or the FCC. Many cable operators have begun to offer stereo service that is connected via an FM tuner or receiver to provide a stereo audio companion for cable video. Warner-Amex's MTV Music Channel is

. . . continued from page 17 —

—AUDIO/VIDEO—
RECORDING

STEREO TELEVISION SOUND

— The Coming Audio Revolution —

providing stereo sound to accompany the video-music visuals.

The Movie Channel from Warner-Amex is also in stereo, and uses stereo source material if available, such as Dolby Stereo soundtracks of the feature films that have them. If the film soundtrack is monaural, a stereo synthesizer is used (Orban models, in particular, are said to be popular with some cable programming suppliers) to create the desired dual-channel effect. The audio programming is inserted into the video signal using multiplexing equipment manufactured by Wegener Communications, and transmitted uplink to satellite. At the hundreds of earth stations used by the various cable television operating companies, downlink signals are separated again with Wegener equipment, and placed on the broadcast FM spectrum. The satellite system can handle up to four discrete left and right audio channels. The home user draws stereophonic audio from an FM tap on the cable system, enabling a hi-fi system to reproduce the TV audio.

CBS Cable is using a Dolby-encoded stereo signal as part of its new cultural service.

Cable is capable of providing stereo now to the home hi-fi. The home TV set on cable will receive a matrix mono signal.

On another front, CBS Television, in conjunction with Sony, Ikagami, Matsushita, and the National Broadcasting Corporation of Japan, has completed a production evaluation of high-band, high-definition television. Operating with a video picture of at least twice the current 525-line scanning rate, the CBS/NHK 1,125-line system is accompanied by full multitrack stereophonic sound. During production of test segments in Hollywood and at the Tournament of Roses in Pasadena, California, appropriate stereo sound was mixed to accompany the enhanced visuals. In Japanese tests as well, the use of stereophonic audio to accompany high-definition television is a foregone conclusion. The presence of another audio channel is of little consequence to the extended 25 MHz bandwidth of high-definition television.

The question remains, however, who is waiting for the ponderous EIA/FCC decision process. The Japanese and the U.S. cable industry have already set direction, but broadcast stereo is still in limbo.

In summary, the marketplace will follow the strongest movement; the rapid growth of cable television, stereo videodisk and VCR's will attract the public's attention and needs perhaps years before the cumbersome decision process finishes its way through the EIA and the FCC. Even worse, the entire process in America, and indeed Japan and Germany, depends upon the supply of suitable stereo visual software. *Now* is the time to ensure that all production currently in progress is mixed to stereo, even if the initial release is not in that format. The demands of a single cable television channel operating just eight hours per day (4 p.m. to midnight) requires nearly 3,000 hours of programming per year. This translates to about 1,800 productions of more than 1½ hours in length. Hollywood produces only about 100 features a year, and co-produces another 100, plus whatever output is made as television movies; the net is in the range of 400 new features per year.

Since we are on the verge of having HBO, Showtime, Warner-Amex and a number of other programmers offering dedicated movie channels with stereo capability, the need for software cannot be overstated. This software need translates into the potential for new equipment and more jobs involved in all levels of audio and video production for stereo television. □□□

Photo by Richard Aaron

— an unusual view of doing a remote in Africa . . .

Mick Fleetwood Recording *"THE VISITOR"* in Ghana

by Marc Silag

The release last spring by Fleetwood Mac's Mick Fleetwood of *The Visitor* album was notable for a number of reasons. Recorded in Ghana during the early part of 1981 the album was, according to Fleetwood, " . . . the culmination of a life-long ambition." Artistically, the project represented an opportunity to engage western ideas of rhythm and harmonic structure with African dispensations of native rhythms, instrumentations and vocals. Culturally, the sessions provided western musicians and nearly 200

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For additional information circle #10

views

Mick Fleewood in Ghana

Ghanaian players with the opportunity to experience a meeting of dissimilar rhythmic minds and musical roots. For years western musicians have been influenced by the residual effect of African rhythms on the basic structures of blues and rock and roll. To a degree the tables were turned in Africa, but as listeners to *The Visitor* can attest, the jungle beat we've all come to know as "Not Fade Away" is particularly effective in the hands of the Ghanaian players recorded by the specially assembled recording crew that accompanied Fleetwood to Africa last year. Technically speaking, pulling off a project like this, 10,000 miles away from the safety laden luxury of remote recording in Los Angeles, was no piece of cake.

Although Fleetwood himself advanced the Ghanaian locale before loading band, crew and equipment aboard African bound jets, the recording crew was faced with the formidable task of assembling a clean, portable recording set-up, adaptable to whatever situation awaited them in Ghana, using equipment that could be maintained and serviced in the field without reliance of back-up equipment. Originally, Fleetwood had hoped to undertake what would have amounted to a recording safari, carrying multitrack equipment with him through the bush of Ghana, setting up and recording with indigenous musicians along the way, and integrating this playing with his material for the album. On his preliminary survey of Ghana, however, Fleetwood discovered that such a plan would have to be scrapped; the roads of the Gold Coast nation are in terrible condition (where they exist at all), and gas is not a readily available commodity. As recent events in the news point out, Ghana's stability has been a constant cause for concern since the departure of British rule nearly 40 years ago.

Committed as he was to the project, Fleetwood made arrangements with the Ghanaian Musicians Union to utilize the services of African players. All of the negotiations for the project required an inordinate amount of government assistance (hence the diplomatic "Who's Who" worth of credits on the album sleeve). Through these contacts Fleetwood was able to secure an ideal space to set up his recording system: a cavernous film soundstage the British had built, but which had stood dormant for several years.

Back in Los Angeles, Fleetwood's co-producer Richard Dashut was working closely with Randy Ezratty, founder and chief engineer of Effanel Music, which has gained an excellent reputation by offering a *proven* portable (as opposed to simply mobile) recording system. Effanel Music deserves a great deal of credit for pulling off this little venture. Having spent years on the road with a variety of performing artists, Ezratty began to pick up dates engineering recording projects on the West Coast. It was in this manner that he first met Dashut and Fleetwood, while working on a Bob Welch session. During this time Ezratty was also beginning to assemble equipment for use as a portable recording package, intending to go into business for himself. When the opportunity arose to equip the Fleetwood excursion to Africa, Randy quickly secured the balance of equipment necessary to do the job.

Portable Recording System

Fortunately for Ezratty, the forethought and planning he'd already expended on Effanel proved to be an ace in the hole when the Fleetwood project came up. It was coincidental that the African trip demanded many of the specifications Ezratty had used as prerequisites for his portable studio concept. Working with Dashut and engineer Billy Youdelman of the Wally Heider remote recording squad, Ezratty found that he shared a mutual method of approach with his two associates: the

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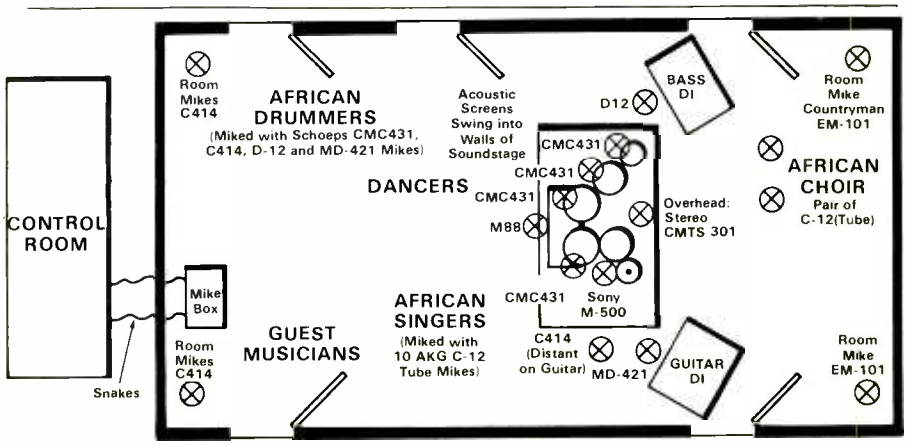

STUDIO A

- NEVE 8088 CONSOLE (52 IN 48 OUT) WITH NECAM
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- CUSTOM JBL/TAD MONITOR SYSTEM
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Once you go through a recording session with the new ATR-124 24-channel recorder by Ampex, you'll want to go through another. Because with each new session you'll discover something new you can do. Things that you can only do with a recorder that's full of features of the future.

ATR-124 gives you the unheard of: Time on your hands.
Which means you can use that time to give clients more of what they're paying for—your creative skills. With the ATR-124 microprocessor-based control system, you can pre-program what you want to do ahead of time so you won't waste studio time setting things up. When their time starts, you're ready to record by touching a single recall button.

ATR-124 also lets you duplicate a technique you may have used earlier in the session without having to rethink what you did. Just touch the memory button and it'll all come back to you. ATR-124 lets you rehearse what you've got in mind, without recording it, to make sure what you've got in mind is right. Tape can be manipulated faster which means you'll get the sound you want sooner. And the chance to try something "a little different." All because of the speed and accuracy that ATR-124 puts at your fingertips.

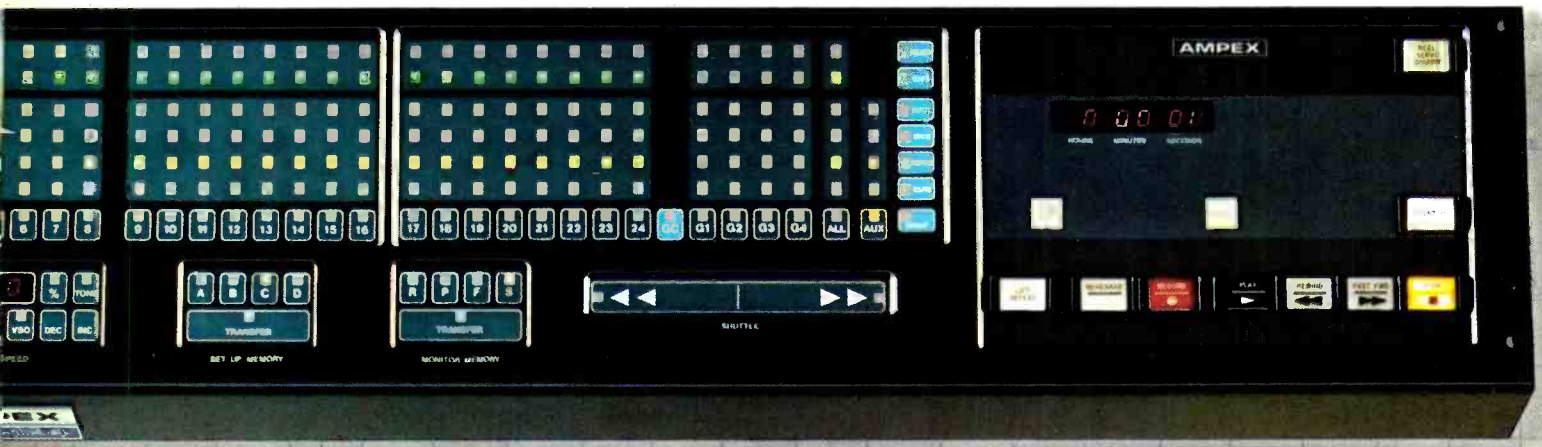
ATR-124 doesn't take away your creativity, it adds to it.
The less time spent setting up, correcting, and redoing, the more time spent creating. And when you add features that help you create to the ones that help you save time, you've got one very potent piece of audio machinery. Take the control panel for instance. It's like nothing you've ever seen. Pushpads linked to a microprocessor give you a new level of creative flexibility. Program a setup, then change it. Then change it back, all with a single fingertip.

A repeatable, variable speed oscillator for pitch correction and special effects is built in. In addition

from the future: ATR-124.

to the standard output, there is an optional auxiliary output with each channel that enhances flexibility. So don't think that ATR-124 is going to replace anything that you do. On the contrary, it's going to improve the skills you have, if not help you develop some new ones.

Memory, and Record Mode diagnostics. The point is this: If you like the ATR-100, you're going to love working with the ATR-124.



ATR-124's Control Panel. Speed and accuracy at your fingertips.

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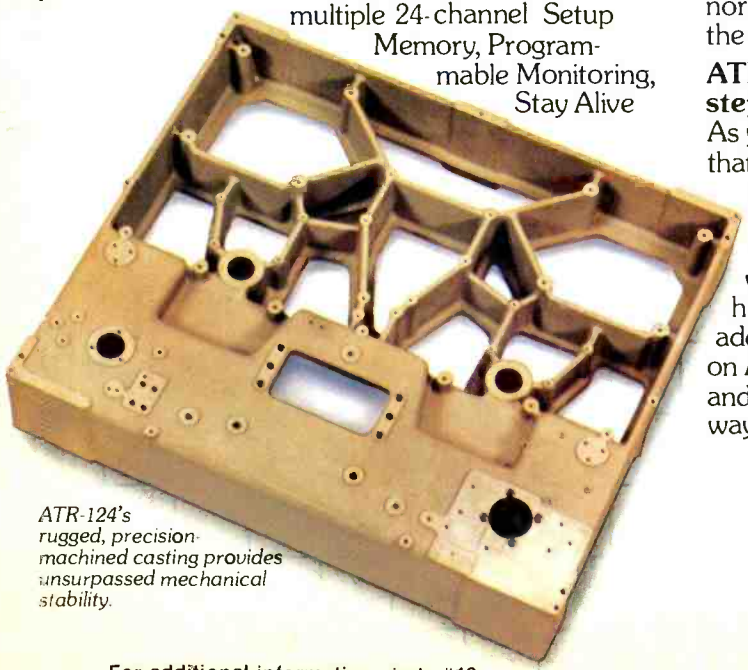
It's only natural that the people who brought you the ATR-100 should be the ones to bring you something better. ATR-124 offers you 24 channels instead of 4. You also get many new and exclusive features. The kind that have set Ampex apart from the crowd for the last 30 years. Features like balanced, transformerless inputs and outputs; a patented flux gate record head; 16" reel capability; input and output signal bus for setup alignment; membrane switch setup panel; fingertip-operated shuttle speed control; and microprocessor-based synthesized Varispeed -50% to +200% in .1% steps or in ¼ tone steps. ATR-124 also features microprocessor-based control of Channel Grouping, multiple 24-channel Setup Memory, Programmable Monitoring, Stay Alive



ATR-124's rugged, precision-machined casting provides unsurpassed mechanical stability.

ATR-124 options.
As impressive as the ATR-124 itself.
With the addition of a built-in Multi-Point Search-To-Cue (MPSTC), you can rehearse edits and control five tape-time actuated events and be compatible with SMPTE time code. Separately controlled auxiliary output amplifiers with each channel provide simultaneous monitoring of normal and sync playback as well as all other monitoring modes. A roll-around remote control unit can also be added to the ATR-124 which contains all control features normally found on the main unit.

ATR-124's Multi-Point Search-To-Cue (MPSTC). Provides 100 cue locations.

ATR-124. Your next step is to experience it firsthand.
As you scan the points we've covered, remember that you're scanning just a small portion of ATR-124's story. We haven't even begun to discuss the accessibility of key components for easy servicing and minimal downtime, or the features we've built in to give you greatly improved tape handling. To find out more, write to us at the address shown below. We'll send you a brochure on ATR-124, our latest audio effort. Better yet, call us and we'll set up a demonstration. It's really the only way to listen to the future.

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For additional information circle #13

Sound Workshop on the right to bear ARMS.

When Sound Workshop introduced its computer automation system several years ago, we named it ARMS — a tongue in cheek acronym for the **A**uto-Recall **M**ix-down **S**ystem. At that time, recording industry use of console computer automation was focussed on the multitrack mixdown process and a system designed to aid that process would thereby provide additional "arms" for the engineer.

Technology has continued to evolve since that time, and so has the idea of using a computer to do more than just assist in the mixing process. One can spend more than a quarter of a million dollars for a computerized recording console nowadays. And the computer in that board will eliminate the use of pencil and paper forever by allowing the "recall" of virtually all of the console set-up information. A definite advantage in the creative process, but the price tag can be forbidding (even when you consider the money saved on pencil and paper).

Sound Workshop is not presently building consoles in the highest price brackets. We have concentrated our expertise on designing and building cost-effective professional console systems that in many ways outperform their more expensive counterparts. The Series 30 shown here provides a perfect example of what we do. And we have maintained this same approach regarding console automation.

Although ARMS was specifically designed to aid the recording engineer during complex mixdown situations, it actually functions throughout the recording process by providing computer control/assistance to a number of mechanical operations previously done manually, with the help of other engineers, or not at all. ARMS Automation includes the following functions:

- Automated control of channel levels (Level Write)
- Independent automated control of channel on/off status (Mute Write)
- Full In-Place Solo System
- Total integration of all automated functions into all group structures
- Super-Group

The most vital aspect of ARMS Automation is its ability to control the on/off status of each input channel totally **independent** from its control of channel level information. Even if ARMS was used just to turn channels on and off without writing level information (i.e. having the system control the actual "mix," normally the stated purpose of automation), a number of mechanical operations common to nearly all mixdown sessions would be eliminated. These include: noise gating; erasing unwanted sections on the multitrack master; selecting proper tracks from duplicate performances; switching between "time shared" tracks; changing EQ, Echo, Panning etc. during specific "sections."

Another major asset of ARMS Automation is its computer-controlled sub-group system named **Super-Group.** Super-Group permits all grouping functions to be controlled by the computer, eliminating previously awkward systems of group selection, modification and visual confirmation. Conventional systems require the user to scan each input module's thumbwheel switch (or digital display) to determine which inputs belong to a given group, an often cumbersome process on today's larger consoles. With Super-Group, the user merely pushes the button on any channel and all members of that group light up — **instant visual group confirmation!** Other Super-Group features include:

- Solo **Dim** Allows all channels except the one (or ones) soloed to be attenuated by any preset amount.
- **Negative Grouping** Allows instant selection of a group consisting of all channels **except** those selected.
- **Grand Master** Any fader may be established as the console Grand Master.
- **Local Control** Any Group master can be changed over to local channel control without affecting the group level.

ARMS Automation is available in the Sound Workshop Series 30 and Series 40 recording consoles. The exceptional performance and practical value of these consoles can be confirmed by sitting behind one of them or by consulting with a studio who owns one. Twenty-four track automated consoles from Sound Workshop start at less than $25,000.

Sound Workshop's ARMS Automation is genuinely innovative and amazingly cost-effective. Much more than just a mixdown aid, it provides a variety of functions not found in other systems regardless of cost. And Sound Workshop will soon be introducing DISKMIX™ — a disc-based storage system designed to augment ARMS with the capability to store and merge a number of mixes while providing off-line editing, computer control and storage of session documentation.

Just a part of your right to bear ARMS.

sound workshop

Sound Workshop Professional Audio Products, Inc.
1324 Motor Parkway, Hauppauge, New York 11788
(516) 582-6210 Telex 649230.

For additional information circle #15

Mick Fleetwood in Ghana

recording should be *clean*, with a minimal amount of circuitry to interfere with the signal flow from mike to tape stack.

With Ezratty calling the shots, Dashut and Youdelman gave their preliminary okay to a Sound Workshop Series 30 console, custom installed in three main-frames, allowing each to pack into a foam-lined road case measuring approximately 3 feet by 4 feet. The 28 transformerless inputs provide clean control of the signal, and Dashut and Youdelman, both of whom can claim experience on the highest caliber consoles available, had high praise for the Sound Workshop board. Keeping in mind that air freight to Ghana runs over $3.00 per pound, the light weight of the Series 30 console was a big bonus.

The rest of the equipment Effanel supplied for the recording was no less efficient. The modular console was accompanied by two tape machines: a Stephens 821-B 24-track; and a Stephens 811-B 16-track machine. The decision had been made to record 16-track at 30 IPS with no noise reduction, but Ezratty brought the 24-track headstack in case Fleetwood needed it. Stephens machines are modular, and feature a pinch rollerless tape drive, while the electronics are located in a separate heavy-duty flight case housing. The remainder of the gear provided for the African Safari included: two pairs of Rodgers LS-3/5A BBC Monitor Speaker Systems; two Quad 405 amplifiers; two Nagra 4-S stereo tape recorders; two Inovonics Model 201 limiters; 12 pairs of Sony DR5-M headphones; a Sony TCD-5 cassette recorder; 18 AKG mike stands; 30 mike cables; three 10-pair, 150-foot snakes; plus miscellaneous spare parts, electrical accessories, tools, test gear, fuses, etc.

The cargo bins to Africa also contained 100 rolls of 3M 250 2-inch tape, 50 rolls of 250 ¼-inch tape, and cassettes. A total of 10,000 pounds of freight, which accounted for less than one third of the payload to Ghana, filled two containers, and was comprised of recording gear, numerous musical instruments, the tape and film stock (provisions had been made for Fleetwood to supply a certain amount of film stock as payment for use of the soundstage) and last, but not least, survival gear that could have outfitted Stanley's search party for Dr. Livingstone!

Having pored over equipment lists for weeks, Ezratty, Youdelman, drum technician Tony Todaro, and Jim Barnes (an expert stage rigger who was to function as "quartermaster and liason" for the entourage) were confident the loaded cargo bins contained everything they needed to record in the wilds of Africa for six weeks. They left the freight offices at LAX and enjoyed a festive dinner; they were on their way.

But . . . Murphy's law claims that whatever can go wrong, will go wrong. Just prior to the sound crew's departure a few days after the gear had departed the U.S., it seems an overzealous American musicologist working on behalf of the Fleetwood project had insulted some Ghanaian officials. The government spoke of their fears of harm to the visitors and their equipment.

"All I could think about was the gear, thousands of miles away, on the verge of being sacrificed in some jungle ritual," says Ezratty in retrospect. "Fortunately an official of the

Producer Richard Dashut with Mick Fleetwood at the customized Sound Workshop Series 30 console

American Embassy quickly smoothed the way for our departure. As it turned out, I don't think I've ever felt safer or more secure for myself or my gear than the six weeks we spent in Ghana."

A week late, Ezratty and his technical cohorts were en route to a surprising, festive celebration of their arrival at the Accra, Ghana, airfield, complete with tribal dancing and a retinue of accompanying musicians. After a five-day hiatus in Copenhagen, the recording equipment arrived safely, and the crew supervised its transport to the film soundstage. Describing the space Fleetwood had secured for the recording, Ezratty says that "it was ideal for the project — an older building, circa 1950. The room had a very viable sound of its own, and an interesting design element in the form of very large partitions that swung out from the walls, allowing us a lot of flexibility in the shape of the room. With the exception of the electricity supply to the building, very few of the technical facilities were operable. There was a small room on the second level of the building that just happened to have a window looking down on the length of the soundstage. Though it was small and not so convenient, it suited our needs."

. . . *continued overleaf* —

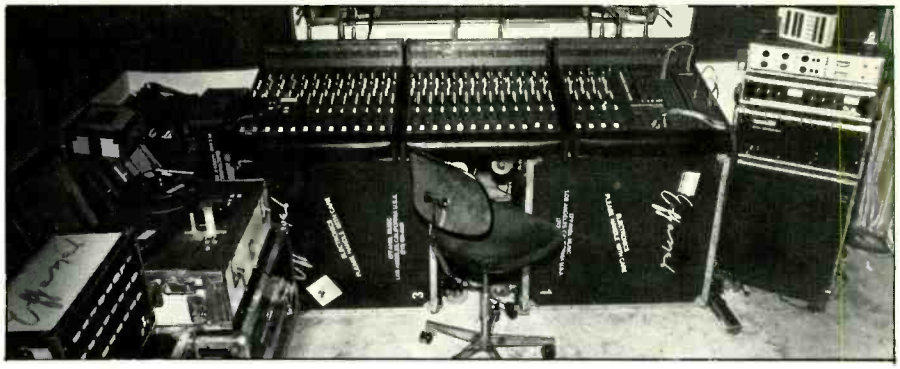
views

Mick Fleetwood in Ghana

In the Wilds of Africa

The first week of the trip was spent setting up the equipment and testing everything after its long, probably not-so-smooth trip. Once the band arrived some trial recordings were done in preparation for the sessions. The room was sufficiently large for the Anglo-American rhythm section of Fleetwood and Todd Sharp on guitars, and George Hawkins on bass, to maintain the same set-up, while allowing more than enough space for all the African musicians, singers and dancers to arrange themselves in front of Fleetwood's drum area. While the dancers may not be heard on *The Visitor*, their presence was a necessary adjunct to the player's performance in the studio, tradition being what it is in Ghana.

Ezratty and Youdelman took advantage of the room's swinging partitions, and the 30-foot ceiling gave them sufficient headspace to mount two AKG C414 condenser mikes about 20 feet off the floor in each corner facing the drum area, which was distinguishable by a plywood floor that had been laid over the concrete floor of the soundstage. Flanking the drums left and right were the bass rig and guitar set-up. Behind this section of the room, opposite the control-room window, the engineers mounted a Countryman EM101 mike 20 feet above the floor, again one in each corner. (Add another new recording application for the new mini-mike from Countryman!)



The Effanel Music portable 24-track recording setup in Ghana . . .

Microphones are an important consideration when taking the "straight-wire" approach to 16-track recording. Such a technique demands a selection of high-quality mikes, and engineers who understand the mechanics of a microphone relative to the instrument or voice it covers. The Effanel mike list is impressive. Ezratty is a fan of what many might consider exotica in the world of transducers; there's nothing fanatical, however, about clean signal processing. The Schoeps CMC431 microphones are solid-state, transformerless models that are as hard to find as they are gratifying to use. Many engineers in the business would sell their mother to get their hands on the four AKG C-12 tube mikes of which Effanel is so fond. (This attitude toward mike selection is in tune with the concern to save weight when shipping such a recording system around the world; a good microphone weighs less and takes much less space than the average piece of outboard gear needed to compensate for ineffective means of signal transmission.)

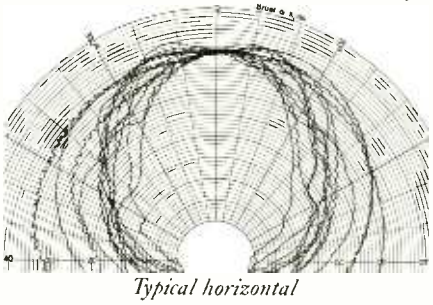
In the past, the small room on the second level of the soundstage had probably served as a projection room. The Sound Workshop board fitted well, and there was room enough for the engineers and the band. One problem still remained, however, that nothing in the survival gear could rectify: the building's air-conditioning had long been defunct. It sounds like a bad joke: "How do you keep a 28-input console, two tape machines, two power amps, a full crew and a 6½-foot drummer cool?" answer: "*Anyway Possible!!*" It's understandable that there was some air of urgency to finding a solution (if you'll pardon the pun). As the tape machines heated up in the sweltering African clime, the barely detectable, but nonetheless audible, distortion of increased wow could be heard during playback. The Ghanaian technical crew got to work immediately and, for the first time in 20 years, made the archaic air-conditioning system work in the control room, augmented by heavy-duty fans to maintain ventilation. The tape-machine wow disappeared, and it seemed that things were going exceptionally well until, according to Ezratty, "the console began blowing IC's like popcorn! It's not the kind of problem you're apt to encounter often, but every time I replaced one, it would blow again."

Using every bit of maintenance know-how available to him, Ezratty, as systems engineer, began the nailbiting process of trying to isolate the culprit that would eventually pop a total of 22 IC's! He needed advice quickly, but soon discovered that it takes about two months to get an overseas call through from Ghana. All of which called for

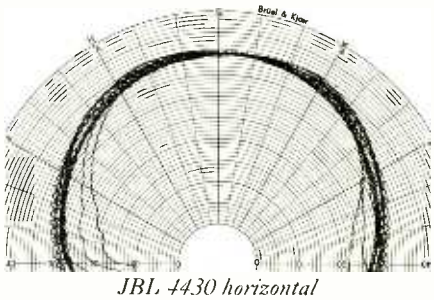
. . . *continued overleaf* —

Before you invest in new studio monitors, consider all the angles.

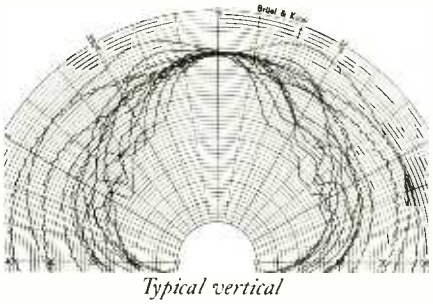
Polar response comparison of a typical two-way coaxial studio monitor and JBL's new 4430 Bi-Radial studio monitor from 1 kHz to 16 kHz.



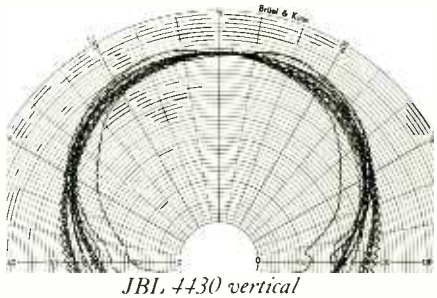
Typical horizontal



JBL 4430 horizontal



Typical vertical



JBL 4430 vertical

No one has to tell you how important flat frequency response is in a studio monitor. But if you judge a monitor's performance by its on-axis response curve, you're only getting part of the story.

Most conventional monitors tend to narrow their dispersion as frequency increases. So while their on-axis response may be flat, their off-axis response can roll off dramatically, literally locking you into the on-axis "sweet spot." Even worse, drastic changes in the horn's directivity contribute significantly to horn colorations.

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At JBL, we've been investigating the relationship between on and off axis frequency response for several years. The result is a new generation of studio monitors that provide flat response over an exceptionally wide range of horizontal and vertical angles. The sweet spot and its traditional restrictions are essentially eliminated.

The key to this improved performance lies in the unique geometry of the monitors' Bi-Radial horn.[1] Developed with the aid of the latest computer design and analysis techniques, the horn provides constant coverage from its crossover point of 1000 Hz to beyond 16 kHz. The Bi-Radial compound flare configuration maintains precise control of the horn's wide 100° x 100° coverage angle.

1. Patent applied for.

And the Bi-Radial horn's performance advantages aren't limited to just beamwidth control. The horn's rapid flare rate, for instance, dramatically reduces second harmonic distortion and its shallow depth allows for optimal acoustic alignment of the drivers. This alignment lets the monitors fall well below the Blauert and Laws criteria for minimum audible time delay discrepancies.

But while the Bi-Radial horn offers outstanding performance, it's only part of the total package. The new monitors also incorporate JBL's most advanced high and low frequency transducers and dividing networks. Working together, these components provide exceptionally smooth response, high power capacity, extended bandwidth, and extremely low distortion.

Judge For Yourself

Of course, the only way to really judge a studio monitor is to listen for yourself. So before you invest in new monitors, ask your local JBL professional products dealer for a Bi-Radial monitor demonstration. And consider all the angles.

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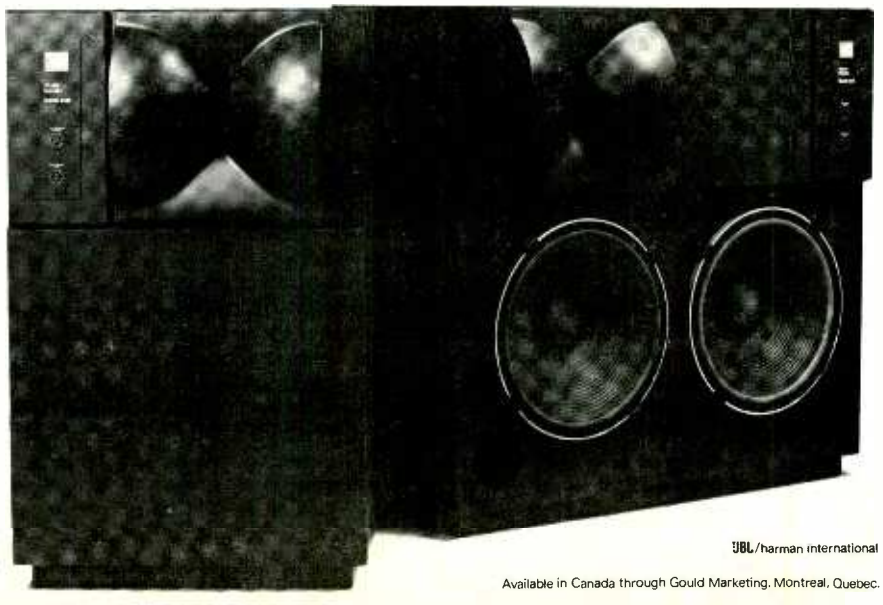

JBL Professional Products Division

For additional information circle #17

JBL/harman international

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some creative maneuvering. Eventually, Ezratty secured the graces of a local missionary who owned a ham radio, and lost no time in contacting his friends at Sound Workshop. It's not permissible to transact or make business arrangements via short wave, so Ezratty was forced to disguise his description of the problem, and his request for an emergency shipment of IC's from the console manufacturer in New York. The Sound Workshop engineer suggested that a grounding problem was the cause of the trouble. Sure enough, Ezratty found a ground-liftless direct box to be the wrench in the works. The errant DI was quickly replaced with one more sympathetic to the Series 30 electronics. The requested IC's arrived on the Sunday night flight from New York, never to be used.

A Range of Sounds

If anyone had to adapt to the unusual location, it was the members of the engineering staff. Their's was the task of miking a battery of percussion and wind instruments they'd never seen — much less heard before their arrival in Accra. Capturing the colors and textures of these native instruments was a challenge ably handled by Heider engineer Billy Youdelman. The variety of instruments to be recorded ranged from percussion "toys," made of discarded brake drums, to hollow tree trunks. How many engineers would pass up the hands-on experience of recording a xylophone-like instrument, made of gourds and tuned by filling them with the proper density of spider webs! (Studio to control room: "Can we get some spider webs in the cans please?") Youdelman simply miked in close, and utilized ambient microphones to fill out the sound.

Under these conditions the recording set-up had better be a constant; from all accounts the recording system at Accra met every demand placed on it, up to and including recording a large group of Ghanaian singers Fleetwood assembled in a field hard by the soundstage doors. Ezratty and Youdelman ran their mike lines outside, placing close-in microphones around the semi-circle of Africans, ambient mikes being set up some feet in front of the group. During what is one of the most enchanting tunes on *The Visitor* album, "O'Niamali," the vocals of the African singers are heard clearly and distinctly out doors, where, Fleetwood points out, they were meant to be heard. To Fleetwood's credit, the recording captures quite audibly a native spirit and openness.

Seventy rolls of 2-inch tape were used during the six weeks of location recordings in Ghana. Safeties were made on the remaining tape stock, and shipped to L.A. The rest of the tapes went to Europe, where the project was mixed after a small amount of overdubbing. By and large, no one was more surprised with the final outcome of the recording than Mick Fleetwood himself. Ostensibly, the burden of the project rested on his shoulders but, like most operations, the sign of a good leader is his selection of support personnel.

Richard Dashut, as Fleetwood's co-producer, engineer and long time associate, could not say enough about the approach and technique of both Randy Ezratty and Billy Youdelman, not to mention the recording system. That such a small, portable system could be put together at all, let alone in the six weeks it had taken Ezratty to close the cargo bay door and send it to Africa, made its impression on the West Coast-based producer. Dashut was especially surprised about the quality of the small Rodgers monitor loudspeakers Ezratty had convinced him could do the job.

"I hate small speakers!" states Dashut. "I want to hear things *loud*. Randy kept telling me I'd like them, and he was right. I was blown away when we started listening to the masters in Europe on a pair of UREI 813's. Everything I had heard on location was right there. I was impressed." ■■■

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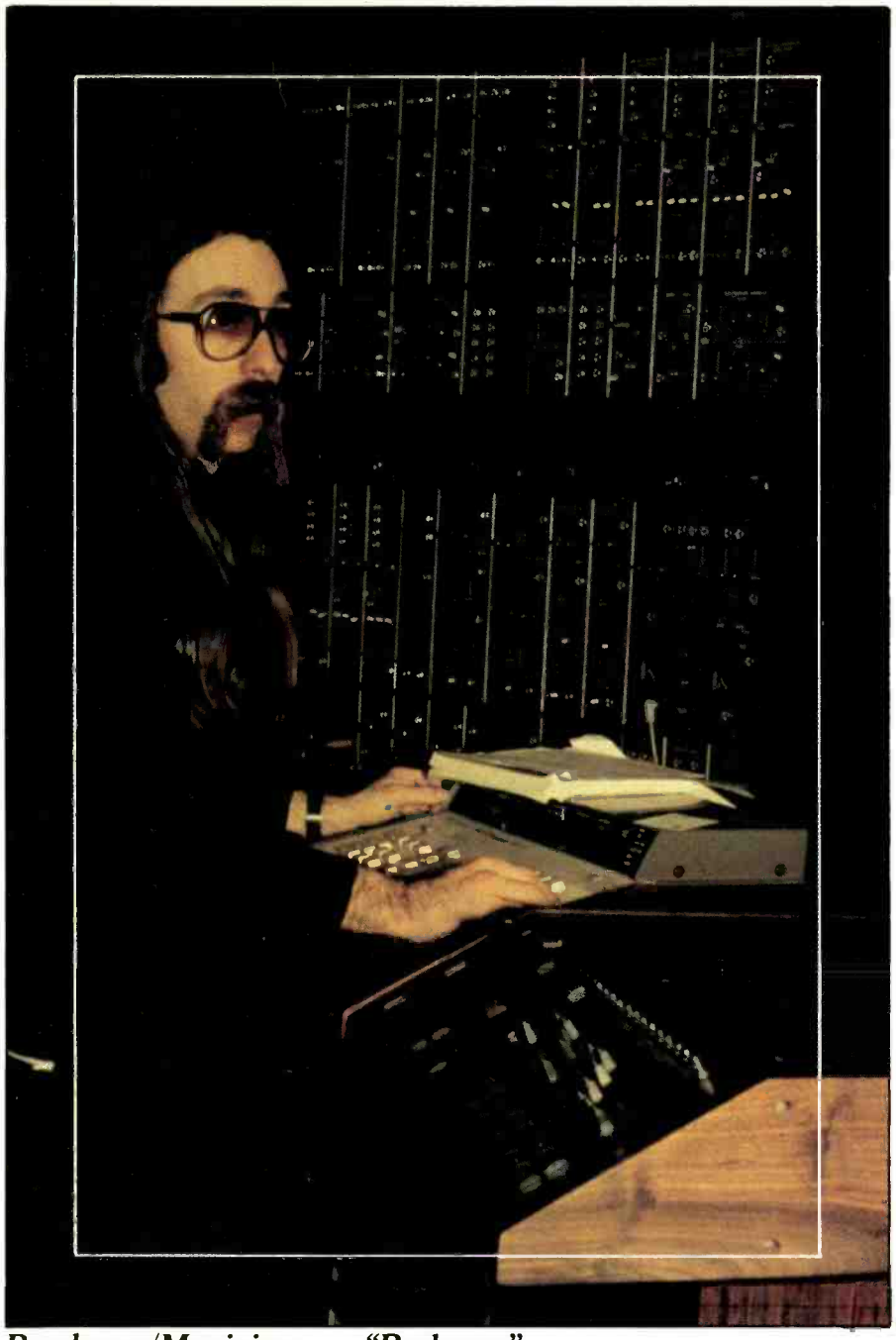
For additional information circle #19

by Robert Carr



Producer/Musician . . . "Reducer"—

JEFF BAXTER

In an age when specialization seems to be the norm throughout the recording industry, a refreshing change of pace is the person who refuses to be categorized. Such is Jeff Baxter — versatile L.A. session guitarist, producer and engineer. Baxter's reputation as a lead guitarist and steel player was established during his two-year stint with Steely Dan, his next four years with the Doobie Brothers and, whenever his spare time permitted, as session player for artists such as Donna Summers, Barbra Streisand, Dolly Parton, Glen Campbell and Harry Nilsson. His knack for customizing guitars and amp electronics lead naturally to the pursuit of audio engineering, just as his first-call status with top record producers afforded him the opportunity to learn the best production techniques of a highly competitive and eclectic industry. To date, Jeff Baxter's production credits have included Livingston Taylor, Nazareth, Nils Lofgren, Billy and the Beaters, and most recently, Sneaker. *R-e/p* met with Baxter at his Mandeville Canyon home recording studio during a brief respite from his hectic pre-production schedule for Carl Wilson's upcoming solo album.

R-e/p (Robert Carr): In a recent interview you are quoted as saying that after producing Livingston Taylor and Nazareth, "All of a sudden it's like I'm a 'reducer' now." Do you really visualize the producer's role as that of a "reducer" — cutting the fat out of an artist's music?

Jeff Baxter: You have to reduce whatever the band's statement is, into something that you can hold on to. It's physical in the sense of a disk; you try to reduce all their talent, and all they want to say, into 10 songs. In a lot of ways it is reduction.

R-e/p (Robert Carr): What about reduction in the sense that the band plays too much, or they play parts that clash? That, too, would be a reduction in the amount of music, a cleaning up?

Jeff Baxter: You do both. Sometimes when I produce, I find I do almost as much songwriting as anything else. I'll hear something a certain way, and want to make a change — maybe take out a bridge, or put in a couple of verses.

Up front, I try to get a good overview of what the band can do. For instance, if they're having trouble on a particular song — like getting a good bridge, or trying to get the verse to come out right into the bridge — I'll say, "Remember this particular song you wrote? Why don't you take the bridge from this song, put it in here, and see how that works?" Or, "Try the intro and the outro from this tune over here." That's reducing in a sense, but you also add, too.

R-e/p (Robert Carr): The three albums we've included as examples of your production work comprise the new Sneaker album; Billy and the Beaters' live record; and Nils Lofgren's Night Fades Away. *As such they represent a pretty unique cross-section, because the set includes one example of a self-contained group, one artist with studio players as a regular group, and a solo artist with a pick-up session band, respectively. Is there one of those combinations that you prefer to work with more than the others? Maybe one kind of project is easier to produce?*

Jeff Baxter: Well, bands are usually a little bit harder to do than a single artist, because you have to focus on more people, and give them more input. But, in my experience, Nils Lofgren was a sweetheart to work with, and so was Sneaker. This is Sneaker's first album. They wanted to do everything right; they tried really hard, and communication was no problem. But that's a tough question to answer.

I don't think it's a matter of one being *easier* than another. I think the main point has to do with what buttons a project concept pushes in you. Certain combinations push certain buttons; they make you produce differently, and bring out different parts of your ability.

To me, something that's easy may

. . . continued on page 33 —

Photography by: Kathy Cotter

Beginning as a Nashville session musician with a burning desire to be a producer, Larry Butler watched and listened. His first break came when he got a producer job with Capital Records in Nashville. The first record he ever cut, with Jean Shepard, was a hit. Since then he has cut over 50 gold and platinum records as producer for CBS, Johnny Cash Productions, Tree International, United Artists and now as an independent. His recent relationship with a man named Kenny Rogers, has produced hits like Lucille, She Believes In Me and The Gambler. Larry won the Grammy Award as producer of the year in 1980.

ON DEVELOPING A STYLE

"When I started producing, I was producing like everybody in town. I started to produce a record like Billy Sherrill would do it or like Owen Bradley would do it or whatever. And then one day I listened to a lot of records I had done and I thought now wait a minute. If somebody wants a record that sounds like a Billy Sherrill record they can go get the real thing. So I started producing the way I wanted to produce. It was a great lesson for me. It was a big turning point in my career. I think that nobody is really going to sell or really succeed until they reach that point where they're putting themselves into it, instead of making a copy of someone else's work."

ON REACHING THE LISTENER

"I'm a believer in the simplicity of a song. I believe in laying something in somebody's lap they don't have to search for mentally. I've said this before, if a guy's driving home from work he's got a million things on his mind. He's got to spank the kids when he gets there. He's got a flat tire on the way home. And through all of this there's a song. He's got his radio turned down kind of low and a song cuts through all of that and he finds himself humming along with it. When that happens you've hit one in the upper decks."

ON KENNY ROGERS

"Kenny is such a universal name, such a big name. I try not to let any prejudice enter into comments about Kenny because we've been so close, but I guess he has to be the strongest single male artist in the United States. I can't think of anybody that's reaching the mass of people that he's reaching and I think it's unfair that people say he's the new Elvis. Well, there's never going to be another Elvis. There's Elvis Presley. That's it. Forever. But as far as sales, you might compare them."

ON KNOWING WHEN TO STOP

"I think the most common mistake for an engineer and producer to make is maybe not really realizing the take when they've gotten it. Sometimes going too far because they're looking for that emotion or magic. Sometimes you can have it and not realize it. Sometimes you can have maybe one guitar part that bothers you, so you go ahead and do another take. Well, you have gone by the one that had the feeling, the one that had the emotion."

ON TAPE

"I use the philosophy and theory of surrounding myself with people who know what the hell they're doing and letting them do it. I let the engineer do his job.

The only things I've heard them say about 3M is it's dependable, you can trust it, you don't have to worry about it. When you're spending money and you get good service you're not going anywhere else. You're going to stay there with whoever it is.

I just know 3M has always been very, very open for ideas and suggestions. It's just like "money making music." Three M's. That's the way I think of the tape, because it works and it sounds great."

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actually be something that's more difficult, because it presents more of a challenge, and therefore it's easier to act upon. The complexity pushes all your buttons, and you go, "Well, that all makes sense to me." As opposed to: "I'm gonna have to listen to the tunes a couple of times, and see if I even want to do it."

R-e/p: So all these groups were ones that you heard, and immediately said, "That's a project I really want to do?"
JB: Sneaker: yes. Billy and the Beaters: I was sort of in the band. I'd play with the band once in a while when I wasn't working, and just thought it would be a nice idea to produce the album for Billy Vera. Fortunately, the record worked out for him; it got him a little bit of notoriety.

R-e/p: I would think that your approach would be different for each kind of project. For example: on Nils Lofgren's album, Night Fades Away, *you'd have to go out and find all the session players.*
JB: I didn't have to go very far. I work with those guys every day, or at least pretty often.

R-e/p: Wouldn't that kind of a situation offer you more freedom? If you need a guitar player, you can do some guitar parts, or you call session guitarist Elliott Randall. A bass part? "Let's get David Hungate."
JB: Freedom doesn't necessarily mean options to me. You can have a lot of options, and no freedom.

R-e/p: Having a restrictive form can actually make you be more creative?
JB: That's what I was saying when I mentioned that it pushes more buttons. The more restricted a project is, the more it brings your . . . producers salivate aurally. An exciting project brings all your audio 'digestive juices' to the fore and, in essence, makes the work easier. I think projects that present themselves as being cut and dried can sometimes fool you. That's when you begin to take things for granted; as soon as you underestimate anything or anybody, you're in big trouble.

R-e/p: Can you put into words how your approach would differ from one type of group to the next?
JB: Sneaker didn't even have a record deal when I first thought of working with them. I just believed in the band, because I thought they were really good. I went into the studio with them on spec.

Village Recorders [Los Angeles] gave us a good rate, but we started recording at midnight — midnight to eight in the morning — because that was the only hours we could get. We had to remain creative while being completely exhausted, and still make a decent record. We did a couple of tunes, and shopped them around — not demos, but the real thing. You can't wait around, you know.

R-e/p: Did you do much pre-production with the group?
JB: Mainly, not enough, because of finances, and because I had to get right into the project.

R-e/p: Did you find that going into the studio with an unsigned group was anymore difficult, because of them perhaps not having that much experience of working in a studio?
JB: No. As a matter of fact, in a lot of ways, not having had the band in the studio before made it a little bit easier for them to adapt. I tried to make the studio comfortable for them, but the circumstances we were working under were tough. They came through, though; I can hardly wait to get back in the studio with them.

Nils Lofgren presents another situation, since I had to work on songs with him. When he came out here, we worked together picking the different musicians and, during the actual recording, the primary focus was

"Inspiration . . . has a lot to do with learning to enjoy being in the studio . . . really enjoy recording!"

getting tracks — apart from Nils' — that I thought were strong enough for his vocals. I had to cut tracks that could push him to sing. The guys played so good that it inspired him to a great performance. The music got him off, and he really worked at getting some great tracks.

R-e/p: Did you have a chance to rehearse with Nils' musicians, or was that a matter of: "Here's a chart," or "Here's a head arrangement?"
JB: No. Ritchie Hayward [session drummer] came up a couple of days, put some work into the parts, and helped us out. We laid some tracks down here at the studio in my house and, when we got them the way we wanted them, we went down to Cherokee and re-recorded them there.

Ritchie is one of those guys who's more than just a studio musician, or more than just a monster player. Bassist Neil Stubenhaus, who I'm using on Carl Wilson's new solo album, is the same way. If Neil has the time, he'll come up and work on the tape with me and the artist, or just help with the arrangement to really get a handle on it. That way, when he goes in to do his parts, they're the best they can be. That's one of the reasons I pick guys like that to work with; they give more than 100%. I think I'd like to do the next album with Nils in a live situation. He's got a hell of a band now — some cats from LA.

To do the Billy and the Beaters' album, the band had to be loose enough to keep the personality of what was the band, but tight enough to get good performances when we played the tunes for the record. I wasn't really interested in telling anybody *what* to play; I loved the way everybody played anyway, so there was no reason to. I thought every musician understood his responsibility.

As long as the arrangements were there, it was just a matter of going to The Roxy [Los Angeles rock club], and going through the whole horror show of doing a live record.

R-e/p: So really very little production was required?
JB: No, that's just the recording of the basic tracks. Then someone had to decide which performances out of 108 were going to be on the record, and how the ones that were there could be fixed up. We had to figure out what was needed, and then mix it. It's true that the recording only took four days with the Record Plant Mobile #1, but the mixing took me damn near three weeks.

R-e/p: Did you do a lot of overdubs on the basic tracks?
JB: Not very much. I doubled the horns sometimes, because we had mike dropouts. Rather than lose consistency, I used a thread of overdubbed horns so they wouldn't drop out altogether. There were a couple of overdubbed solos, and that was it. We used the same horn section; the same guys playing the same parts, and mixing it in such a way so as to boost the track up a little bit. In the live situation, I had Bruce Robb, an excellent engineer, doing the engineering in the truck for me. But to have one guy trying to bird-dog 48 tracks in real time — this is not computer time, but *real* time — it's damn near impossible. He did a great job with what he had to do.

The other problem I had with the Billy and the Beaters' live project was that I was playing with the band at the same time as producing. I had people yelling at me, and me yelling at them, and two sets of headphones on: a stereo left and right mix from the truck on one set while, on the other, the right mix from the house monitor on my left side, and the comm line for the audio and video on the right. I would switch back and forth between those two sets of

"Picking the [recording] rooms is probably one of the all-time most important jobs in making a record."

cans. I had never done it before, and I wanted to see if I could. So, I was playing guitar on stage, drinking beer — having a great time talking to Bruce in the truck. We were all singing over the intercom.

R-e/p: Other than the singing, what did you use the comm line for?
JB: Cues, and solos; it was a hot line. Bruce would tell me, "Hey, I'm getting a little too much edge out of your guitar," or "Tell George to turn down. Tell Beau the mike's off his snare drum." We were keeping information flowing back and forth just like you would be in the studio. That communication *has* to be there.

R-e/p: Wasn't there a support crew to pass along those messages and take care of all the logistics? Didn't you have a stage crew?
JB: There certainly was; the best crew in the world — the guys from the Steely Dan road crew. Also, there was a stage manager who was on the comm line, and then there were guys out in the audience watching, listening, and calling cues in on the comm line to the house-sound engineer. If a mike dropped out of a holder, they'd call up and say, "Hey, mike #23 has dropped off the ride tom." And somebody would go out there and get it. We had a couple of real hairy times, but the boys pulled through.

Half the reason for doing the project was the challenge of solving the problems. If it went easy, it wouldn't be any fun. The idea is to set up a situation that's so insane, that you have so many problems to deal with. Again, this is the situation where, if you're faced with something that's totally nuts, it brings the best out in you.

R-e/p: Directly before the gig, did you feel it was important to hang out with the band as one of the musicians, or were you still in the role of producer?
JB: I wasn't doing any one thing. I guess being a producer is doing whatever comes up.

The band had been playing together for almost two years; that's why I felt so confident about the live recording. That was also one of the reasons why people liked the band, and why Alfa Records signed them. Live, the band had that easy-going, confident, well-rehearsed, but loose, rock 'n' roll feel. I left a lot of the responsibility of the playing to the players. If I had had to write the arangements and stuff, it would have taken me another couple of weeks. Luckily, that was all done ahead of time.

My relationship with the band was unique in that I knew the band from the inside. I felt that I had covered most of what would be needed to record the band live, because I could look around me; I knew the guys.

R-e/p: But, after the live album, you won't be producing the band anymore.
JB: That's right. We had done another album with Billy and the Beaters, but the record company wasn't satisfied with the tapes, and with The Beaters in particular. Billy Vera went to Muscle Shoals to make a record with the rhythm section there, as well as to work with Jerry Wexler; Billy had always wanted to work with him. I guess they had worked together a long time ago when Billy was first getting started.

Now Billy has another band which is comprised of some other players from LA, but still the same horn section and piano player.

My situation with the band was, interestingly enough, that I'd work with the band when I had time to fit it into my schedule. The problem now is that I can't split up my time. I'm not working directly with the band, because I have other priorities. But it's a great band. Makes good music.

R-e/p: There's no hard feelings?
JB: Well, for me it was different. I can't really speak for the rest of the guys in the band. It was fun for me, and that includes the fact that I had a chance to produce a record for them, and make something happen for a band that was jamming at The Troubador. That's kind of a great charge; that's all I need, really. So for me, it turned out okay.

R-e/p: What attracts you to a group? Is it the same criteria that you'd use if you were going to play in the band?
JB: A lot of the same, yeah. I like anything that I can sit back and rock to. It's just a gut feeling; if I like it, it gets me off, and I think it's cool, then I'll do it. I think it's better to be bound by that than, "I'm an R&B producer." Or, "I produce jazz records."

R-e/p: So you'd take a shot at producing just about anything?
JB: Absolutely. I was at Air Studios in Montserrat doing Nazareth, and got a call from the boys at A&M. "Listen, we're doing a live satellite broadcast from the A&M soundlot to Tokyo and the rest of Japan for Yellow Magic Orchestra. It's going to 13 million people. Do you want to produce it?" I said, "Sure. When is it?" "In three days," they said.

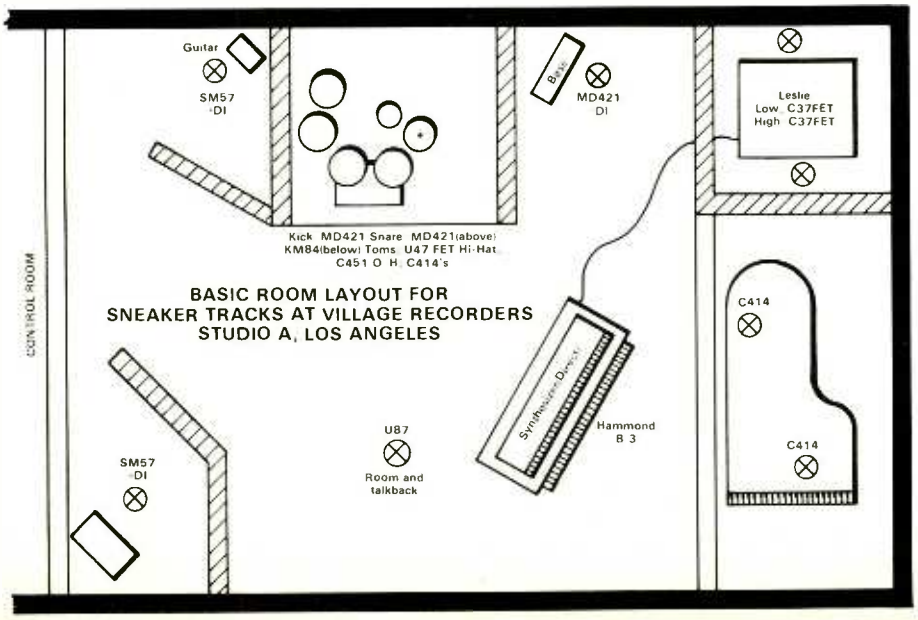
I'd never done anything like that in my life, but I put a bunch of guys together that I could trust, went in, and that was it.

R-e/p: It was sufficient pressure for you?
JB: It pushed all the buttons. It was insane. Here's what happened:

There were two feeds: an audio feed going to the 24-track tape machine; and a separate mix with a summed and bussed mix to the video truck. All the drums were on one fader, bass on another, all guitars on a third, all keyboards on a fourth, and so on. Which made their mix much simpler. Well, it didn't matter, 'cause the video truck went down. We had to immediately switch over to put everything through the mike pre-amps to the tape machine, and use the rest of the console as a live mixing desk. We did all this in five minutes.

But see, to me, that's the whole reason for doing anything. If you do something you know is going to go right, it's not that much fun. The fun is coping with problems.

Yellow Magic Orchestra was a strange band — all computers and synthesizers. I'd never done anything like that in my life. We had a big fat sound on the drum machines. It was



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For additional information circle #22

Jeff Baxter

by Robert Carr

great. We were doing a video of the band, transmitting the whole thing to Oakland, and then blasting it out to Tokyo via satellite. The band was sequencers and synthesizers; all clocked together with "beep, blip, burp, bip, beep, bimp, beep" (sings strange sounds at different pitches).

R-e/p: I was going to ask if you were technically oriented, but I think that's pretty obvious.
JB: [Sounding like an addled junky] If it lights up . . . I have to have one!

R-e/p: Do you think that helps you make better music?
JB: No. It helps me make music better.

R-e/p: I'm thinking more in terms of knowing how to run the equipment. Does that help make the music better?
JB: Say my friend Murray Adler comes in with his $92,000 Stradivarius, and I want to do a Gypsy violin part. If I spend two hours getting a great sound, I don't want to hear that song on the radio, and have the violin sound like dying swans. I want it to sound like what it sounds like. So the fact that I know how to get it on tape, how to get those 24 tracks mixed down to a two-track stereo, and then on to a disk, is going to be to my advantage. That technical knowledge is going to allow me to produce a record that sounds more like the original recording, or more like I originally heard it in my head. If I want it to sound like violins through fuzztones, that's what I'll do in between the violin and the record.

R-e/p: You could always hire somebody, like one of the Robb Brothers, to come in and do that for you at your direction.
JB: But the goal is the overlapping of responsibility. I delegate a lot of responsibility. When my production assistant Marylata Kastner came in to the studio, and said that the vocal on "Jaymes" [a Sneaker track] was garbage, and I ought to do it again, I said, "Okay. We'll do it again." I really didn't question her, because I knew she was telling me something from the gut. Obviously, she had to be right. In the back of my mind I knew I couldn't make up my mind about it; she made up my mind for me.

Or, for instance: I'll be recording or mixing with [engineer] Larold Rebhun, and my ears will be burned out. I'll say, "Larold, you're now the producer. I want you to keep your eye on tuning and enunciation, and I'm going to engineer." So the duties overlap.

R-e/p: The technology helps you make better records?
JB: It certainly does. It doesn't give you soul, but it helps you translate and reproduce that soul if you can actually get it at the time. I think that good photographers can be better photographers if they develop their own pictures. When you're dealing in a medium that's technologically oriented, I think it's kind of silly not to know about it.

R-e/p: It gives you more option which, in itself, increases your creativity?
JB: That's what creativity is: vocabulary. And vocabulary is an option of ways to say things.

R-e/p: Many people get into technology and say, "This idea can't work, because this piece of equipment doesn't work that way." Yet someone who doesn't know any better . . .
JB: . . . plugs it in back assward and makes it sound cool. That's absolutely right. But I don't think there is such a thing as too much knowledge.

R-e/p: Is there any particular outboard gear that you keep coming back to?
JB: I use a lot of Roland gear, believe it or not. I do a lot of consulting and testing for the company, so I know how the equipment is built. I use their tape echo in mixing all the time; it's very clean. I use the company's top of the line flangers, and the Dimension D effects unit when I want to put a little "sparkle" in the chamber. I use a fair amount of "time" — more for space than dynamics.

R-e/p: Recreating room ambience?
JB: No, excitement. Physical movement is what really makes music exciting. Listening to a symphony you hear the melody from the trumpets on one side of the stage go burning over to the violins, then over to the other side to the 'bones, then back again to the strings. Just the physics of hearing a high-hat on the left during a ballad shooting via echo to the right side is exciting. It's not gimmicks; it's the physics of moving, breathing, living.

Sometimes I'll do a song where I won't touch effects. I only like to have them available. I don't believe you have to use effects on everything. As a matter of fact, the less you have to do the better. Again we go back to the basic concept: if your instrument is recorded correctly, you really don't have to use too many effects. But if you use a little digital delay, and you split them left and right, you have an *incredible* sound.

R-e/p: For bass, do you usually go with a mike or direct?
JB: Both. I might take guitars direct, and from the amp as well, because there are so many things you can do with time devices. For example: where you might not want to use the amp sound, because it's a bit too fuzzy; the delay may be sampling all the noise in the middle, instead of the high-end. So put a little of the original dry signal through the effect, and add that to the amp sound. Since it's the same instrument recorded at the same time, your ears have a difficult time in sensing differentiation between the two; the envelopes — the attack and release — are exactly the same.

I use the same principle with the Vocoder. The voice I use to trigger it has already been recorded on tape, so the envelopes blend; it's one split signal that's routed directly to the mix as well as to the effect. Usually an effect is something you can grab on to after the fact; you hear a trail after the vocal so you can say, "Oh, I know what that is." By mixing the effected signal with the original, the two blend perfectly, and there's no way to tell. It doesn't take away from the continuity; you're not taking anybody's attention away from the music.

Say you want to do a real nice stereo effect on an acoustic guitar, and put some chorusing on it, too, but there's so much ambience in the room that you pick up more ambience through the chorusing than you do the guitar. If you're using an Ovation acoustic guitar, for example, and you've got two tracks directly off the Ovation's electric pickup, you can process those, and split them left and right. All of a sudden, in addition to your acoustic sound, you've got just the right amount of clarity without sacrificing the sound of the acoustic guitar itself.

For vocals there's a trick I learned from a good cat by the name of Jeremy Smith, an engineer. The big problem in mixing is trying to get enough track around the vocal so that it sounds "ballsy." If you try and get centered enough, you can't get the track loud enough. We used two separate microphones, and two separate tracks of the same performance, and that way there is no vocal center — the vocal seems to come from everywhere. You can have a lot of track around it, but still hear the vocal *very* clearly.

R-e/p: You've been quoted as saying that you never like to play a solo the same way twice. When you produce a group with guitarists, or other soloists

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For additional information circle #23

for that matter, do you try to keep alive that kind of free environment for the players. Take Sneaker, for example. Do you want to give them the freedom, or do you keep polishing and polishing the same solo over and over?
JB: Solos are like mini-compositions. They can either be on-the-spot, or thoughout ahead of time. That's only a very personal opinion when I say that I don't like to do the same solo twice. If it's a solo that's a particular piece of music, then I like to reproduce it.

R-e/p: Like the solo on Donna Summer's "Hot Stuff" single?
JB: No. That was just a few beers, the last solo of the night, and burning to do the gig. That was "run the tape machine and roar!"

R-e/p: Is there a certain approach you take with guitar players, or soloists in general?
JB: Yeah. I give them all the space they want — pure and simple. If they ask for input, I give it to them; if not, they don't need it.

R-e/p: Whatever they feel is good is okay with you?
JB: Well, up to a point. For solos, again, that's just a particular expression. Unless it's outlandishly out of place, I see no reason why not to have it.

Rhythm guitar playing is a different ball game altogether. That's part of the basic tracks; that's part of the music — part of the map. I try to work everything out in pre-production, except for the solos.

For Nils Lofgren's album, a lot of the tracks were cut live. As a matter of fact, on two tunes I had nine guys in this little studio — Studio 3 in the back of Cherokee. There was one drummer, one bass player, four guitarists, and three keyboard players. I wanted to just do the record. I wanted to hear it. *BIG!* [Laughter] A lot of the rhythm tracks — all the material with Ritchie Hayward — we did in Studio 1 with all the players. We overdubbed solos, but most of it was live.

Sneaker was more layering, because we had to work in different studios. I figured I would get the continuity of the album in the mix.

R-e/p: Continuity in what sense?
JB: Continuity in a movie is: "The guy has to be wearing the same shirt in each take of a single scene," or else the scene isn't believeable. With the music, I don't want to take away from the impact of the record by using a studio where the sound is so completely different, that it distracts the listener. Picking the rooms is probably one of the all-time most important jobs when it comes to making a record.

Once that's settled, I would go for whatever sound I could get in the room I was in, and worry about blending it later. If I keep in my mind certain parameters within which I like to record, then I can record just about anywhere and get a good enough sound — so I could layer in San Diego, or cut vocals in Montserrat.

I've done so much recording out of the country, in strange places like the Bahamas, Montserrat, Canada, and so on, that I've gotten to the point where I know most of the mixing consoles. If I have an overdub in the middle of the Indian Ocean on a console I'm familiar with, as long as I've got my tones, I could care less where I am.

R-e/p: Several producers have commented to us, explaining their production technique, "We'll record the whole band, but we're just going to keep the drum track. Then once we have that, we'll record the whole band for just the bass track, and add that to the drums." And so on with the rest of the basics. Do you hold with that theory of recording?

"Technology! . . . Sure you're surrounded by all this technical gear [science] . . . but making music is an alchemical [magic] process . . . getting that spark that transmutes guys just playing to a song, a record . . ."

JB: Well, it depends on the musicians you're working with. For instance: yesterday I was cutting some tracks here at my home studio with Carl Wilson, and Neil Stubenhaus, who I have playing bass. Neil is a *monster* bass player but, for some reason, he's more of a monster when he's overdubbing. So I cut the track with the band, and then Neil comes in and overdubs the bass part.
[At which point the phone rings, and it's contractor Gene Simmons asking Jeff Baxter to fill in for a three-day date behind Diana Ross — a project he'd like to do, but he's scheduled to rehearse with Carl Wilson.]

I always try to do my best on each project. I'm not trying to overbook myself. Usually you have to book a lot more than you can do, because a lot of gigs don't come to fruition. You have to cover yourself. I like to work hard; it makes me feel good.
[Another phone call — this time someone calling for his advice on a documentary about Carl Wilson.]

R-e/p: When you're working with a particular player, is there a method you employ to draw emotion out of him for a solo?
JB: Certainly. I try to inspire them with their own playing.

For example: Nazareth is a band that plays virgin rock and roll. What I mean by that is that they've managed to keep alive a type or form of music for almost 10 years. But it seemed to me when I first met them that, although they really enjoyed playing, they never really learned to enjoy the *studio*. The studio was a place to make records, but the real fun was on the road. That was their concept.

I introduced drummer Darryl Sweet to the Linn Drum Machine. When I first did it, I thought he was going to kill me. By the time the second album came around, he was saying, "Where's the drum machine?"

I sat down and worked with the guys on their parts. I'd work with Pete Agnew on a bass line. I didn't write out the part, because he doesn't read musical notes, but we worked it out note for note, and recorded it. When he heard it in the control room he said, "Is that me?" I said: "Yeah, that's you. You're great."

The idea being that when he listened back to the part, I told him, "Yeah that's you when you have a really *cohesive* thought — a real concept of what your part is from the beginning, right through to the end of the song." He goes, "Wow, that's amazing." Then they began to start pushing each other to excel. The next thing I knew, they were coming to me and saying, "That drum beat is not 'on'; I'm going to replace that one beat." I found they were well into the studio!

It was the same thing with the Doobie Brothers, and getting them into the studio "consciousness." I started booking the band as a rhythm section for other artists. After some initial resistance, everybody started to enjoy the fact of being really precise, and making good music as a craft. So I think in terms of inspiration — showing the person you're working with what they can do; letting them drive themselves crazy over their own music, and enjoy it.

R-e/p: On most of the recent albums you've produced, you also played some of the guitar parts. Many times when a great player such as yourself is put with a band, that may turn off or intimidate the group's guitarist. How do you avoid that?
JB: I never play unless I'm asked. The way I see it, when Sneaker asked me to play on the record, they asked me as

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"Chain Saw Guitar, for example . . . you have to assign inanimate abstractions, symbols for ethereal [musical] concepts . . . people can't communicate very well in the abstract."

another studio musician. They said, "Hey, we need another guitar part. We'll pay you $300. Give us a guitar part."

R-e/p: So there was never the case where you said, "I hear a great part in here; let me play it?"
JB: No. Except for one tune they asked me, "Would you like to take the solo on the end? Sort of a present from us to you."

The only other time I play guitar is when I have to — like when you're doing the studio sausage scene, where you need acoustic guitar parts. Now the guitar players in Sneaker or Nils' band, for example, could have done the guitar parts, but I do tons every day. I'm laying those things down all the time. I just go in and do it, just to get it done to my satisfaction. It makes it easier for me to place things around it, and it saves me the hassle of having to put a guy in the position where he might not feel comfortable. Not everybody is comfortable playing acoustic guitar parts — "chang-ga chang-ga chang" [with exaggerated strumming motions].

R-e/p: Do you do track acoustic guitar when nobody else is around?
JB: I don't care. It's whatever has to be done. There are always a few people at sessions that aren't necessarily with the band. Most of the time they're guys that I work with, and I look to for input — other engineers, other producers.

A friend of mine might come in and say, "Hey, I've got a great idea," and go in and start playing tambourine. That's the one thing about music that I try to explain to people. It's the one part of high technology that isn't "high technology." You're surrounded by all this technical gear, but the actual making of music is an alchemical process. It's something that you do, and do, and do, and do, and repeat the same formula, the same procedure, over and over again until it is blessed with the *spark;* until it transmutates from just a bunch of guys playing away, to a song, a record.

R-e/p: After a while, those concepts actually become chords and patterns.
JB: Certainly. I once got a message on my service: "Donna Summer; Westlake Studios; Chainsaws." I know exactly that contractor Trevor Veitch has called me for a Donna Summer session, and I'm supposed to bring my Les Paul guitars and my fat amps, 'cause it's going to be chainsaw-guitar stuff. You have to assign inanimate examples, or symbols, for ethereal concepts, because people can't communicate very well in the abstract. You and I need something to relate to so that my abstraction can be seen by your abstraction. Something in common may be the color green, or blue. There are similarities.

Indeed, I think you'll find that if you just say "The Blues" or "blue" to any musician, for a very basic example, he'll think pentatonic scales, 12-bar, 16-bar, three chords. He'll have a picture in his mind of what the blues is, how to play it, and how to present that emotion. So you run the gamut of the colors, and each has a relatively common abstraction connected with it.

R-e/p: Which brings up an interesting point. You mentioned earlier that some artists or producers ask you as a player to play something "green," or "yellow." Dolly Parton once asked for "something like fireworks". Do you find yourself talking to players that way — in vague terms — or, since you've been there, do you articulate your meaning more clearly?
JB: Oh, certainly. I can verbalize my ideas much better as a guitar player. Obviously I'm familiar with the genre of the guitar. Instead of saying "Make it more blue," or "green," I can say: "Play more major seventh chords," or something like that. "Hang on the minor nine." Or, "Play around the minor nine."

Even though there are an infinite number of possibilities to a human being, there are a finite number of musical combinations. As you become

A Conversation with LAROLD REBHUN . . . Session Engineer for Jeff Baxter

Larold Rebhun has been working regularly with Jeff Baxter for about three years. In addition to being a fine engineer in his own right, Larold is also in charge of the second engineering staff at Cherokee Rcording Studios in Los Angeles. Having collaborated with the guitarist turned producer on numerous sessions, as well as knowing Cherokee studios from the inside, Larold Rebhun provides a unique vantage point from which to view the mechanics of Jeff Baxter's productions.

During the interview Jeff mentioned that he'll move from studio to studio during a project in order to utilize the different room sizes at Cherokee. How does that work, we asked?

"Cherokee has three totally different kinds of studios," Larold explained. "Studio 2 is used mostly for mixing, but there is a small studio — probably 12 feet by 8 or 10 feet connected with it that's fine for lead vocals, backgrounds, or some overdubs. Studio 1, on the other hand, is huge, and perfect for strings, and big, powerful guitar sounds like Marshall stacks. We can get a mike about 40 feet from the amp, which can't physically be done in the other rooms.

"Studio 1 is everyone's favorite. Supposedly, Olympic in London and Musicland in, I think, Munich, Germany, are considered amongst the best rooms in the world. Aside from the last album, Rod Stewart's drum sounds were done in Cherokee Studio 1. The room affords that bigger, very open, 'English-style' drum sound. We're definitely going to record the drums for Carl Wilson's new solo album there.

"The acoustics of the large room range from live-end to dead-end; live towards the control room window, and a deader sound as you move towards the back of the room. Linoleum covers the front half of the floor, and carpet the back half. One wall is wooden, and the opposite wall alternates between wood, rock, and glass. You really can get any sound you want in there but, at the same time, the room ambience is special; it's *distinctive.*

"Studio 3 used to be the Osmond's studio, and is a smaller tracking room — probably 30 by 20 feet. That room we use for a tighter-type of sound on our basic tracks, where we don't really need the ambience. A lot of disco material was cut back there.

"Since Studio 1 is booked so much, many times a producer will record in Studio 3. Then, during the mix, he'll pipe some of the sounds out into Studio 1's room, and re-mike it just for that room's sound."

Speaking of distinctive sounds, do you have a mike technique you keep coming back to while recording guitar?

"I like to use two mikes on guitars if I can. Actually, Bruce Robb [session engineer, and one of Cherokee's owners] showed me a technique that works really well with open-backed amplifiers, like a Fender Twin Reverb, and gives me a tight type of sound.

"Usually there's some thumpy, low-end going around back there. In that case, I may use one mike in front, and one mike in back behind the amplifier, with the phase reversed so that both mikes are in-phase at the board. The back microphone is generally a little more 'bassy,' while the front mike has more top-end. Together they make the guitar nice and fat.

"By putting that back mike through a really quick limiter, and hitting the strings, there's a very percussive, almost physical, feeling to the sound. We add just a little of that in with the signal from the front microphone, and put both of them on the same track. We usually know what we're going for, so the relationship between the two is worked out during recording.

. . . *continued overleaf* —

"The drums always go to the middle of the console . . . especially on a live mix, with two guys on the board, it's good to have the drums in between . . . they're really the foundation."

experienced as a musician, you pass through most of those combinations. If you've done sessions for 10 years for a lot of different people, as I have, you'll notice that many common concepts keep turning up.

Maybe 75% of the people who ask for green are satisfied with this particular kind of playing, or this kind of feeling. You realize that there are similarities between different people's abstract perceptions; there are things that link them together. You can permutate from that, and extrapolate a general set of factors that will satisfy their concept. You put those on file in your head, and can always call on those when they're needed. As you become more and more experienced, you add more and more experiences to the file, categorize them, and cross-reference them. Some people ask for green; some people ask for "hot."

R-e/p: I think it's time to get back to the safe and somewhat mundane: track assignments. Would you assign mikes differently in a live situation than you would in the studio?

JB: Absolutely! Drums get routed to the middle of the console for a live date — always. What I like to do when I'm recording live is to mix live, too. In other words, not stop and splice or fix parts, but do a running mix — where you bring things in and out instead of just going with a pre-set volume. To do that with two guys sitting at the console, it's good to have the drums in between, because the drums are really the foundation — at least for my tracks, anyway. I'm a snare pig/drum pig; I love it. I have to have them accessible.

Not to mention sometimes when you bring back tracks to different studios, tracks 1 to 8 may have certain capabilities that other tracks don't have. I've worked on a lot of Neve consoles, for example, where the first eight inputs or 1 through 16 are transformered, and 17 through 24 or 32 are not. But, basically, we have the drums in the middle so we can both get at them.

R-e/p: You work mostly at Cherokee. Is there a particular reason why you keep going back there, other than just knowing the room so well?

JB: That's a big reason — knowing the rooms and the studio so well. I use a combination of the rooms. Again, it's like instruments. Each room has something to offer, and I take what I think is the best that each room has to offer.

R-e/p: So for one date you may go from room to room?

JB: Absolutely. And usually I know the guys who are working there as well — especially if the Robbs are engineering. Say Steve Cropper is in Studio 3, I'm in Studio 1, and Roy Thomas Baker is in Studio 2. Roy might have to do a string date, and I'm in the big room — Studio 1. I say, "Okay, I have one more song to mix, but you're set up to mix." Since, let's say Bruce Robb was his engineer, and since all our mixes are set up the same, I can go into Studio 2 and mix. Everything is just the way I know it — all the chambers are set up, and so on. He can go in Studio 1 and work there for a while, and he's just as familiar with that set-up.

Cherokee is more than just the room; it's the people who work there. We spend a lot of time bouncing ideas off each other. That breeds a very creative atmosphere, especially when guys are going from room to room checking their tapes against yours, and helping you with ideas. To me, that atmosphere is worth its weight in gold.

Then I've noticed that when I get the record out of Cherokee Studios and take the tapes to Precision Laquers — or "Larry's Laquers" as we call it — and hear it on the radio, it sounds like my record. That makes me excited.

R-e/p: Do you use much automation?

JB: No, I haven't really found a use for it yet. I can't say that I don't use any automation, ever. Sometimes for real difficult soundtracks, I'll use it. But I prefer the hands-on approach. Part of the liveness in making the music is driving the console. You drive it like an airplane; you play it.

When I flew a plane, I very rarely ever put on the autopilot, because I just enjoyed flying the plane. I enjoyed being there and flying. The plane flew better with the autopilot, but it wasn't as much fun! ■■■

A Conversation With LAROLD REBHUN — Session Engineer for Jeff Baxter

"For the ambient sound, I generally position one microphone a couple of feet in front of the amp, and back off with another mike. You need to move away from the amp a little bit in order to capture some of the real deep low-end — the kind of low-end you can't get with EQ. Anything outside of about a 32-foot radius lets you pick up the really big bass waves. That's when the amp starts to open up."

Going to the other extreme, do you find you record much direct guitar?

"Jeff likes to go direct if he wants a smooth guitar sound," Larold recalls, "and especially with his steel guitar. He has so many guitars, only he knows what's going on with each one of them. For 'chainsaw' guitars, most of the time you have to use an amp.

"Sneaker's album called for some really delicate guitar parts. Those we did direct. In fact, I think we usually recorded the guitar direct on a separate channel from the amp track, just in case. The direct is handy if you want to change the amp sound later on. We just pipe the direct channel back into the studio to whatever amp we want, and re-record it.

"Using that technique during mixing means you don't have to use up another track; you bring the signal right back to the final mix. Standing in the studio, the sound from the amp is just as good as if the guy was there playing. It even gives you the flexibility of miking the amp in stereo without using up two tracks, by sending the direct signal out to the amp, and bringing the two mike signals back to the board while you're mixing.

"Roland makes a nice amp for a stereo sound; I think it's the JC120. We mike each of the speakers, and keep them on separate tracks. I don't think the chorus circuit in the amp is stereo. Instead, the chorus goes only to one speaker, while the other speaker is straight. The overall effect is stereo — as though the sound were going back and forth."

You used an unusual stereo mike on the vocals for Sneaker's album. How were they achieved?

"On 'No More Lonely Days' we used an old AKG C24 stereo mike in an M-S configuration. An M-S box sits in your lap, with two big knobs on it; one knob controls the cardioid capsule, and the other is for the figure-of-eight capsule that's pointed sideways. When you mix them together just right, it produces a beautiful stereo sound. Fortunately, we had a lot of open tracks, and were able to do all the vocals in stereo, which is unusual. The important factor though, was getting the vocalist to stand *very* still. If he moved from one side to the other, his image shifted.

"The M-S technique was originally developed for recording orchestras. It's another form of the X-Y, where you have a stereo mike with two cardioid capsules pointed at a 45-degree angle. With the setup we used, however, one capsule is pointed straight at whatever is being recorded, and the other is a sideways figure-of-eight. The mike sounds *great* in the headphones.

"Dee Robb used the technique a couple of years ago on Art Garfunkle's album, *Watermark*, I think it was — 'Crying in My Sleep', and songs like that. It's not the kind of mike you'd want for a screamer; it's just for something nice and subtle." ■■■

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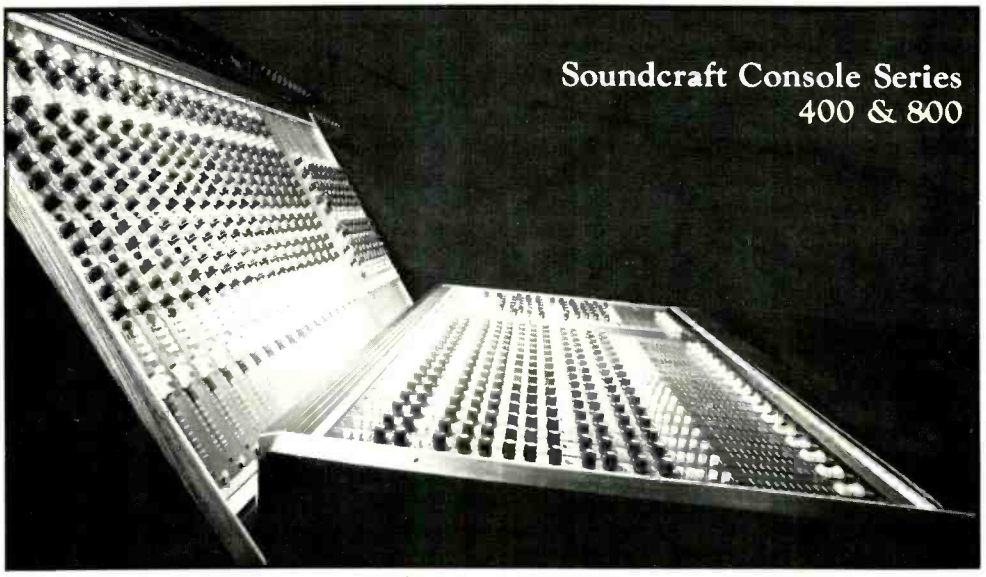

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SPECTRUM ANALYZER & EQUALIZER DESIGNS

by ETHAN WINER
The Recording Center
Norwalk, Connecticut

A spectrum analyzer can be a powerful tool for keeping your studio well tuned. Besides its obvious uses — for adjusting octave or one-third octave room equalizers — a spectrum analyzer can also be used to check the response of a recorder, or to even help identify the rumble frequencies in an air conditioning system. Unfortunately for many studios, the high price of an analyzer presents a major problem; LED versions cost many hundreds of dollars, and a good oscilloscope model can run into thousands. But, once again, the IC op-amp comes to the rescue, allowing construction of a sophisticated "manual" analyzer for a parts cost of around $50, including an unusual digital pink-noise source.

Although a real-time analyzer can be more convenient to use, because of its simultaneous display of all frequency bands, a manually swept model is no less professional, and its use follows well established practice. In fact, a continuous sweep approach can be used to detect narrow peaks and dips, which would be ignored by a real-time unit with its fixed third-octave spacing.

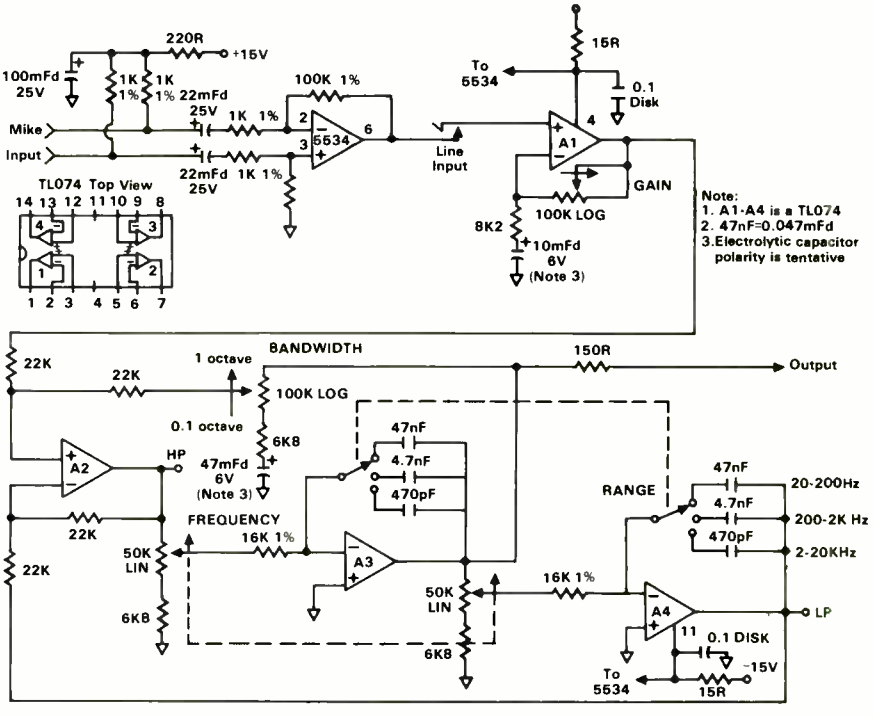
The circuit shown in Figure 1 can be built using only two ICs, and has a bandwidth that is adjustable from more than one octave, down to a tenth of an octave. As shown, the frequency range will be 20 Hz to 20 kHz in three bands, although a fourth subsonic band could easily be added to allow very low frequency analysis. A low-noise mike pre-amplifier is included that will be needed for any acoustical measuring, and a line input is also available to provide up to 20 dB of gain. Of course, you could use a mike pre-amp in your console instead, but at the expense of portability.

The selection of a suitable measuring microphone is crucial for serious room analysis, since no doubt the mike will be the weakest link in the measuring chain. First of all, it should be omnidirectional, as reflections of walls, ceilings and floors contribute greatly to the sound of that room, and must be taken into account. Second, the mike element must be a small capsule, condenser type, since the large diaphragm in a Neumann U-87 or an AKG C-414 is bigger than some of the wavelengths being measured, causing inaccuracies at high frequencies. When cost is no object, special measuring microphones are available with really tiny elements; and although their output is relatively low, the response can be made to be extremely flat. One possible exception would be the voicing of a PA system, where a typical stage mike might be a better choice. In this way, any irregularities originating in the mikes will be included in the measurement and correction process.

I chose an AKG CK-22 capsule that fits a C451 mike, because I trust the manufacturer to give me an accurate graph of its response, and because I already have the mike. No doubt other small capsule omnidirectionals from Neumann and others could be used, although the AKG C451/CK-22 can be reliably operated from as little as 12 V, which makes phantom powering much easier.

Pink noise — which contains all frequencies simultaneously - is frequently used as a signal source when measuring speakers, because its random nature minimizes the generation of standing waves. Many noise generator designs use a reverse biased semiconductor junction as the actual noise source, although this requires trying many transistors, selecting one that produces "good" noise, and then amplifying it a lot, followed by a pink

Figure 1



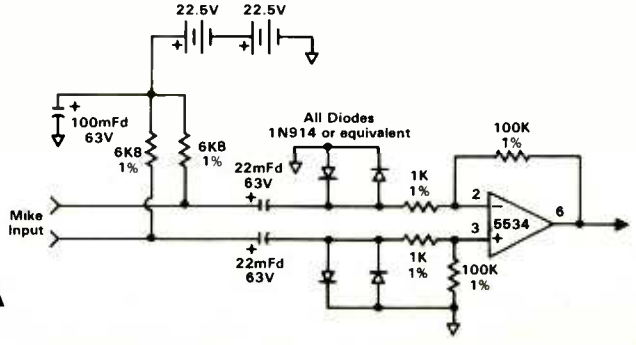


Figure 1A

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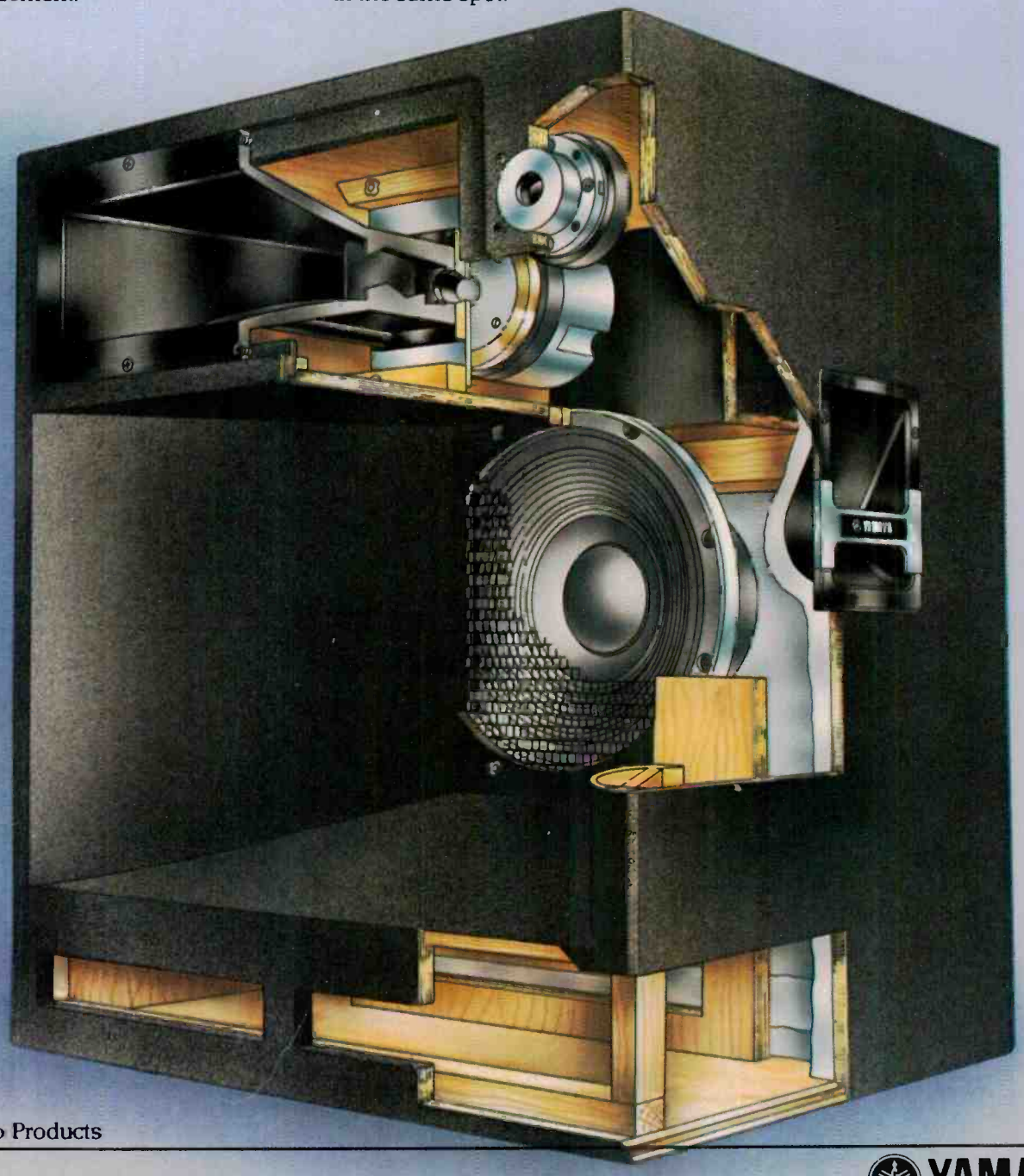

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noise filter. While the circuit of Figure 2 represents a somewhat more complicated approach, the results obtained will be much more predictable.

A high-frequency CMOS IC oscillator is used as a clock connected to a 31-stage digital shift register. Without getting into the gory details, this shift register is configured as a "pseudo-random binary sequence generator" (whew!) which, when clocked fast enough, creates extremely pure, wide bandwidth white noise[1]. White noise, however, has a rising high-end when measured on an octave basis, since every frequency is represented by the same amount of energy. (In other words, the octave from 10 kHz to 20 kHz has twice as much power as the octave from 5 kHz to 10 kHz, since twice as many cycles are being considered.) Since most audio work involves measurement by the octave or fraction of an octave, a continuous high-end rolloff of the noise source amounting to 3 dB per octave is required, to maintain the proper perspective.

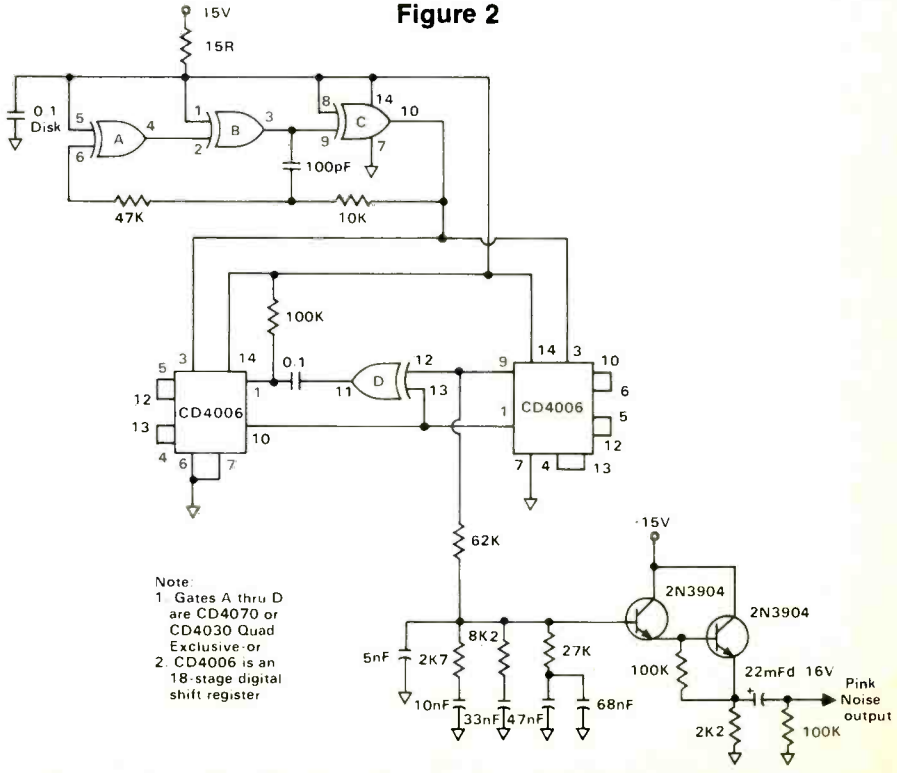
Now the only problem is that filters always come in multiples of 6 dB per octave, since this is the minimum amount of attenuation obtainable with a single resistor and capacitor. Therefore, a compound filter must be designed employing several RC networks, each contributing a little to the overall curve, as shown in Figure 2. The output of this filter prefers to "see" a relatively high impedance load; consequently, a two transistor Darlington follower is used as an output buffer, rather than a simple emitter follower. An op-amp follower could also be used, but the transistors are fully sufficient.

Returning to the analyzer, a state-variable filter is used in the bandpass mode, and can be swept in three switch selected ranges: 20 to 200 Hz; 200 Hz to 2 kHz; and 2 to 20 kHz. The state-variable design was chosen for several reasons — the tuning frequency can be easily varied while maintaining a perfectly flat response; and the Q, or bandwidth, can be controlled independently. If you notice a similarity in design to that of a parametric equalizer, you won't be surprised to find that many parametrics are based on a state-variable design. Admittedly, this type of active filter is more complicated than many that you'll see, since three op-amps are required in place of the usual one. But many useful audio gadgets can be built with such a state variable filter design, as we will be seeing later in this article.

Circuit Design

Referring to Figure 1, an NE5534 op-amp was chosen for the mike pre-amplifier stage because of its low input noise. Forty dB of gain should be sufficient for most microphones, since the pink noise level will be played through the speakers at a moderately high level. If you can't obtain 1% resistors, 5% types will probably do, but only if you aren't located near a radio transmitter, or in a metropoliton area. The 22 mFd capacitors block the phantom power from the pre-amp's input, and the 100 mFd capacitor is used as a filter to remove any traces of noise on the power supply. (Of course,

Figure 2



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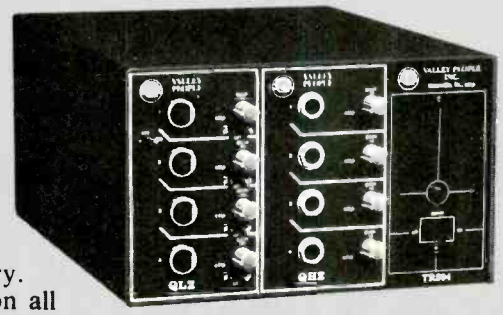

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whenever building sensitive circuits you *should* use a regulated supply to begin with.)

One important note: if you do not use a C451 microphone, you will need to provide the full 48 volts as a phantom power source. Batteries of 22.5 volts, the size of a penlight cell are available at camera and Radio Shack stores; using two of these would probably make the most sense. Also, you must increase the voltage rating of the input capacitor, the resistors for phantom supply, and add diodes to protect the op-amp from turn-on spikes — of which are shown in Figure 1A.

Once the input has been sufficiently amplified, it is sent to the state-variable filter comprising op-amps A1 through A4. As with many active filters, the input is to a summing junction, or negative feedback node. Unlike most summing junctions, however, this one is at the *plus* input of an op-amp (A2), although the feedback is still negative, since an inverting stage (A3) has been placed in its feedback loop. A3 and A4 are connected as integrators, with a capacitor as the feedback element; the tuning frequency is determined by the setting of the dual 50 kohm pot, the 16 kohm integrator resistors, and the capacitors.

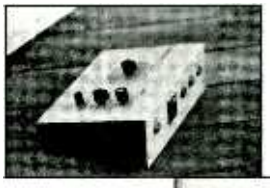
The bandwidth is varied by adjusting the amount of feedback to A2's plus input and, as you might expect, the filter's gain increases along with the reduction in feedback, just like a normal amplifier stage. Fortunately, this increase in gain is partially offset by the reduced bandwidth being measured, as was mentioned earlier. In fact, *any* noise measurement or specification is meaningless if you don't know the range of frequencies being considered.

Besides being able to easily vary the Q and tune its frequency, a state-variable filter also provides a high-pass and low-pass output, and these are available at A2 and A4 respectively. A swept notch can also be achieved simultaneously, by combining the input to the filter (A1's output) with the band-pass output in a summing amp, as shown in the THD analyzer circuit provided on page 100 of the December 1981 issue of *R-e/p*. An ambitious builder could then add the 20 dB gain stage required, and have a super test instrument to use in the studio. Now, if we just add a . . .

For all of the filter components it is important to use the closest tolerance parts you can get your hands on. One-percent metal film resistors are the best, not only because of their accuracy, but also due to their reduced sensitivity to temperature change. For this same reason, polycarbonate or Mylar capacitors are the best choice for the actual filter stages. For power supply bypass, disk capacitors are not only sufficient but preferred, since they possess less series inductance. The tubular types, though generally more stable, are constructed of long strips of foil which, when rolled up for compactness, become inductive, thereby reducing their effectiveness at high frequencies.

Construction and Calibration

In the prototype, I used 2% metal film resistors because I had them around, and 2% and 5% capacitors because I could get them, though at a price. Capacitor tolerance *is* important if you plan to label only one band of frequencies on the panel, as shown in the accompanying photograph.



Completed Manual Spectrum Analyzer

Front Panel Detail

Otherwise, 1 kHz may not occur at the same dial setting as 100 Hz or 10 kHz. If you can't obtain at least 5% types, you can always "tweak" a capacitor to a higher value by connecting a suitable smaller cap in parallel.

Earlier, I mentioned the possibility of adding a fourth band for analyzing very low frequency information. Besides increasing the tuning caps to 0.47 mFd, you will also need to replace the 22 mFd input capacitors with 220 mFd types. Personally, I'd be pretty surprised if the low-end response of your microphone came even close to 2 Hz. Perhaps bypassing the mike's output transformer would help, although I don't know for sure, since I've never tried it. The 10 mFd capacitor in the line-in gain stage must also be increased, as well as the 47 mFd capacitor with the bandwidth control. Needless to say, that's a heap of big capacitors to squeeze into a compact package!

Generally, I dislike having to repeat the same standard construction details in each of these articles, regarding points to watch out for. For example: be sure to use bypass capacitors at each IC; always use a regulated power supply; don't use acid core solder with a blowtorch — you know, things like that. Well, in the not too distant future *R-e/p* will present an article discussing construction practices, setting up a lab, making nice panels, etc. In the meantime, however, none of the projects presented here are particularly critical in any way, although you would be wise not to have the input wires bundled in the same harness with the AC power cord!

Once the unit has been successfully completed, you will need to fabricate some type of frequency dial. One possibility is to make it from heavy paper stock, marking the exact third-octave frequency points with a pen. I was very concerned with accuracy during calibration of the prototype unit, since my test oscillator is spec'd at only 5% frequency tolerance. By borrowing a frequency counter from a friend, I was able to create a chart showing the exact dial settings required to obtain each standard third-octave frequency from my oscillator. From there, it was a simple matter to locate the proper places to mark the paper dial, which in this case became the basis for a silk screen artwork. Note that a large diameter dial will increase resolution and ease adjustment.

You can begin by connecting your oscillator to the analyzer's input, while monitoring its output with a meter. With the oscillator set for a frequency of 1 kHz, sweep the analyzer dial until the meter indicates a maximum reading. To really pinpoint the exact location, it helps to have the bandwidth control set for the narrowest response (1/10th octave). Repeat the procedure until you have marked all of the standard third-octve frequencies from 200 Hz to 2 kHz. These same marks should serve to identify the proper points on the other ranges as well.

Besides calibrating the frequency dial, you should also mark the bandwidth control at points corresponding to one octave, one-half octave, one-third, one-sixth, and one-tenth octave. This will make you compatible with just about *anything*, though come to think of it, some equalizers are based on two-thirds of an octave. Of course, that's no problem either, and a complete listing of calibration points is shown in Table 1, related to a 1 kHz center frequency.

Table 1
Bandwidth Calibration Points for a 1 kHz Center Frequency

Bandwidth	Lower 3 dB Point	Upper 3 dB Point
1 Octave	707 Hz	1.414 kHz
2/3 Octave	794 Hz	1.260 kHz
1/2 Octave	841 Hz	1.189 Hz
1/3 Octave	891 Hz	1.122 kHz
1/6 Octave	944 Hz	1.059 kHz
1/10 Octave	966 Hz	1.035 kHz

To find these points, once again tune both the oscillator and the analyzer to exactly 1 kHz, only this time detune the *oscillator* in each direction until the response drops 3 dB from the maximum value. There's no escaping that this is a tedious trial and error process, although

you can console yourself with the knowledge that you'll only have to do it once.

Due to the state-variable design, the filter will be perfectly flat at all frequencies up to 12 or 13 kHz, where you may observe a slight rise. This is caused by phase shift within the op-amps, but shouldn't exceed a dB or so. Just be sure to take this into account — along with the microphone's calibration chart — when using the analyzer. If you check its flatness with an oscillator, first verify the oscillator's own frequency response, as many are not all that flat. (Here is where the function generator excels.) Also, run the pink noise source through the analyzer once on a third-octave basis, to ensure that its response is flat as well. Once you have accounted for all possible measurement errors, you should make up a graph showing the total error versus frequency, which you could then paste to the bottom of the unit.

Measurement Technique

If you are measuring a control room, set the analyzer's bandwidth control to the appropriate setting, dependent on the type of EQ you'll be using, and place the measuring microphone where your head would be when mixing. Feed the pink noise source at a moderately high level to one speaker only and, with the frequency set to 1 kHz, adjust the gain control for a reading of 0 dB on an AC voltmeter. Ensure that the meter measures flat over the full range as well, or else you could use the VU meters in your console if they are up to it, feeding the analyzer into a line input. Either way, using graph paper mark a dot at the measured response level for each of the frequencies being considered — applying corrections where required. Then connect the dots to make viewing easier. Repeat the procedure for the other speaker, and again with both of them powered.

At each frequency, you must watch the meter for several seconds, especially at the low end, and mentally average the pointer indications. After all, pink noise is, by definition, inconsistant; occasional large excursions are unimportant and can be ignored — it's the average that's important here.

Don't even think about using EQ until you have achieved the best response using the level controls on the monitor's crossover units. The less EQ that you apply to your monitors — especially the third-octave kind — the better. In fact, a recent trend is to use parametric equalization instead of graphic EQ, particularly when broad curves are indicated, since applying a lot of boost with a graphic can cause ringing. Similarly, if overall high- or low-end correction is required, you may do better with shelving-type controls, where you could customize the turnover frequencies. And as long as we're on the subject of equalizers, let's take a look at some of the circuits that are commonly used.

Equalizer Designs

Figure 3 shows the schematic for one band of a parametric equalizer, and I'm sure that you'll immediately recognize the state-variable filter. The input op-amp can be either a TL0 Series for line-level use or, if you want to connect an instrument directly, use a 5534 with 30 dB of gain. Simply change the 100 kohm resistor between the minus input and ground to 3.3 kohm, though you might want to add a 10 mFd capacitor in series with the 3.3 kohm to minimize DC offsets. This is why all of the controls have similar capacitors to ground; otherwise you would get a scratching sound as you turned them. One problem does arise with these capacitors regarding polarity, since there is no way to determine in advance which way the op-amp offsets will go. I would suggest building the device as shown, but be prepared to switch any caps that end up reverse biased. Alternately, you could jumper across them initially, and then measure the DC voltage at the relevant op-amps for an indication of which way to put them in. Tantalum capacitors don't seem to mind a few millivolts being placed across them backwards, but large values are rare and quite expensive.

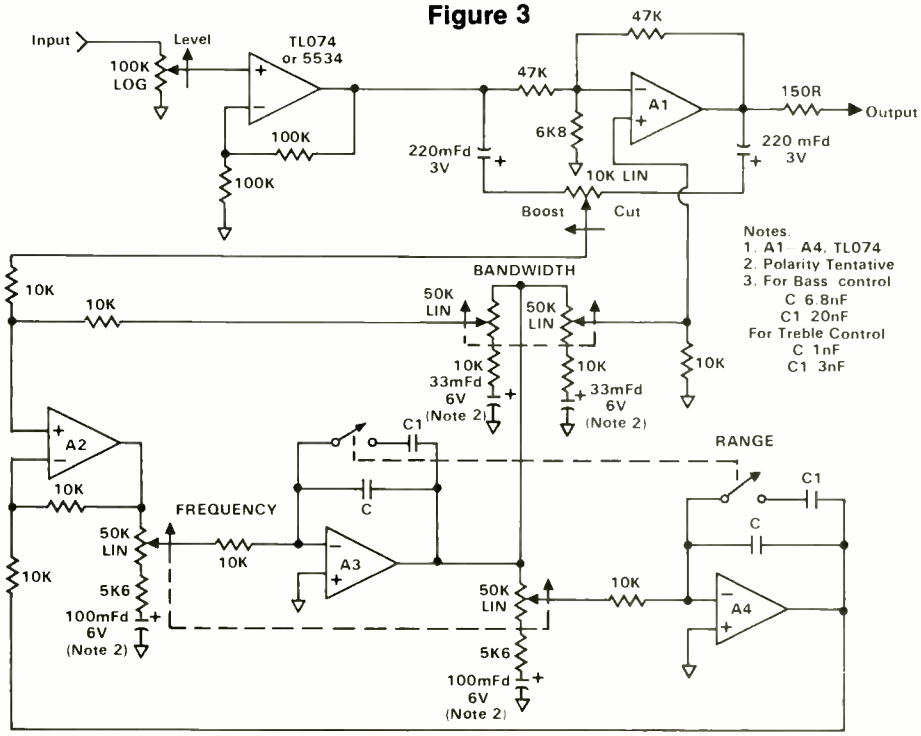
Following the input stage is an inverting op-amp, and it is here that the actual EQ takes place. The boost/cut pot is connected from input to output of this op-amp and, since it is inverting, a null point is created at the center of the pot. One of the things that makes this circuit so good is that the entire filter is effectively out of the picture when the control is in the center position, thus eliminating the need for a separate in/out switch for each band. As the pot is turned toward boost, some of the input signal is sent to the filter, which eventually feeds the plus input of this op-amp. The 6.8 kohm resistor to ground allows nearly 20 dB of gain, which in this case translates to available boost. Cut is applied in a similar manner, only now the signal is inverted causing the filter to *oppose* the original input.

If you build the circuit exactly as shown, you will notice an area of reduced sensitivity near the middle of the control. While this makes the boost and cut slightly touchy at the extremes, it allows small amounts to be applied easily, and also reduces errors caused if the knob is not precisely centered. None the less, some people may prefer a more uniform dB spacing, and this can be readily accomplished by adding a 4.7 kohm resistor from the boost/cut wiper to ground. I should also mention that this particular circuit arrangement produces the so-called "reciprocal" type parametric, which provides the same bandwidth for both cut and boost.

Unlike that provided on the spectrum analyzer, the bandwidth control should be a dual device, with one section used to attenuate the output of the filter. This arrangement exactly compensates for the increase in gain as the bandwidth is reduced. Otherwise, the amount of boost or cut would diminish as the bandwidth is increased. But, then again, some people might consider this interaction to be beneficial, since as the bandwidth is widened more frequencies are affected. I happen to know of at least one commercial product using this "constant loudness" approach as a feature.

Often, when a track needs equalizing,

Figure 3



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only one or two frequency ranges will be employed. Therefore, when I built four parametrics for my studio, each was configured as a dual two-band unit with a switch to connect them in series when required. This allows me the use of up to eight separate equalizers when I need them.

As many stages can be cascaded as you like, although always provide an input buffer before the first one, and use a 150 ohm resistor between the last stage and the output jack. If desired, an in/out switch can be effected for each band by opening the connection between the wiper of the boost/cut control and everything else. Of course, the entire equalizer can be bypassed as well, using a SPDT switch. To enhance the two-band format, I chose a dual-range approach, as shown in the schematic. This required overlaying *two* silkscreen artworks — one color for each range — but it came out great, and was well worth the trouble. The bass control spans the range from 40 Hz to 2.4 kHz, and the treble from 270 Hz to 16 kHz.

Electronic Crossover

Another control room goody can also be built with a state-variable filter, and that is an electronic crossover for biamping, or for that subwoofer you've always wanted. Taking advantage of the simultaneous high-pass and low-pass outputs, we get precisely complimentary curves that fall at a rate of 12 dB per octave. There has been a lot of talk lately about the advantages of using 18 dB per octave slopes instead, however, so I contacted Altec, JBL, and Audiotechniques (Big Reds) for their recommendations. JBL was emphatic that 12 dB per octave crossovers be used with their components, and Audiotechniques also felt that this would create less ringing and phase shift than the 18 dB types. On the other hand, Bob Davis from Altec explained that his company offers crossovers in many different formats, and as such wasn't in a position to endorse any particular design. Frankly, I'd just as soon stay out of this controversy altogether, and simply present the facts.

The major concern of the 18 dB per octave proponents is that phase shift in the crossover not interfere with the proper acoustical combining of the various drivers in the system. Phase coherence in the state-variable design can indeed be maintained, however, by simply adding op-amp A4 and the additional 27 kohm resistors, as shown in Figure 4. This creates an alternate low-frequency output that will accurately combine with the highs, although the roll-off is now reduced to 6 dB per octave. I recommend that you try it both ways, choosing the one that sounds best to you. (Sure that's a cop-out, but I *told* you I was going to stay out of it.)

You can select any crossover frequency in the audio range by changing either the integrating capacitors or resistors, or both. Although the actual formula is fairly complex, doubling the capacitance will cut the frequency by a half, as will doubling the resistance. Halving both the capacitance and the resistance will quadruple the frequency, and so on. The capacitors should be the same value — use 2% if possible — and the resistor ratio should be maintained at precisely two to one, as shown in the schematic. Keep the caps larger than 1,000 pFd to minimize the contribution of wiring capacitance, and keep the resistors larger than, say, 3.3 kohm, so that the op-amps don't have to work too hard. The unusual resistor values shown for 800 Hz are what came out of the calculator, and are shown for reference only. In practice, you would use two resistors in series to come as close as possible to the design center. This can be achieved either with one-percent resistors, or by using a trimpot in series with a resistor for precise tunability.

Just to be sure we don't lose anyone, here are a few examples using this "minimum math" approach. For a subwoofer crossover at 80 Hz — or ten times lower than shown — simply multiply the capacitor values by 10, which would yield 22 nFd (0.022 mFd). Going to 8 kHz is equally easy, but we don't want to let the capacitors get that small. Therefore, leave them at 2.2 nFd, and drop the resistors by a factor of 10 instead, to 6.39 kohm and 12.78 kohm. Okay fine, but what if you want 1,100 Hz? Simply create your own multiplier by dividing 800 by the new frequency:

$$\frac{(800 \times 2.2\ \text{nFd})}{\text{New Frequency}} = \text{New Capacitor}$$

$$\frac{800}{1100} = 0.727$$

$$0.727 \times 2.2 = 1.6$$

As it turns out, 1.6 nFd is a standard value; unfortunately it is not that common so, instead, let's try it with the resistors:

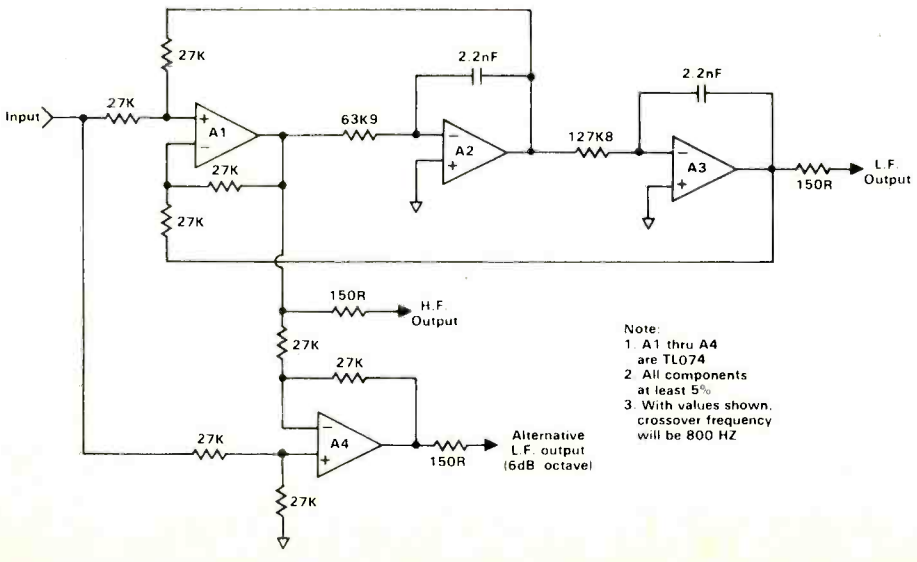
$$0.727 \text{ X } 63.9 = 46.47$$

So far so good, since that's very close to 47 kohm, a standard value. Now double it to find the other resistor, and you're done (93 kohm). The same formula works for below 800 Hz, only now the multiplier will be greater than one. Rest assured, this is *not* how the big boys do it, though this method is no less accurate.

There are several ways to make an octave or third-octave graphic equalizer, and my favorite is shown generically in Figure 5. In this circuit, only one op-amp is used to do the equalizing, regardless of how many bands are involved. Notice that instead of using inductors, a gyrator circuit is used at each band though, like the parametric, no noise or distortion is ever contributed when the pot is centered. (And it's not too bad when you use it either!) Other designs consist of one op-amp filter section for each band, with all of the outputs combined in a mixer. This approach limits the maximum signal-to-noise ratio you can

. . . *continued overleaf* —

Figure 4



References

1. "Build a Low Cost White/Pink Noise Generator," by Bill Eppler and John Pfeifer; *Popular Electronics, February, 1980, page 67.*
2. "The Active Filter Cookbook," by Don Lancaster; Howard W. Sams #21168 (Indianapolis, 46268).
3. "National Semiconductor Linear Applications Handbook"
4. "National Semiconductor Audio/Radio Handbook" (Both from: National Semiconductor, 2900 Semiconductor Drive, Santa Clara, CA 95051.)
5. "Filters — Theory and Practice," by Ethan Winer; *R-e/p, August 1981, page 70.*

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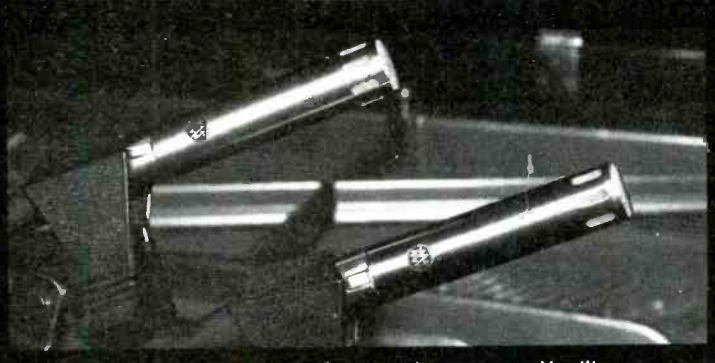

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achieve, especially when many bands are used.

Due to the large number of possible combinations of frequencies and bandwidths, let me apologize for not including a comprehensive listing of component values. However, I have included a complete set of formulas for the ambitious masochists among our readers. □□□

Filter Design for Graphic Equalizers

Before attempting any calculations in the design of these filter sections, some basic decisions will need to be made. First of all, the center frequencies must be choosen and, of course, the number of bands. This will be relatively easy, since the Standards people have already decided this for you; for example, (/ISO octave and third-octave spacings). Less obvious, however, is how many dB boost or cut should be allowed, and the choice of filter Q. These are questions only you can answer, and I'm afraid even the manufacturers don't always agree on such things; I have seen octave graphics with Q's ranging anywhere between 1.4 and 3 or more.

The National Semiconductor Audio/Radio Handbook[1] recommends a Q of 1.7 for an octave equalizer described in its pages, although obviously it must be higher for a third-octave unit. You probably wouldn't want the Q to be much more than 5 or 6, however, to keep phase shift and "ringing" to a minimum. The trade-offs in the other direction (lower Q's) are, of course, less control — since the filters will interact — and bumps in the response when adjacent bands are boosted.

What makes the determination of filter Q even more elusive is the fact that it is constantly changing; the bandwidth of any equalizer will vary, depending on the setting of its boost/cut control. Furthermore, there is no way of knowing ahead of time how much boost or cut will be required, which dictates the optimum Q. This is one reason why a three- or four-band parametric may be a better choice, at least when voicing a permanent installation, such as a control room.

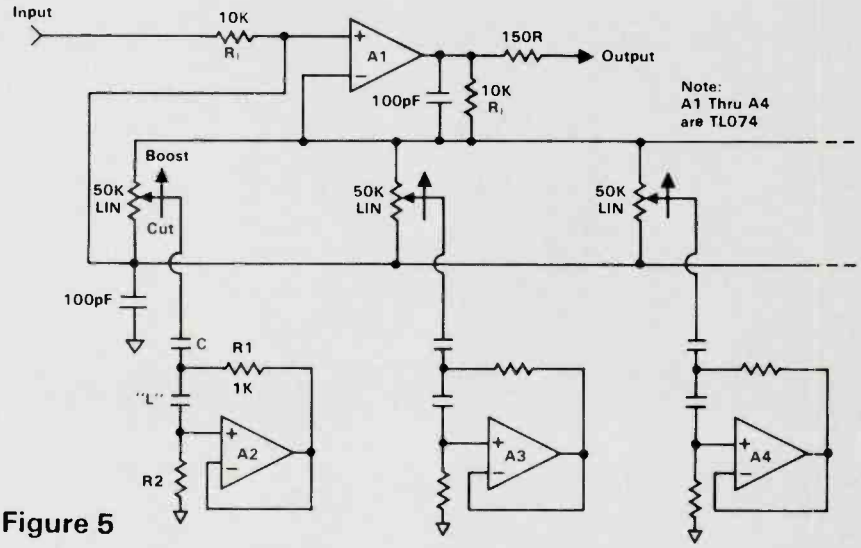
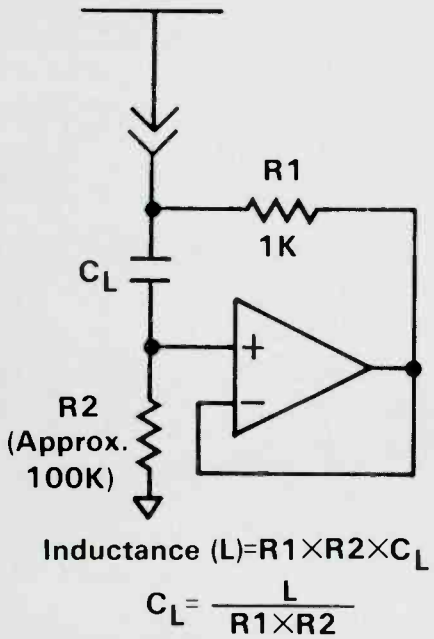


Figure 5

You will also need to choose some starting points regarding component selection. Referring to Figure 5, most of the units that I have come across use a 1 kohm resistor for R1. With 10 kohm input and feedback resistors, a range of plus and minus 20 dB will be obtained. These resistors are referred to as R_E in the formulas presented below.

Figure 5A shows the simulated inductor circuit alone, with the formula to find the capacitor's value. R2 may be varied about the nominal 100 kohm, to fine tune the filter to the correct frequency. I don't mean using a trimpot, but since capacitors only come in certain values, it is easier to juggle the resistor value up or down a notch or two. Actually, going up in value is generally preferred, but so long as you keep R2 greater than 80 kohm, there shouldn't be any problems.

Figure 5A



Inductance (L)=R1×R2×C_L

$$C_L = \frac{L}{R1 \times R2}$$

Maximum boost/cut = $20\log[(R_E+R1)/R1]$

$C = 1/2\pi FR1Q$ also, $C = 1/(2\pi F)^2L$

$L = R1Q/2\pi F$ also, $L = 1/(2\pi F)^2C$

$Q = 2\pi FL/R1$

$F = 1/2\pi\sqrt{LC}$

where F = center frequency

R_E = 10 kohm

R1 = 1 kohm

π = 3.14159

Again, just to be sure, let's do a sample filter section. The equalizer will be a 10-band octave model, and we'll use the Q of 1.7 that National recommends. To find all of the component values for the 1 kHz filter section, the following procedure will take place:

$C = 1/(2\times\pi\times1000\times1000\times1.7)$

$= 9.4\times10^{-8} = 0.094$ mFd

You won't find a 0.094 mFd capacitor at any store, but a 0.1 mFd isn't too far away, so let's use that instead. Now we need to find L (the inductor):

$L = 1/[(2\times\pi\times1000)^2\times0.1\times10^{-6}]$

$= 0.25$ Henry

That's not a standard value either, but since we're synthesizing the inductors, it doesn't matter. Now on to the "simulated" inductor.

$C_L = 0.25/(1000\times100000)$

$= 2.5$ nFd

The closest standard value is 2.7 nFd (0.0027 mFd), so let's go with that for the moment. Since the cap is larger than we needed, we'll have to make R2 **smaller** to compensate. The next step down is 91 kohm — well within the lower limit — so let's see how that combination does. Note that we're using the formula in Figure 5A this time:

$L = 1000\times91000\times2.7\times10^{-9}$

$= 0.246$ Henry

Voila! Very close to the required 0.25; see, I told you we wouldn't be needing any trimmers. Now that we've found the proper values, let's double-check our work by calculating the frequency and Q:

$F = 1/[2\times\pi\times\sqrt{(0.246\times0.1\times10^{-6})}]$

$= 1.015$ kHz

I would call an error of 1.5% quite acceptable, especially for an octave graphic; just don't spoil everything by using 20% capacitors!

$Q = (2\times\pi\times1015\times0.246)/1000$

$= 1.57$

Well, not exactly 1.7, but really not too bad either. If you're not willing to compromise, you'll have to go all the way back to when we chose 0.1 mFd as the value for C. If instead we had gone **down** one step to a 0.082 mFd, the inductance required would have been 0.3 Henry, which then makes the Q about 1.9 — not any closer. In this case, if you still won't budge, you will need to use two capacitors in parallel, perhaps a pair of 0.047 mFd. Hey, that's **exactly** 0.094. Why didn't I think of that before? Oh well, my fingertips are getting sore from all this button pushing, so I'll turn it over to you. Good luck.

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FILM SOUND EDITING TECHNIQUES

by Steve Barnett

The first installment of this article, published in the August issue, outlined the basic responsibilities of a motion-picture sound editor, along with a detailed description of their first area of concern: the dialogue and Foley tracks. This concluding part details the remaining tasks in 'preparing for the mix' — backgrounds and sound effects.

— The Participants —

The editing team of **Stephen Flick** and **Richard Anderson,** both of whom are graduates of the University of Southern California, has been involved with the sound for such films as *Star Trek — The Motion Picture, The Final Countdown, The Hand,* and *Raiders of the Lost Ark.*

Richard Portman has mixed the sound for numerous motion pictures, including *The Deerhunter, Nashville,* and *Coal Miner's Daughter.*

Having recently served as a sound editor at the post-production house of Neiman-Tillar Associates, where the sound was handled for *Roots, Centennial, Days of Heaven,* and *The National Geographic Specials,* **Elodie Keene** is currently working as a free-lance editor.

Sound editing is not necessarily a linear job, for although dialogue is really the first concern of the sound editor, circumstances — rather than an orderly progression — will often dictate how the task is approached. In the case of a sound editing team with assignments classified according to the nature of the tasks, rather than by the reel, work will proceed concurrently, with all the different areas — dialogue, A.D.R., Foley, and sound effects — keeping in close contact with one another. If all the sounds of a reel are assigned to one editor, the job may not be scheduled so much based on a Point 'A' to Point 'B' logic, but rather around the availability of various personnel, artists, and equipment.

In any case, the sound editor is required to keep in mind the *whole work,* even though what he is doing is proceeding out of sequence. Whether the final mix will be in six-channel stereo, four-channel stereo, or mono, the finished product will be greater as a whole than the sum of its parts. Therefore, it is critical that the editor be aware of how all the individual sounds of a particular reel will interact with each other, and the rest of the soundtrack.

To facilitate this interaction, all the reels of sound for a single reel of picture are carefully designed for ease of mixing, with like sounds put on the same reels or tracks. Also, similar individual tracks are grouped together during the pre-dub process, so that when the final mix is made, a specific sound can be taken down without altering the dialogue or background levels. Even with all the imagination a sound editor can muster, when all the reels of sound are run and heard together for the first time, adjustments will have to be made.

With the dialogue and basic background or presence tracks

. . . *continued overleaf* —

Recording Sound Effects in the field: Engineer Vince Zabaly using a Nagra IV portable with a Sennheiser 815 MK Shotgun Microphone.

Photography by Gary Kenny

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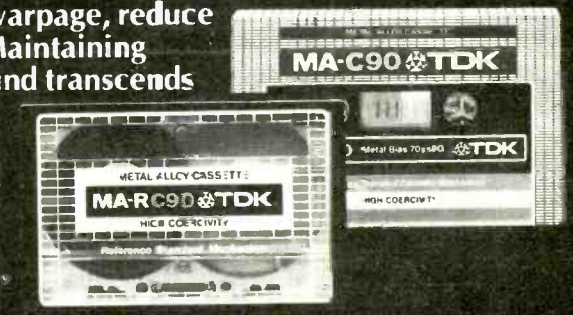

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completed, they are grouped with one another. There may be three to four tracks of production dialogue, two or three tracks of looped or replaced dialogue, one to two tracks of production presence to fill in behind the replaced dialogue, and three to five reels of Foley work to replace the movement sounds which, like the location ambience, was lost in the looping process. All of these tracks were just to get acceptably realistic and understandable dialogue. The mood of a film comes from the careful building of sound effects and background reels. These tracks are designed with similar priorities in mind.

The Pre-Dub Session

"Ultimately," says Richard Anderson, "you're going to mix all the effects down to between four and six pre-dubs [for the final mix], so at one point you have to decide to combine this reel, and this reel, and that reel."

Anderson

A typical pre-dub session on a big effects film, such as *Star Trek — The Motion Picture*, or *Raiders of the Lost Ark*, will involve all of the mixer positions on the console (up to 36 tracks), and utilize all the available 35 millimeter dummies in the machine room. This pre-dub or pre-mix session may yield only four of the 36 channels that will be blended in the final mix. It can be seen then, how a single reel picture can easily have 50 to 80 corresponding reels of sound.

In designing and building these reels, the tracks are layed out by pre-dub session.

"We call them 'A,' 'B,' and 'C' effects pre-dubs," Anderson explains. "We put all the major stuff — the cars, the airplanes, and the explosions; the 'primo' sounds — on to the 'A' reels. Then we put all the smaller stuff, like the door opens and closes, the button clicks and all, on the 'B' reels. Then we put all the backgrounds — the crickets, the rain, and wind, and traffic — on the 'C' reels."

With the pre-dubs designed in this fashion, the mixer working on the final mix can now increase the volume of a car door slam, for instance, without affecting the level of the rain or the volume of the car's engine as it revs. Grouping in this way provides options, and avoids locking the mixer in to a particular way of working.

On these reels, as on the Foley reels and dialogue tracks, cue sheets are kept by the sound editors that indicate at precisely what footage of a reel a given sound occurs. These cue sheets are then employed by the mixer so that he knows what sound is on a given reel, and when to expect it. Cue sheets are also useful to the sound editor, since he can lay out his reels in pencil before locking them in on mag, and then having to rearrange them if they don't work.

In building these reels, says Anderson, "the first thing we'll do is go through and make a list of all the sound effects we're going to need."

The first decisions on what specific effects are needed are made under Flick and Anderson's "See a dog, Hear a dog" axiom.

Flick

"If you see something on the screen," explains sound editor Stephen Flick, "then it's got to be on the soundtrack. If you don't see something [but you hear it], then that's extra; something the sound editor or the producer or the director feels should be in, as opposed to things that are obvious."

"Now," continues Anderson, "we have to figure out where to get all these effects. There are basically three main things you can do: first, you can go out and get a recorder and make them yourself; secondly, you can get them from a library; or third, you can get them from the production sound recordings."

"There are production wild tracks that come in with the dialogue tracks on a show," adds Flick. "But the number of these depends on the production mixer and the schedule he was given."

"The documentaries that I've worked on," Elodie Keene recalls, "almost invariably supplied us with production sound effects. They were a whole lot more careful, since the subjects of the films were often so unusual that the actual sounds could not be found in existing libraries, or even created with any accuracy. On a feature, we try to use as much production sound as possible, because, after all, that is the *real* sound. The effects, however, are seldom provided.

Keene

"In features," says Anderson, "it's not unusual for the production company to be really rushed just to make the movie, let alone to go around and record a lot of sound effects."

"This presents a number of problems," continues Flick, "as often there are things on location the sound of which we just cannot recreate."

Portman

"Lets say they shoot a picture in which they have a race car that costs $400,000 to build," offers Richard Portman. "Well, it makes certain sounds and if they don't record those sounds and bring them back to the editors, you end up with a picture with a chase sequence featuring this high powered vehicle, and nowhere in town can you find a sound that suggests or even gives you an idea of the

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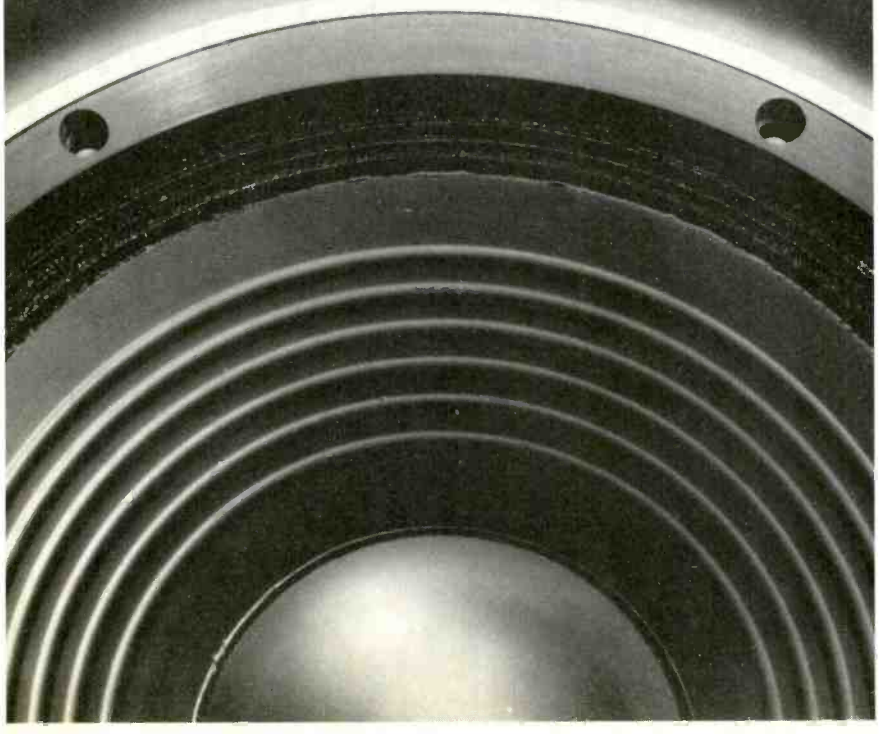

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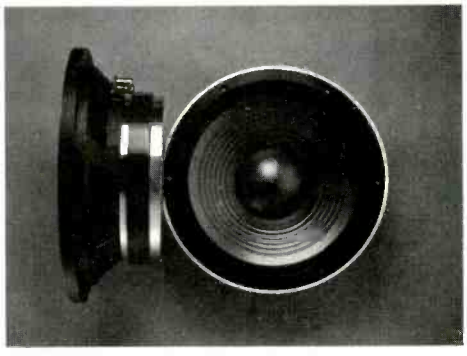

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power of that car. Now the producer and the director have got something on film that they love, but they don't get the same excitement from the sound, because it's not the *correct* sound.

Sources for Sound Effects

For reasons that are now buried in the years-old Hollywood hierarchy, "We'll get it later," and "Fix it in the mix," are standard operating procedures.

"I was working on a compilation film," offers Keene, "and there was a huge shot of a crowd of Japanese soldiers screaming 'Banzai.' So the director turns to me and says, 'I'm sure you have that sound in your library, don't you?' Well, we didn't, and it's difficult to make that sort of effect. So what you do is start to make the rounds to see if somebody's got something that will work."

On a seemingly simpler level, Flick continues, "take a crowd of 150 people shouting and cheering. Now you might think that maybe we have the effects in the library, and perhaps we do."

"But," counters Anderson, "that crowd may be in an area that is acoustically unique, and the scene may require that we use some of the production track. Consequently, when we fade-in the library crowd sounds to fill in between production recording, it will have a totally different sound to it, because it was recorded in a totally different location, maybe 20 years before."

Another problem with library effects, Flick continues, "is that they might be so old with so much hiss and generation loss that for modern sound mixes they are unacceptable."

"Most of the material was recorded a long time ago," explains Anderson, "sometimes on 17½ millimeter or, even worse, on optical tracks. By the time they get to the various libraries, they've gone through several generations, and are degraded from a source that was questionable in the first place."

"Many of the studios quit developing their libraries about 15 or 20 years ago," Flick recalls. "That's when they stopped having libraries that were really prepared and organized."

The practice was to take the new effects recorded for each of the studio's movies, and place them in the sound effects library categorized and cross indexed for use in later features.

"Nowadays," continues Flick, "you can go into a major studio's library, and still find all the sounds from every feature from the past 20 years. But they're just thrown into a box by film. You can go to the box for, say, 'Islands in the Stream,' and find all kinds of interesting effects, but they're not categorized."

Flick and Anderson once retrieved the sound of a fishing reel from the boxes of effects marked "Islands in the Stream," but the studio had not yet catalogued any of the sounds from this or any other late feature film.

"Their view is that they already have one," says Anderson. "They have a fishing reel in the library that was recorded in 1938. It's already been catalogued and the editors don't need to look for it. So that's the effect they usually use."

There are other sources aside from the studios, but they can be just as nefarious.

"I'll never understand why the BBC sound effects library comes in disk form," complains Anderson, "and then the instruction manual recommends that for use in plays or films, you transfer it on to ¼-inch tape. It makes no sense, because even with a virgin

. . . *continued overleaf* —

The Potential Role of a Film Sound Director

"I got into editing film sound," explains re-recording mixer Richard Portman, "because it's impossible to separate sound editing from sound re-recording. You have to know how to do editing to become a good mixer of any type. Musicians and record people understand this very well. They understand about masking; that's why when they orchestrate, they write voices in different ranges so that you can hear one against the other."

Sound editing, feels Portman, is not unlike music orchestration: constructing the tracks is the same as writing the different parts of the score. All will be blended together later on the re-recording stage, or in the concert hall, but the original parts have to be well thought out and designed before that final step can be accomplished.

It was Portman's interest in sound editing, and the problems he encountered in that craft, that led him further back in the film production process to production-sound recording. There he discovered the sources of many of the problems he had encountered on the re-recording stage. Most were due to the second-class status in which sound is seen during production.

Although upcoming technological advances will affect the mechanics of sound recording and editing, Portman feels that it is the people and their ideas that will have the greatest impact on motion-picture sound. The new tools and all that they have to offer can only be used to their best advantage if, and only if, sound is taken more seriously right from the moment a film is given a production date.

Portman suggests that one reason it is difficult to do the best sound job possible on a film is because no one individual is given charge of that area for the entire production and post-production period. The film goes from the production recording team to the sound editors to the re-recording mixers, and at every step of the process each unit must become familiar with the story and the audio material available. No single person, from the start of production to the film's release, is responsible for the sound needs of the motion picture.

It is Portman's view that there should be such a position in motion-picture production. The Director of Photography on a film is not merely in charge of the lights and camera. He is responsible for the entire visual image of the film. Along with the director, he supervises the art direction; set design, construction, and dressing; costume design; and make-up.

Portman's proposed 'Director of Sound' would have the responsibility and commensurate authority for the aural image of a film. He would follow the production from location scouting, through the first day of shooting, the editing, the final mix, and release; just as the cinematographer takes the film from the first stages of pre-production through the shooting, the lab's processing and timing (color and light adjustments), to the final answer print. Among the results, Portman offers, would be a cleaner sound track to support the story, a less expensive sound budget, and the opportunity for greater artistic expression in the audio area of motion pictures. The end product is a better film.

Portman lists a number of areas of increased efficiency:

"If sound personnel were taken along and their end result considered when scouting locations," he says, "they would avoid a number of problems. Most of the time the first time the sound mixer comes on the job is during day #1 on location. They don't even get the opportunity to test the cameras, to run the equipment, and find out if there are things that are wrong, like a noisy camera or a noisy location. There's no control.

"In the studio days, there used to be more cooperation. There was a studio sound director; someone you could go to if there was a problem in the dailies. As a result you could stop the trouble in the beginning, because there's no way to correct a production mistake except in production."

A Marriage of Skills

"Ideally, though not necessarily," says Portman, "the sound supervisor should be the production sound mixer, and the man that does the re-recording.

"Doing dialogue pre-dubs can take from four to six days on the re-recording stage, usually just fixing problems, smoothing out dialogue tracks that are noisy, or what have you. Whereas if you had been there as the mixer, or supervisor, with a voice in pre-production planning, you would have been able to solve your problems right on the spot where they happen.

. . . *continued overleaf* —

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— continued . . .

The Potential Role of a Film Sound Director

"For the best sound possible, the sound editor should be on the production right from the start to ensure that he or she gets the right sounds on tape.

"Currently, when they shoot motion pictures, they take the picture editor right to the location, and he'll sync the dailies, start to cut the picture on location, and advise the director as to coverage. But they *never* think to take the sound editor on location. He would have a second-unit sound team with him, or even just advise the production recordist as to what effects he needs for a given scene — be they 500 horses, a high-powered race car, or what have you."

Since most production companies feel that they cannot afford to take the time to record clean sound effects, sound editors must often pull effects from the production tracks.

"They'll film their 500 horses," says Portman, "and, because everything costs so much, they just keep going. Then when the sound editor gets the track, he says, 'Wow, these are great horses.' But the director's talking all they way through the track, there are airplanes flying over, and assistant directors blowing whistles. So it's unusable.

"Currently, a lot of production sound recordists will record location-sound effects, but the sound editor has not even been hired yet. The daily tracks then go to the transfer department where they only transfer the dialogue tracks from the printed takes. And, since the sound editor comes on often after the picture is cut — maybe nine months after production — he has no idea that these effects exist, so he has to go looking through old, often inferior libraries to get what he needs.

"Sound editing is far more complicated than most people realize, and it takes a lot longer to do a good job when there is little cooperation at the top.

"During production, the sound editor can also be building the production-sound library. As filming proceeds, the picture editor is getting his dailies synced up, catalogued and stored, and he is putting together a rough cut of the picture. The sound editor should be doing the *same* thing, so that when a sound is required, he can go and get it out of the original, brand new, clean sound track. Otherwise, he has to dig through some library for an effect that has been used so many times, and is of such poor quality that it has very little definition."

The cumulative effect of such sounds, if they cannot be fixed in the mix, says Portman, is listener fatigue similar to that caused by a poor audio pressing.

"You can only fix in the mix to a certain degree," he offers, "and it is very, very difficult. There is the stage time, the pressure, and the end result is compromise."

A second unit sound team would be helpful in this regard to record sound effects in the original acoustic environment immediately after the rest of the production company has moved on. With such a team involved from the start, not only would there be a better finished sound track, but the sound budget of the picture could be drastically reduced, with the savings of time in post-production sound effects recording and editing, and in the re-recording stage.

Though Portman is suggesting this method of approaching motion-picture sound to a number of production companies around Hollywood, he is not alone in his view of the 'aural image' of a motion picture. In Britain, it is common practice to have the sound editor on the set during filming, and the Northern California group of film makers is exceedingly concerned with motion picture sound. Notably, Lucasfilms, Ltd. in Marin County, California, enjoys the services of sound designer Ben Burtt in a company level position. Burtt was responsible for the sound design of the first *Star Wars* installment, as well as the sequel, *The Empire Strikes Back*. He is also credited as sound designer on *Raiders of the Lost Ark*, and personally cut the sound for key sections of that film.

Additionally, Francis Ford Coppolla's Zoetrope Studios regularly uses Walter Murch in a number of capacities, including sound. Murch was sound mixer on *The Rain People*, sound supervisor on *The Godfather*, picture editor and sound mixer on *The Conversation*, and sound mixer on *The Godfather, Part II*. His most notable credits are for sound montage and design on *Apocalypse Now*, for which he also served as a picture editor, and as the chief re-recording mixer. His crossing over the normally clear cut boundaries that surround the motion picture crafts served well the production, and Coppola's artistic vision. It can be done.

To make such a practice industry-wide, however, will take some persuasion, both in labor and management circles. However, with audience awareness of sound on the increase, and first-class audio one of the few draws left to the movie theater over home video, it is to the industry's benefit to listen.

Murch was recently interviewed in *The Los Angeles Times*, where he said: "Sound can say, 'This is how I want you to take what you're going to experience,' and do it quicker than anything else. Like a picture frame which enhances a painting by echoing the colors inside, sound sets you in a mood to take certain things."

With such power in sound, it is a shame it has been reduced to its present status.

"Currently," Portman concludes, "sound jobs tend to be salvage jobs — 'Gee, boy, that was a good save.' You're saving sound all the time."

What he is proposing is that motion picture sound, in this age of astonishing advancement in the audio field, be taken from the level of "a good save," and be given the artistic status it deserves. □□□

record, there's always going to be a certain amount of surface noise." [The BBC effects library is now reported to be available on cassette tape in Britain — SB.]

"Basically," says Flick, "there are very few modern sound libraries."

In addition to the lack of high-fidelity sound quality, most sound libraries do not contain all the effects needed for a modern feature film. This leads the sound editor into a scavenger hunt around town to find the right sound.

"You call your friends," says Anderson. "You remember, 'Wow, there was a great sound in such and such a picture.' So you track down who did the sound on the film, and then track down the effect.

"Another problem is that most libraries will not have a series of effects of one thing. Say you need a car start sound, so you go to the library and get a 1959 Plymouth starting, recorded for one show at one time. And then you need the same car pulling away, but they don't have that same Plymouth. They have something totally foreign — like a 1963 Ford going by — and that was recorded at a different time, and in a different location."

The result is a bothersome lack of consistancy.

"We feel," continues Anderson, "that you should go out, get a car, and record virtually everything that it does."

Flick and Anderson have done this to assemble their own modern effects library. Their collection consists of sounds recorded fresh for the library; sound effects culled from the tracks of recent features the team has worked on; and fresh effects acquired from other editors and production mixers. Quarter-inch masters make up the backbone of the library, so that the fewest possible generations are involved in the transfer and mix process.

Nevertheless, Flick offers, "the best thing to do is to get the production recordist to go out and record most of the sounds that you need for the show. If the recordist is any good, then the show will have a fresh, special feel to it."

Recording Effects on Location

In order to accomplish this, however,

Transferring sound effects and dialogue from ¼-inch tape to 35 mm magnetic stock at Goldwyn Sound Services, Warner Hollywood Studios. Engineer Jim Vavarretta is adjusting the resolver unit to ensure that tapes are replayed at the same speed as that used when the original picture was shot.

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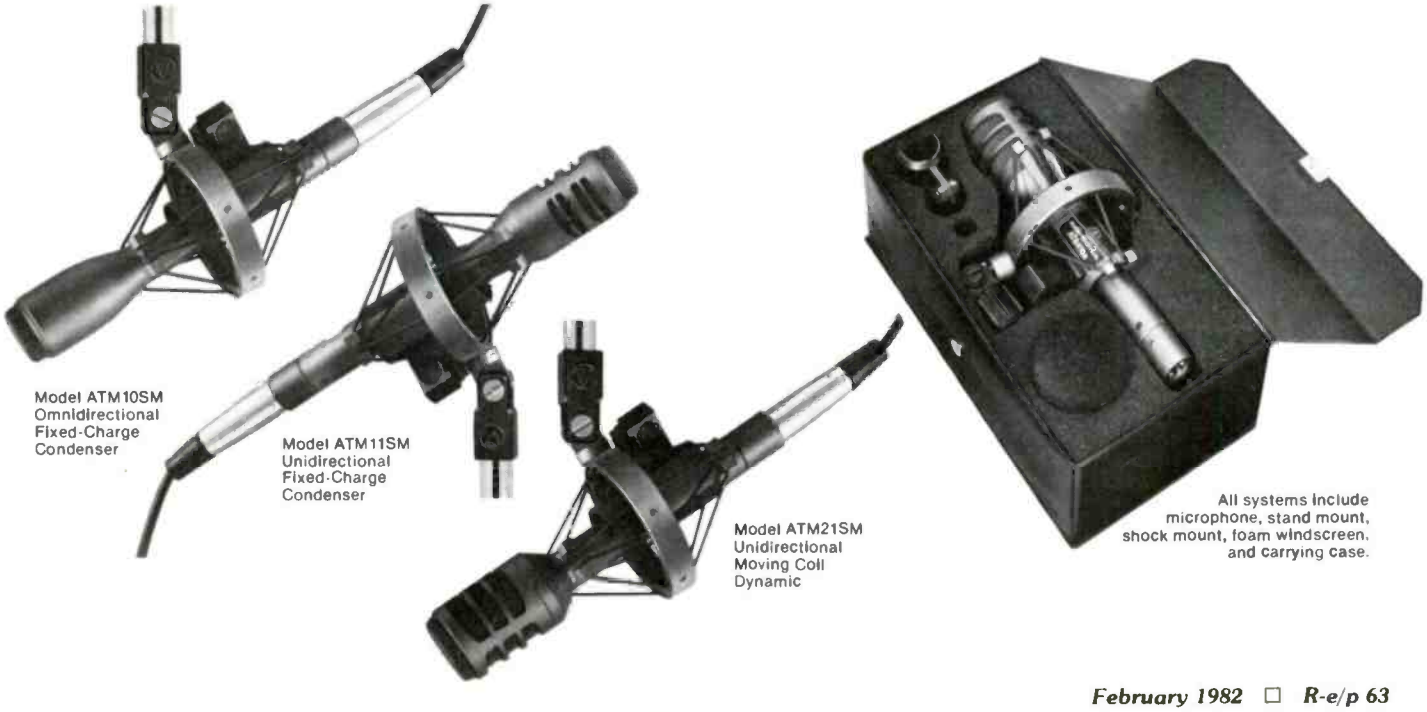

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the production mixer requires a quiet set, which in turn requires that all work stop for the time it takes to record whatever sound it may be. It is difficult to achieve that kind of cooperation on a large production, where the cast and crew would end up being paid about $2,000 per minute to do nothing while the recordist taped the sound of a train engine hissing steam.

The company may also be faced with the cold reality of a disappearing sun and waning light, which forces the production to move on, forsaking the sound editor's needs. It is an expensive proposition, and most companies try to cut costs on the set and pass the problem on to the sound editors in post-production.

Consequently, most of the production sound effects that find their way into the motion picture's soundtrack were taken from printed shots or scenes, trims of unwanted sound, or out-takes (¼-inch tape transferred at the request of the sound editor, but not originally intended for use in the film). None of these sources were originally intended or designed for optimum sound recording since, among other things, the mike had to be out of the camera's view. While they may not be the best, at least they're available, and an imaginative sound editor can find possibilities in anything.

"One picture we did was called 'The Big Brawl,'" offers Anderson, "and about four reels of the show involve these exterior fights in front of large crowds. What they should have done was to have taken five minutes, quieted everybody down, and then recorded a couple of cheers from start, to peak, to decline, plus a couple of different boos and, from the same perspective, a general crowd 'walla.'

"Five minutes of somebody thinking on the set would have saved us days and days of agony, and covered our needs for those reels. Instead, we had to go through tons of production tracks to find material that would work.

"We discovered that when they shot the scenes, people were just yelling at one constant level. As a result, when they cheered there was a minimal rise. What we used, then, was the head of a sound take when the assistant director yelled, 'Background!,' and the extras would go from quiet and build to full volume. We found that that would work as a cheer, though in reality it was just the crowd 'walla' starting up."

If they can be arranged, another advantage to effects recorded specifically for a film — either on the set or in post-production — is the relative ease with which they can be made to work in a shot.

"Now nothing's going to drop right in," says Anderson, "because you can't usually do it to picture like A.D.R. [Automatic Dialogue Replacement], but if you go out and record, say, a car doing exactly what the car does in the show, it will be a lot easier to fit the sound into sync and into the mood of the scene than an effect from a sound library."

The machine room at GSS Re-Recording houses 16 three-track replay machines which are interlocked with the projector during dubbing sessions, shown with machine room engineer Kevin O'Connell.

The Complexity of Effects

Specific effects, such as those discussed so far, appear in themselves to be rather simple, but they are surprisingly rich and complex. Take, for example, one of the most fundamental sounds in the motion picture: that of a door.

"There are door slams and there are door slams," explains Keene. "You have to learn how to discriminate. Is it a hardwood door? Is it a solid-core door? Is it a front door? Is it a bedroom door? Does it rattle? Is it a cheap apartment? Just what kind of door is it?"

Also, there are many individual elements that make up what is often viewed as the single sound of a door.

"Do you know how many sounds a wood door has?" asks Flick. "It chatters . . . it squeaks . . . it has latch noise. The door knob turns . . . and then doors bang. They close lightly. Just think for a moment about how many sounds a door has."

This manner of thinking and methodology is applied to nearly every object seen on the screen that makes a sound. Consequently, regardless of the source of the sound effect, some sort of manipulation is usually required to make the sound work with the visible object.

Putting the two aspects in sync is the most basic concern. The time interval between a door opening and then closing in the library effect may be shorter than the time it takes on the screen, so that space has to be extended. In addition, the quality of the sound may require adjustment. This can be done with signal-processing gear in the re-recording process, or in the editing stages through physical manipulation by the cutter.

"I worked on the 'Gone With the Wind' segment of *Movieola*," says Keene, "and there was a scene in which [the actor portraying] Clark Gable is out shooting ducks on a lake. Well, in the dialogue track they had cut in a duck call, probably because they had screened it for someone in rough cut. The effect had been recorded hot, apparently, because they had carefully

scraped it until it sounded just like a part of the production track. I actually pulled that thing and used it in my effects track, because it was better than anything I could come up with. You could see how someone had spent hours scraping this thing to get it to work."

Scraping and sand papering the mag-film to reduce the oxide level of a recording is as common in sound effects as it is in dialogue cutting. But sound effects, because they are not the recognizable human voice, hold many other possibilities.

"Often," says Keene, "we have to manipulate a sound to get it to sound like something else."

"We took the fishing reel from 'Islands in the Stream'," recalls Flick, "slowed it down in transfer with a vari-speed device, and used it as a cable release in *The Final Countdown*."

"Sometimes you've got to make something that doesn't exist," Anderson offers. "A scene may have a giant castle door opening [probably a light wood set]. Now you've got a pretty big scraping, sliding door in your library, but it doesn't sound 'big' enough. So you slow it down a certain percentage, and suddenly it's big, huge, and rumbly; a simple slowing down can affect the nature of a sound."

While equalization and reverb can be applied in the mix, as Flick explains, "you cannot efficiently do a speed up and slow down on a re-recording stage, so we've got to do that tailored preparation for the mix."

Along these lines, Portman discusses further the creation of sound effects that do not exist in reality: "An editor is often required to manufacture sound from other, not necessarily related, sounds."

Taking for example, a gun shot, Portman offers, "If it's fired by the good guy — say, The Lone Ranger — it has to sound different from the rest. Consequently, it may be made out of several different elements: it may have the 'click' part from anything; it may have as the explosion the head of a roll of thunder cut off short; it may be five or six things which, with reverb and what have you added, will make that sound.

"Now the sound editor has got to be enormously clever to do this, because he can usually only hear the sounds one track at a time. So a good editor uses his or her imagination, and can 'hear' what it will sound like. But it's still like playing Russian Roulette, because the first time the editor will hear the composite sound is on the re-recording stage when everybody's there."

The subject of composite sound effects is only one area that points to the critical need for a good working relationship between the mixers and sound editors. If they are unable to communicate, the composite sound will not come out anything like the way it was planned — much to the disappointment of all concerned.

Synthesized Sound Effects

Along with these "editor manufactured" effects, recent science-fiction films have seen an increase in the use of synthesized sound effects for motion pictures.

Alan Howarth was associated originally with Flick and Anderson on *Star Trek — The Motion Picture*, and has worked with them since on *The Final Countdown*, and *The Hand*, among others. Though Howarth was largely responsible for the sounds of the time-shifting storm in *The Final Countdown*, his talents were also called upon to enhance the sounds of Navy jets seen in the picture, since the existing recordings lacked impact. He used a Sequential Circuits Prophet synthesizer to generate sounds that helped supplement the plane effects. This form of sound effect is a new source that Flick and Anderson are taking advantage of to their benefit, as well as that of the synthesizer creators.

Use of Effects to Enhance the Drama

Effects called for by action on screen — sounds that fall in the "See a dog, Hear a dog" category — or off-screen effects called for by the script, are what is required to make a movie's soundtrack believable on the most basic level. However, the sound editor, when

permitted, can go beyond that to add to the drama, to make a serious creative contribution to the film in the areas of backgrounds and specific effects.

"At the start of *In the Heat of the Night*," recalls Portman, "the cook took a cover off some food and a fly flew out. Now that buzzing had to be added to carry the moment, but an even better example came later when the editor helped provide a light moment in an intense scene.

"In the first confrontation between Portier and Steiger, there was an air conditioner that malfunctioned, stopped, and then went back to running when Steiger hit it. To get the sound, Jim Richard took an electric fan on to a sound stage and stuck a dowel into it to make the 'bonk, bonk, bonk.' Then he made another sound for when Steiger hit it, then a 'crenk, crenk, crenk,' and then it settled down."

This seemingly mechanical procedure is more than just that, for although Steiger brilliantly used the prop of the air conditioner to help establish his character, it had to be made believable by the sound editor. It may even be the sound editor's idea, approved subsequently by a director who welcomes contributions from all quarters.

"In *Kotch*," explains Portman, "Walter Matthau's character had an old car, a Packard or something. When they shot the film, it was just a regular car, but the sound editors, Kay Rose and Frank Warner, went out after shooting and got the car. Then Frank took a ball peen hammer and pounded in the oil pan under the engine just far enough so that when the connecting rod came around, it would hit the oil pan. Since the car had a unique sound, they shot [taped] a whole series of sound effects of this car with this new sound. They had it starting and stopping, car in, car out; a whole library of this car.

"Now the thing had real character; a special sound reflecting the character of the old man. In the picture, the chase was essentially the night time ride down the hill to get the girl to the hospital to have her baby. The car develops more menace, because the audience is asking itself, 'Will it get there?,' and that can be built upon by increasing the knocking sound as the drive progresses, or by adding another sound — from 'klank, klank, klank,' to 'klank, boom, klank, boom.' A lot of work.

"They've added so much. What can be more dull than watching a car going from 'A' to 'B'? But they needed the shot, and now that the sound is added, it isn't dull; it continues the man's character, even though you're only seeing the car."

On another picture with sound editor Kay Rose, *Ordinary People*, there was a close shot of the back end of the car as it pulled into the garage. Rather than just having the car doors open and close,

Engineer Vince Zabaly operating a sound transfer unit comprised of a Nagra IV ¼-inch replay machine and a self resolving 35 mm mag recorder, at Market Street /Tony Bill Productions, Venice, California.

Rose cut in the buzzer warning that the keys were still in the ignition. While the shot simply established, via the license plate, the locale of the story, by adding the buzzing sound a subtle tension was created, and character was revealed: the keys in the ignition said that these people trust their neighborhood, and feel secure in their house.

Highlighting or 'Signature' Effects

Sound can be used to keep characters straight if they cannot really be seen, but are only represented by objects or machines.

"On *Grand Theft Auto*," offers Anderson, "there were a million cars, and often you didn't even see the people in them. Because you identified the characters by the cars they drove, we had to come up with a different sound for each car to give them individual characters."

Flick and Anderson call these types of effects "signature effects."

"In *Invasion of the BodySnatchers*," explains Flick, "the pods had a signature sound — a special sort of greasy, bubbly, popping sound — that they made as they became human beings." When heard off screen, the sound told the audience that a pod was nearby, performing its awful function.

"These types of signature effects can be both literal or physical [as described above], or they can work on a more evocative level," Flick hazards.

"With regard to the latter," offers Anderson, "A friend of ours was working on the new *Lone Ranger* movie. Well, his gunshots have got to have a different sound from the bad guy's gunshots."

"And Silver's hoof beats have got to sound big and powerful," adds Flick, "because he's The Hero's horse! These sounds are evocative."

Signature sounds can also work on a more subliminal, almost unconcious level.

"In *Sorcerer*," recalls Flick, "the two trucks had a lion and a tiger roar cut into the engine effects. They had them growling and roaring when the engine reved, or what have you. But they were mixed in very subtly, so that you weren't consciously aware of them."

Another way to subtly evoke feeling and mood is through the use of sounds that Flick and Anderson refer to as motiffs, delineated from signature

effects.

"Now the pods had their signature," explains Anderson, "but one of the main motiffs of the city in *Invasion of the Body Snatchers* was the off-screen sounds of sirens and garbage trucks, which indicated the accidents and disasters and people dying, and the carrying away of bodies that had been replaced by pods."

It is in background sounds and off-screen effects that the sound editor can make a tremendous contribution to the mood of the picture. Following the first step of making them realistic comes the dramatic enhancement via background sounds.

You start, according to Anderson, "by sitting down and asking yourself, 'If I was at this location, what would I hear?' You ask: 'Where is this place?' Say it's a suburban tract home at seven o'clock in the morning. Now you ask, 'What would there be?' Well, there might be some wind; there might be a dog barking at someone; there might be lawn sprinklers going. And you go on."

These and dozens of other sounds are built into the background reels by the sound editor to evoke a real sense of the location, or, as Flick adds, "to characterize the place with sound, so that it aids the drama without distracting from the story."

"An on-screen car starting up and driving off is your star," explains Anderson. "But then you need all the supporting characters in the backgrounds."

"Backgrounds are where it's at," says Portman. "That's the tone, the mood of a scene; the fun stuff. Tenements are marvelous places for backgrounds.

"In a picture I did called *The Landlord*, we had things happening on every floor. Somebody would flush a toilet, and you could hear it progressively running down through the walls. People were wailing through the halls and having arguments. Televisions were breathing in and out. It was great."

Beyond sounding simply realistic, the drama can be underlined with what can be termed an audio sub-text to the story.

"In *The Hand*," Anderson explains, "Michael Caine's character starts out happy in the country, so we made it light — some crickets, and a frog or two; real nice. Now, after losing his hand, they move to New York into an apartment his wife wanted to get but he didn't. He hates New York, so we filled it up with noise: wall to wall traffic; sirens off screen; an off-screen kid in the next apartment practicing piano scales and hitting clunker notes; the trashmen emptying the dumpster, going 'bam, bam, bam' with their hydraulic truck."

This is an example of using contrast for dramatic effect. One can move in the reverse direction as well, by going from an oppressive environment to a pleasant one; again contrast is the key. To emphasize quietness in film, say both Flick and Anderson, you put the noise before it. It is the lack of noise that will call attention to itself, after the audience has become accustomed to an obtrusive, noisy background. Quiet by itself can become quite dull.

***

With the picture finished and mixed, the sound editor takes his own satisfaction and the respect of his colleagues, and moves on to his next assignment. Curiously, when the Acadamy Awards are given for best motion-picture sound, Oscars are presented to the production and re-recording mixers. The craftsmen in between are not currently accorded the honor of an annual award. Most are thanked by those who do receive the recognition.

In 1980, however, a special award was presented by the Academy of Motion Picture Arts and Sciences to Alan Splet, for his outstanding achievement in the field of film sound editing on the motion picture, *The Black Stallion*. It's a beginning.

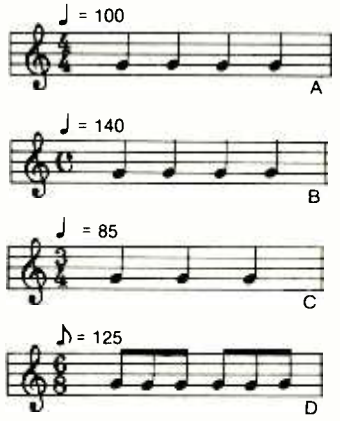
Creative Use of Delay Rhythms . . . with Comprehensive Tabulations of Tempo and Delay Settings . . . for Both DDL and Tape Application . . . with Examples from Various Musical Idioms . . . by Roman Olearczuk

The use of time delay and repeats is standard practice for today's pop music recordings. A large variety of effects can be created through the use of sophisticated digital delays, in addition to the conventional methods of generating time delay from analog tape machines. In this article, a technique[1] is presented that relies on musical tempo as a basis for selecting time delay values to produce desired delay and repeat effects.

Some Musical Terms and Definitions

Tempo can be defined as the relationship between the beat or pulse of the musical piece in question, and its physical duration in time.[2] Exact tempo is noted through the use of metronome markings, which are usually written above the time signature at the start of a musical chart. Several single-measure examples of tempo markings and common time signatures as found in popular music today are shown in Figure 1.

Figure 1



The meter or time signature in Figure 1A, and its alternate notation in Figure 1B, is frequently used in most rock, popular and ballad meters. The lower number in the time signature (for example, 4 in a 3/4 signature) indicates the length of the note that is the pulse or beat in a given composition. The number of beats within a measure is denoted by the upper number. Therefore, as an example, the time signature of 3/4 — the basic waltz meter — in Figure 1C would require a quarter-note beat with three (quarter-note) beats per measure.

The tempo marking of 100 in Figure 1A signifies that there are 100 quarter-note beats per minute in this particular example.

— the author —

Until recently, **Roman Olearczuk** *served as technical director for Rusk Sound Studios in Hollywood, where he supervised the initial construction stages, and was responsible for the maintenance and engineering staff. His professional audio career started at Kendun Recorders/Sierra Audio, where he was a maintenance/field engineer. Roman is also a Consulting Editor for* R-e/p.

Figure 1D, with a 6/8 time signature used frequently in jazz/rock compositions, shows a tempo marking based on eighth-note pulses. In this case, the tempo would be set at 125 eighth-note beats per minute.

Delay and Rhythm Relationship

The use of delay to provide an ambience and an enhancement to the sound of various instruments and voices is an accepted studio technique. A general range of delay times that are used for effects, such as automatic double tracking (ADT), slap echo, and long echo repeats, when mixed with the original signal, is shown in Table 1[3].

Table 1: Delay Time for Various Special Effects

Delay Time	Effect
20 — 40 msec.	ADT (echo not distinct)
40 — 50 msec.	Echo becomes distinct
50 — 125 msec.	Slap-echo (short)
125 msec., and above	Echo (long)

Frequently in studio practice, an engineer picks a particular delay time by "feel," "experience," or "how it sounds" in the mix — usually without regard to the musical tempo and the underlying pulse of the song under production. The technique presented here relates the beat and the pattern of notes to be delayed as subdivided rhythms. By setting the interval between the original and the delayed note pattern for a value that is a subdivision of the basic rhythmic beat, an engineer can achieve musically pleasing results sooner than with the "hit or miss" methods described above.

Table 2 tabulates tempo versus subdivided rhythms from 70 to 150 beats per minute. This range covers the most common musical styles from ballads to New Wave. An additional column, based on a 4/4 time signature, labelled "TIME (in seconds for 4 bars)" is provided for use when the tempo is unknown. This table was constructed using the following formulae:

$$\frac{1\ \text{minute}}{t\ \text{beats}} \times \frac{60\ \text{seconds}}{1\ \text{minute}} \times \frac{1000\ \text{milliseconds}}{1\ \text{second}}$$

OR

$$\frac{60(10^3)\ \text{milliseconds}}{t\ \text{beats}} = \frac{60(10^3)}{t}\ \text{milliseconds/beat}$$

where t = tempo value

The resulting values listed here provide a useful measure of time duration for each of the subdivided beats. The use of this chart will become evident during example descriptions in the following studio applications section.

Table 3 provides a listing of delay times versus tape speed for an Ampex ATR-100 Series analog tape recorder. Analog tape delay is still used quite extensively in studios today, in part due to the availability of such high-quality machines. The chart shows a range of delay times, from 20 to 270 milliseconds, available at the 30/15 IPS tape speeds with VSO engaged on the recorder. At the slower 7½/3¾ IPS speeds, delays as long as one second can be obtained with use of the VSO. Although both tape saturation and signal-to-

. . . Tables 2 & 3 overleaf . . . text continues on page 72 —

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TABLE 2: DELAY TIME VERSUS RHYTHM

TEMPO (Beats per Min.)	4/4 TIME (Secs for 4 Bars)	DELAY (in millisec) VERSUS RHYTHM (based on ¼ note beat)				
		1/4 BEAT	1/8 BEAT	1/16 BEAT	1/32 BEAT	1/64 BEAT
150	6.40	400.0	200.0	100.0	50.0	25.0
149	6.44	402.7	201.3	100.7	50.3	25.2
148	6.49	405.4	202.7	101.4	50.7	25.3
147	6.53	408.2	204.1	102.0	51.0	25.5
146	6.58	411.0	205.5	102.7	51.4	25.7
145	6.62	413.8	206.9	103.4	51.7	25.9
144	6.67	416.7	208.3	104.2	52.1	26.0
143	6.71	419.6	209.8	104.9	52.4	26.2
142	6.76	422.5	211.3	105.6	52.8	26.4
141	6.81	425.6	212.8	106.4	53.2	26.6
140	6.86	428.6	214.3	107.1	53.5	26.8
139	6.91	431.7	215.8	107.9	53.9	27.0
138	6.96	434.8	217.4	108.7	54.3	27.2
137	7.00	437.9	219.0	109.5	54.7	27.4
136	7.06	441.1	220.6	110.3	55.1	27.6
135	7.11	444.4	222.2	111.1	55.6	27.8
134	7.16	447.8	223.9	111.9	56.0	28.0
133	7.22	451.1	225.6	112.8	56.4	28.2
132	7.27	454.5	227.3	113.6	56.8	28.4
131	7.33	458.0	229.0	114.5	57.3	28.6
130	7.38	461.5	230.8	115.4	57.7	28.8
129	7.44	465.1	232.6	116.3	58.1	29.1
128	7.50	468.8	234.4	117.2	58.6	29.3
127	7.56	472.4	236.2	118.1	59.0	29.5
126	7.62	476.2	238.1	119.0	59.5	29.8
125	7.68	480.0	240.0	120.0	60.0	30.0
124	7.74	483.9	241.9	121.0	60.5	30.2
123	7.80	487.8	243.9	121.9	61.0	30.5
122	7.87	491.8	245.9	122.9	61.5	30.7
121	7.94	495.9	247.9	124.0	62.0	31.0
120	8.00	500.0	250.0	125.0	62.5	31.2
119	8.07	504.2	252.1	126.0	63.0	31.5
118	8.14	508.5	254.2	127.1	63.6	31.8
117	8.21	512.8	256.4	128.2	64.1	32.1
116	8.28	517.2	258.6	129.3	64.7	32.3
115	8.35	521.8	260.9	130.4	65.2	32.6
114	8.42	526.3	263.2	131.6	65.8	32.9
113	8.50	531.0	265.5	132.7	66.4	33.2
112	8.57	535.7	267.9	133.9	67.0	33.5
111	8.65	540.5	270.3	135.1	67.6	33.8
110	8.73	545.5	272.7	136.4	68.2	34.1
109	8.81	550.5	275.2	137.6	68.8	34.4
108	8.89	555.5	277.8	138.9	69.4	34.7
107	8.97	560.7	280.4	140.2	70.1	35.0
106	9.06	566.0	283.0	141.5	70.8	35.4
105	9.14	571.4	285.7	142.9	71.4	35.7
104	9.23	576.9	288.5	144.2	72.1	36.0
103	9.32	582.5	291.3	145.6	72.8	36.4
102	9.41	588.2	294.1	147.1	73.5	36.8
101	9.50	594.0	297.0	148.5	74.2	37.1
100	9.60	600.0	300.0	150.0	75.0	37.5
99	9.70	606.1	303.0	151.5	75.8	37.9
98	9.80	612.2	306.1	153.1	76.5	38.3
97	9.90	618.5	309.3	154.6	77.3	38.6
96	10.00	625.0	312.5	156.3	78.1	39.1
95	10.11	631.6	315.8	157.9	78.9	39.5
94	10.21	638.3	319.1	159.6	79.8	39.9
93	10.32	645.1	322.6	161.3	80.6	40.3
92	10.43	652.2	326.1	163.0	81.5	40.8
91	10.55	659.3	329.7	164.8	82.4	41.2
90	10.67	666.7	333.3	166.7	83.3	41.7
89	10.79	674.2	337.0	168.5	84.3	42.1
88	10.91	681.8	340.9	170.5	85.3	42.6
87	11.03	689.7	344.8	172.4	86.2	43.1
86	11.16	697.7	348.8	174.4	87.2	43.6
85	11.29	705.9	352.9	176.5	88.2	44.1
84	11.43	714.3	357.1	178.6	89.3	44.6
83	11.57	722.9	361.4	180.7	90.4	45.2
82	11.71	731.7	365.8	183.0	91.5	45.7
81	11.85	740.7	370.4	185.2	92.6	46.3
80	12.00	750.0	375.0	187.5	93.7	46.9
79	12.15	759.5	379.8	189.9	94.9	47.5
78	12.30	769.2	384.6	192.3	96.2	48.1
77	12.47	779.2	389.6	194.8	97.4	48.7
76	12.63	789.5	394.7	197.4	98.7	49.3
75	12.80	800.0	400.0	200.0	100.0	50.0
74	12.97	810.8	405.4	202.7	101.4	50.7
73	13.15	821.9	410.9	205.5	102.7	51.4
72	13.33	833.3	416.7	208.3	104.2	52.1
71	13.52	845.1	422.5	211.3	105.6	52.8
70	13.71	857.1	428.6	214.3	107.2	53.6

TABLE 3: DELAY TIME VERSUS TAPE SPEED FOR AMPEX ATR 100

DELAY (Millisec)	TAPE SPEED	%VSO (30 IPS)	%VSO (15 IPS)	%VSO (7½ IPS)	%VSO (3¾ IPS)
20	91.50	305.0			
21	87.15	290.5			
22	83.10	277.3			
23	79.56	265.2			
24	76.26	254.2			
25	73.20	244.0			
26	70.38	234.6			
27	67.77	225.9			
28	65.37	217.9			
29	63.09	210.3			
30	60.99	203.3			
31	59.01	196.7			
32	57.18	190.6			
33	55.44	184.8			
34	53.82	179.4			
35	52.29	174.3			
36	50.82	169.4			
37	49.47	164.9			
38	48.15	160.5			
39	46.92	156.4			
40	45.75	152.5	305.0		
42	43.56	145.2	290.5		
44	41.58	138.6	277.3		
46	39.78	132.6	265.2		
48	38.13	127.1	254.2		
50	36.60	122.0	244.0		
52	35.19	117.3	234.6		
54	33.90	113.0	225.9		
56	32.67	108.9	217.9		
58	31.56	105.2	210.3		
60	30.51	101.7	203.3		
62	29.52	98.4	196.7		
64	28.59	95.3	190.6		
66	27.72	92.4	184.4		
68	26.91	89.7	179.4		
70	26.13	87.1	174.3		
72	25.41	84.7	169.4		
74	24.72	82.4	164.9		
76	24.09	80.3	160.5		
78	23.46	78.2	156.4		
80	22.89	76.3	152.5	305.0	
82	22.32	74.4	148.8	297.6	
84	21.78	72.6	145.2	290.5	
86	21.27	70.9	141.9	283.7	
88	20.79	69.3	138.6	277.3	
90	20.34	67.8	135.5	271.1	
92	19.89	66.3	132.6	265.2	
94	19.47	64.9	129.8	259.6	
96	19.05	63.5	127.1	254.2	
98	18.66	62.2	124.5	249.0	
100	18.30	61.0	122.0	244.0	
102	17.94	59.8	119.6	239.2	
104	17.58	58.6	117.3	234.6	
106	17.25	57.5	115.1	230.2	
108	16.95	56.5	113.0	225.9	
110	16.65	55.5	111.0	221.8	
112	16.38	54.6	108.9	217.9	
114	16.05	53.5	107.0	214.0	
116	15.78	52.6	105.2	210.3	
118	15.54	51.7	103.4	206.8	
120	15.24	50.8	101.7	203.3	
122	15.00	50.0	100.0	200.0	
124	14.76	49.2	98.4	196.7	
126	14.52	48.4	96.8	193.7	
128	14.28	47.6	95.3	190.6	
130	14.07	46.9	93.8	187.7	
132	13.86	46.2	92.4	184.8	
134	13.65	45.5	91.0	182.1	
136	13.47	44.9	89.7	179.4	
138	13.26		88.4	176.8	
140	13.07		87.1	174.3	
142	12.89		85.9	171.8	
144	12.71		84.7	169.4	
146	12.54		83.6	167.1	
148	12.36		82.4	164.9	
150	12.20		81.3	162.7	
152	12.05		80.3	160.5	
154	11.88		79.2	158.4	
156	11.73		78.2	156.4	
158	11.58		77.2	154.4	
160	11.45		76.3	152.5	305.0
165	11.09		73.9	147.9	296.8
170	10.77		71.8	143.5	287.1
175	10.46		69.7	138.4	278.9
180	10.17		67.8	135.6	271.1
185	9.90		66.0	131.9	263.8
190	9.63		64.2	128.4	256.8
195	9.38		62.6	125.1	250.3
200	9.15		61.0	122.0	244.0
205	8.93		59.5	119.0	238.1
210	8.72		58.1	116.2	232.4
215	8.51		56.7	113.5	227.0
220	8.33		55.5	110.9	221.8
225	8.13		54.2	108.4	216.9
230	7.95		53.0	106.1	212.2
235	7.79		51.9	103.8	207.6
240	7.62		50.8	101.7	203.3
245	7.47		49.8	99.6	199.2
250	7.32		48.8	97.6	195.2
255	7.17		47.8	95.7	191.4
260	7.04		46.9	93.8	187.7
265	6.90		46.0	92.1	184.2
270	6.78		45.2	90.4	180.7
280	6.53			87.1	174.3
290	6.31			84.1	168.3
300	6.10			81.3	162.7
310	5.90			78.7	157.4
320	5.72			76.3	152.5
330	5.54			73.9	147.9
340	5.39			71.8	143.5
350	5.23			69.7	139.4
360	5.09			67.8	135.6
370	4.95			66.0	131.9
380	4.82			64.2	128.4
390	4.70			62.6	125.1
400	4.58			61.0	122.0
410	4.46			59.5	119.0
420	4.36			58.1	116.2
430	4.25			56.7	113.5
440	4.16			55.5	110.9
450	4.07			54.2	108.4
460	3.98			53.0	106.1
470	3.89			51.9	103.8
480	3.81			50.8	101.7
490	3.74			49.8	99.6
500	3.66			48.8	97.6
510	3.59			47.8	95.7
520	3.52			46.9	93.9
530	3.45			46.0	92.1
540	3.39			45.2	90.4
560	3.27				87.1
580	3.15				84.1
600	3.05				81.3
620	2.95				78.7
640	2.86				76.3
660	2.77				73.9
680	2.69				71.8
700	2.61				69.7
720	2.54				67.8
740	2.48				66.0
760	2.41				64.2
780	2.35				62.6
800	3.00				61.0
820	2.23				59.5
840	2.18				58.1
860	2.13				56.7
880	2.08				55.5
900	2.03				54.2
920	1.99				53.0
940	1.95				52.0
960	1.91				50.8
980	1.87				49.8
1000	1.83				48.8
1020	1.79				47.8
1040	1.76				46.9

Some Studio Applications of Delay/Rhythm Technique

Example 1 — Snare drum with tape slap echo:

Instrument: Snare drum.

Rhythm: 4/4 rock tune with predominant snare

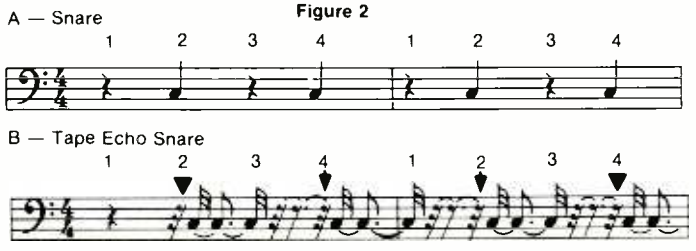
Effect: Tape slap echo after original snare beat.

Tempo: Unknown.

Technique: The engineer obtains a time sample of 4 bars of music using a stop watch. Usually a start of a verse or chorus provides a convenient place to count off "1-2-3-4, 1-2, etc." A value of 8.00 seconds is measured, which corresponds to a tempo of 120 in Table 2. A value of 62.5 milliseconds for slap-echo is chosen from the corresponding subdivided beats. Table 3 yields interpolated tape speed of 29.28 IPS, or a VSO value of approximately 97.6% at 30 IPS on an ATR-100, to provide the engineer with the desired tape slap effect.

The snare beat and resulting tape slap beat are charted musically in Figure 2. The delayed interval of 62.5 milliseconds between the original and slap notes in Figure 2B is indicated as a 1/32-note rest. The remaining notes have the same musical value as the original pattern. However, they have been displaced to the right by the additional 1/32-note rest.

Figure 2



Arrow [▼] denotes 1/32-note rest due to delay.

(Note: This figure is for illustrative purposes only; it would not appear this way on an annotated chart.

Example 2 — Background vocals with ADT effect:

Voices: Background vocals; high and low parts panned left and right.

Rhythm: 4/4 ballad.

Effect: Double-track voices to enlarge parts.

Tempo: 100 beats per minute.

Technique: An engineer sets one channel of a DDL for an approximate setting of 38 milliseconds (exact value = 37.5 msec). This time is chosen from Table 2 for the corresponding tempo of 100. The delay is panned opposite from the original low part. This process is repeated for the high part with another DDL. Delay times of 19 and 57 milliseconds are set on the remaining open channels of each DDL. This additional cross-delay is then adjusted for musical taste. The additional delay times were derived by using the same basic principle of subdividing the 38 millisecond delay in half, and then adding or subtracting this increment to achieve additional values not present on the chart (i.e. 38 ± *19 milliseconds).*

Example 3 — Repeating guitar note during a musical break:

Instrument: Electric lead guitar.

Rhythm: 3/4 slow country ballad.

Effect: Last note played before a one-bar break; note repeats and fades throughout the musical rest

Tempo: 85 beats per minute.

Technique: Table 2 shows that an engineer has several choices of long echo repeats. As a first choice, he decides to try the quarter-note 706 millisecond repeat. Since the last guitar note was played on "3," it should then repeat three times or three quarter-notes over the next bar rest. The fader, with the repeat echo assigned to it, is turned on after "3" and is faded off by the end of three repeats.

The lead guitar and the repeating echo charts are shown in Figure 3. The repeat echo could be achieved using either an appropriate digital delay that has the capability of long delay times, or through analog tape delay by setting up the correct value selected from Table 3. (Note that even though VSO settings are only provided for delay times of 700 and 720 milliseconds, the required value can be obtained by interpolating between these two values.)

noise ratio levels are degraded at these speeds, an engineer can achieve results that rival expensive digital delay lines with the help of a peak limiter and dbx noise reduction. (Dolby noise-reduction units, being single-ended processors with switchable encode/decode functions, normally cannot be used with real-time, tape-based delay systems.)

The formula used to generate this table is as follows:

$$d = 1000 \times 100\ h/(v \times s)$$

Where:

d = delay time, in milliseconds

h = head spacing *measured* between record and repro heads, in inches

v = VSO setting, as percentage of original tape speed

s = tape speed, in inches per second

(NOTE: h = 1.83 inches for an Ampex ATR-102/4 tape machine.)

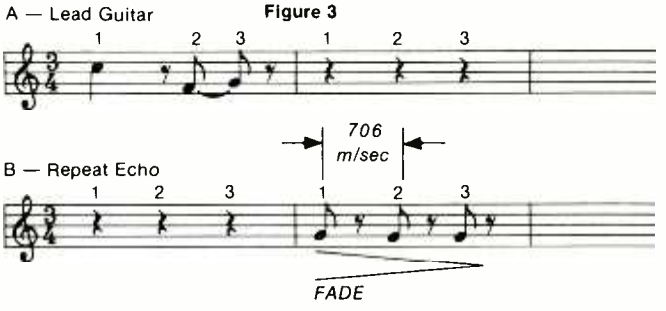
The computed delay times are conveniently arranged in Table 3, in such an order that each time value coincides with the overlapping VSO settings at the different tape speeds.

It should be noted that Table 2 was constructed with a quarter-note as the basic pulse. This fact is acknowledged by the label "1/4 (beat)" over the leftmost delay column. To use this chart for beats other than a quarter-note, just substitute the new beat value for this column heading, and subdivide this heading value for the other columns.

The delay value in each column would remain the same. For example, a 6/8 time signature has an eighth-note pulse, so the headings for each column would now read 1/8, 1/16, 1/32, 1/64, and 1/128.

If the tempo is unknown for a time signature other than 4/4, obtain a time sample of four bars of music, divide the value by four (bars), and then divide this result by the number of beats in a measure. The final value should provide a measurement for the duration of a single beat. In this way the chart shown in Table 2 can be used to obtain other subdivided time delay values.

Figure 3



Conclusion

The basic techniques presented here can be experimented upon and expanded further to include music written in other time signatures besides the commonly used 4/4 meter. Audio effects that have not been described here and involve the manipulation of time should be examined first, for their relationships to the tempo of the music under production.

References

1. This technique was explained to the author by engineer Juergen Koppers during sessions with producer Giorgio Moroder. Apparently, during a mix Moroder asked Koppers if the underlying eighth-note synthesizer pulse could be changed electronically to sixteenth-notes. By time sampling four bars of music, Juergen calculated the necessary needed DDL delay setting to achieve this effect.

2. *A Programmed Introduction to the Fundamentals of Music,* by Theodore Ashford; William C. Brown Company Publishers, Dubuque, Iowa. *Chapter 4 — Other Aspects of Notation: Rhythm.*

3. *Why You Can't Wait For: Digital Delay,* by Ken Schaffer; *R-e/p, Vol. 7, No. 4, August, 1976; pages 55 to 65.*

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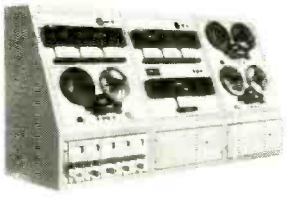

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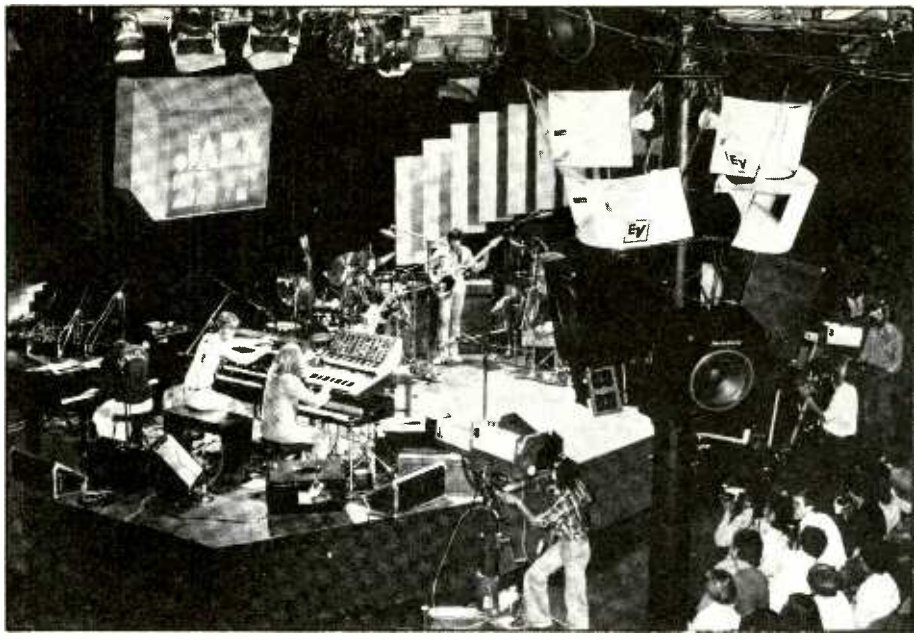

THE CONCERT SOUND REINFORCEMENT SYSTEM FOR THE MONTREUX JAZZ FESTIVAL

In days past the designers of concert environments realized that not only should the venue *look* right, but also it should — wait for it — *sound* right! The ancient Greeks got it together with their marvellous amphitheatres, which were both aesthetically pleasing as well as having a virtually even sound distribution — often reinforced by the use of walls of amphorae resonating at different frequencies backstage. (And this was 2,500 years ago!)

Making the jump into modern history, the famous opera houses were/are works of art, but bent for disaster if the leading musicians and singers of the day could not be heard perfectly. Moreover, they would have been shunned by the selfsame artists.

"Looks right — sounds right"; an easy formula to remember, but all too often forgotten — at least the latter half.

In recent times the prevailing attitude to concert sound has been very much a case of: "I don't give a damn what it sounds like, it just has to look good — or I don't want to see it at all." The *it* in this case, as you will have all guessed, being the sound system.

An Unusual Venue

This preamble is not intended simply to use up valuable space, but rather to set the scene for the problems that faced the installation of a permanent sound system in the Casino concert hall at Montreux, Switzerland, home of the annual Montreux Jazz Festival, held last year in early July. Until 1971, when the old Casino was burnt down during a Frank Zappa concert (and thereafter immortalized by Deep Purple with the song, "Smoke on the Water"), Montreux was a main port of call for touring groups, as well as the venue for the Jazz Festival.

In view of that situation alone, one would have thought that layout of the concert hall for the new Casino would occupy a significant place in the new design. One look at the floor plan (Figure 1) will show that this was not the case. Interior design whims can often be acoustic nightmares — witness the house curve shown in Figure 2 — and although the hall may look pretty enough, it just didn't sound right!

One of the reasons for the hall's peculiar layout was so that it could be divided into three sections by partitions, making for greater "rentability" if the hall was needed for two or three smaller functions on the same day. Acoustic separation was quoted to be 55 dB, although I think someone forgot to specify at what frequency! In

. . . continued overleaf —

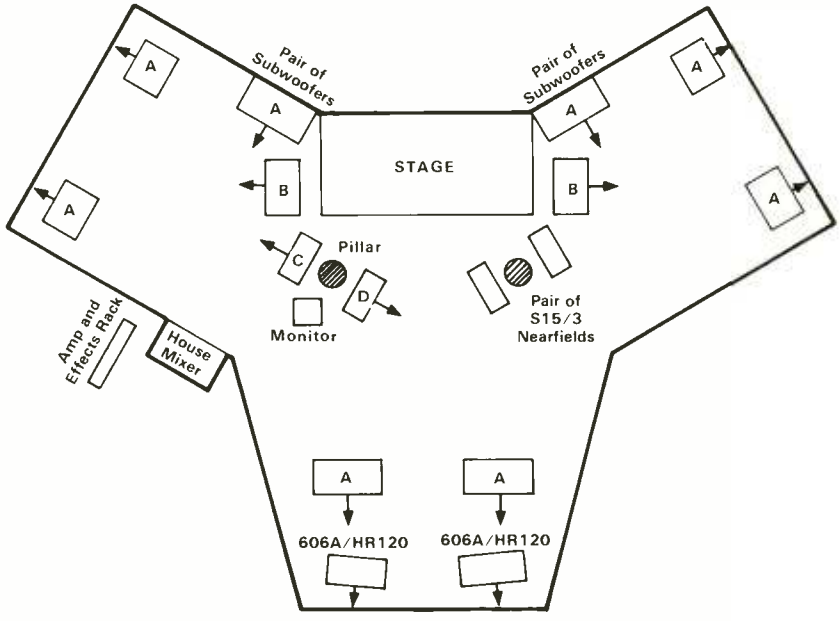

Figure 1: Floor plan of the Montreux Casino sound system. Speaker units 'A' comprise EV TL202S subwoofer cabinets; units 'B' a pair of EV TL606A Theile bass systems, and a single HR9040 mid-range horn; units 'C' a pair of TL606A's, one HR9040, and one HR6040 horn; and units 'D' three TL606A's, plus a HR9040, and a HR6040. Note that the nearfield and main speaker systems mounted on the central pillars have been shown separately for clarity.

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truth, broadband separation was found to be a sad 25 to 30 dB.

So much for the hall, what about the P.A.? This is very much a case of: "Everybody brings their own, and for the Jazz Festival we'll let the distributors of the well-known makes of sound equipment vie with each other for the honor of lending us one! (I kid you not.) The reason for this concept? Economy. After all, why install a house system when people will lend you one or bring their own? Never mind the fact that they may be *totally* unsuited to the venue. After all the thousands of dollars spent on the building and interior decor, the cost of an integrated sound system for the concert hall would hardly have been the straw to break the proverbial camel's back. I am sure that many *R-e/p* readers will have experienced the situation where club owners spend a fortune on decor without a thought, yet will blanch at the idea of paying more than a couple of hundred dollars for a PA system that will be used by high-priced artists, and heard by customers paying equally high ticket prices. Who will, in addition, also complain that "it doesn't sound like the record!"

After the first Casino concerts, and the first complaints, various suggestions to improve the hall were made, including no less than three by Tom Hidley of Sierra/Eastlake. Economy prevailed, however, and none of the projects were taken up. For the 1973 to 1976 concerts a succession of well-known names did their best for the Jazz Festival, but without notable results, and for the 1977 event the organizers turned to the European branch of Electro-Voice, which, by a happy coincidence, also happens to be based in Switzerland.

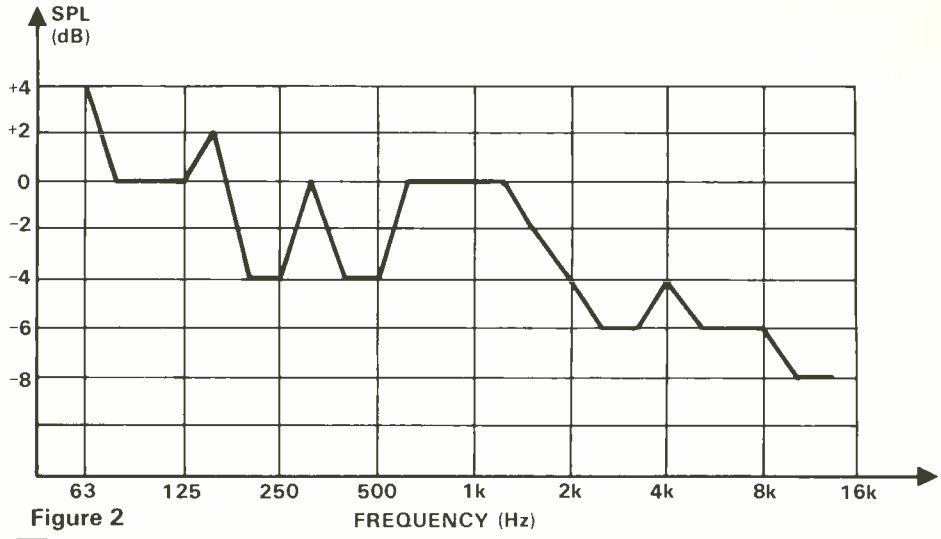


Figure 2

The Montreux Jazz Festival is an event that has not really made up its mind as to its main purpose. Whether the Festival is a series of live concerts, TV shows in front of a paying audience — plus simultaneous radio broadcasting and multitrack recording — or live recording sessions plus TV, has yet to be decided. It is immediately obvious that somewhere along the line there have to be compromises, and that no one party is ever really satisfied.

One of the prevailing problems that made itself felt in the past was the television company's insistence on the PA system being as inconspicuous as possible, together with the organizers not wishing sight lines from the side halls to be obstructed. This meant that conventional PA stacks were *out*, and led to such situations where one year the front row of the audience had the dust blown out of their pants by bass bins, and the mid-section their eardrums by the mid horns and tweeters! Not a very satisfactory state of affairs for patrons paying high prices for concert tickets, especially when they feel that they are often more of a nuisance than anything else. It was obviously time to bring some sanity into the situation.

In accepting installation of a permanent sound system at the Montreux Casino, Electro-Voice also stated its terms. These were that the PA system be designed to work *with the hall* and existing conditions, as it would be the company's reputation being laid on the line. Just try telling an irate customer *why* it doesn't sound as good as it should! Yes, I'm sure that most of you know the feeling. For this reason job was approached in the same way as a normal permanent installation; no arriving one or two days beforehand and hoping for the best.

Sound System Design

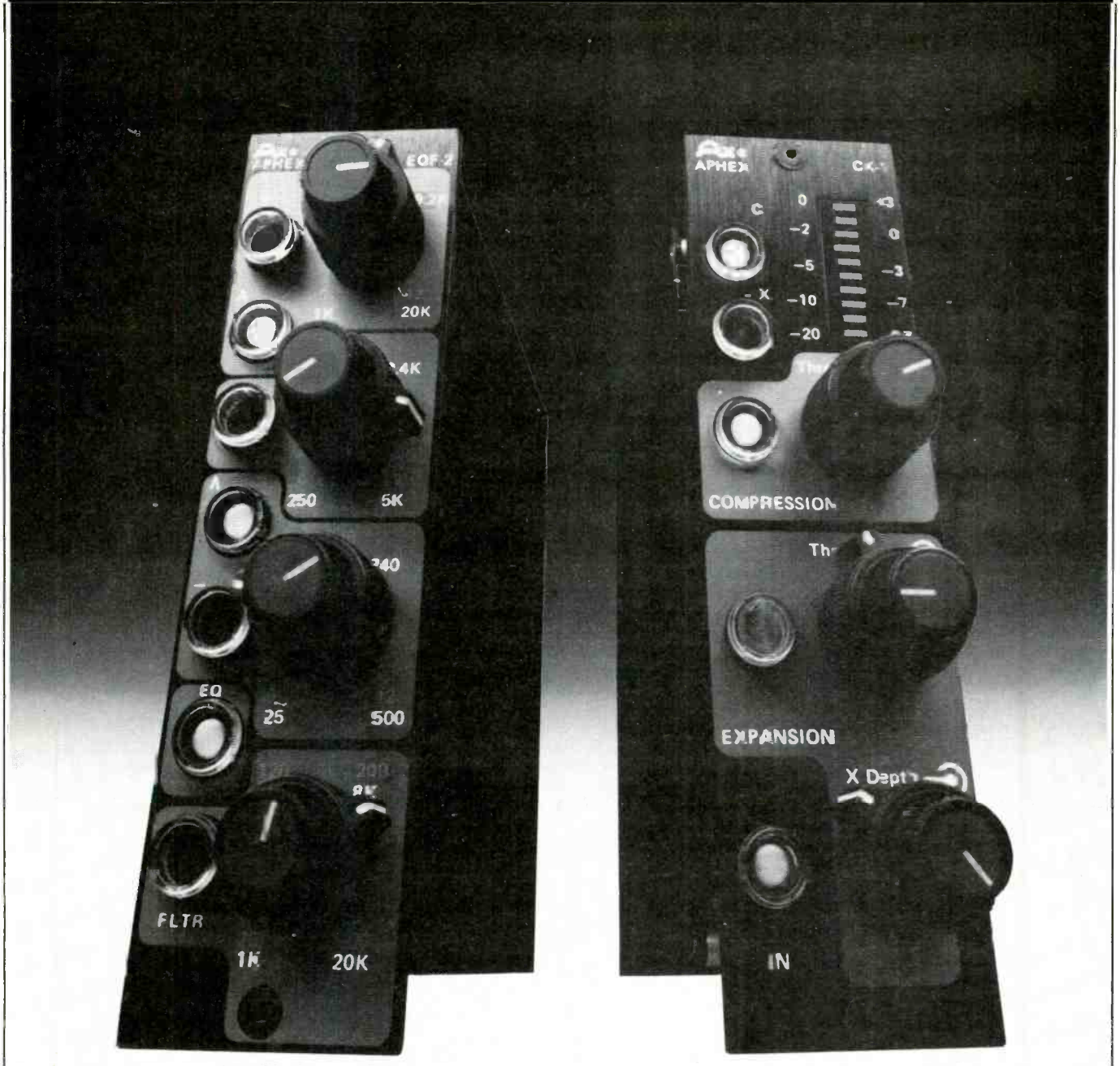
The first step was to get an accurate acoustic analysis of the hall in all the main areas, in order to calculate a system that would give equal coverage broadband to all seats within approximately 1.5 dB, and be as unobtrusive as possible in order to keep most people happy. The present system represents five year's worth of development from the time of the first installation in 1977 to the present day. The main bulk of the equipment was finally bought in 1980, and the main difference in the '81 system was the addition of an experimental subwoofer section to improve low-end response.

From the beginning it was deemed that the only way an acceptable sound would be achieved would be to fly the PA, the main elements being grouped around the two central pillars near the front of the stage (Figure 3). Discounting the subwoofer section the system is basically two-way, with TL606A cabinets handling the bass, and DH 1506 drivers coupled to HR 6040 and HR 9040 horns for mids and trebles. In the

Figure 3: Central clusters on the pillars

— the author —

A musician since 1958, **Terry Nelson** began working PA systems in 1964, with custom installations and stage equipment at Wallace Amplifiers, London. Moving into concert mixing in 1967, he has worked with many "name" groups, and handled front-of-house sound for the 1979 "Woodstock in Europe" tour. In addition, Nelson designed his current studio, Blue Cedar, with considerable help and advice from Tom Hidley. During 1979 he also designed and helped build Studio Maunoir in Geneva, a 24-track facility for composer Thierry Fervant. Last year he started work with the Audiocom 24-track mobile, one album being in mixdown at time of writing. He is currently involved with Blue Cedar and Blue Cedar Productions, with two 45's already out and other productions in preparation.



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past ST350A tweeter horns were also used. However, Electro-Voice finds that the 1506 drivers give a smooth response up to at least 16 kHz and, given the fragility of the tweeters with regard to high-power handling, felt that this is the best solution so far. Having worked personally with the PA system, I did find that the tweeters nevertheless gave that extra "sparkle," and it is possible that a new high-power version currently being worked on may be reinstalled late this year. Time will tell.

As can be seen from the hall floor plan, running the PA in stereo would be somewhat impractical and, for the bulk of the concerts, unnecessary, meaning that at Montreux it's back to mono. Referring to the flow chart shown in Figure 4, the sound system is comprised of three basic systems: the nearfield or audience area immediately in front of the stage; the three hall sections; and the gallery system.

Starting with the nearfield sound system, this consists of four S15-3 full-range cabinets, with two for the center hall and one each side for the two "wings." Formerly, the side halls were served by two S15-3's, but last year two were removed to make room for an unusual stage decor. The system is equalized separately with a White 4001 third-octave, whose main aim in life is to compensate for the dead spots that may occur for listeners near the front of the stage — the main PA mix finding itself a little "forward," due to the positioning of the pillars.

The main PA rig comprises two identical systems for the left and right halls, and a slightly "beefier" one for the center. Each wing is covered by four TL606A Thiele bass cabinets, each containing one EVM 15L speaker, plus two HR9040 and one HR6040 horns for the mids and highs. The center hall is covered by, per side, three TL606A cabinets, and one each HR9040 and HR6040 short- and medium-throw horns. The main system is equalized via a second White 4001, and crossed over at 800 Hz with an Electro-Voice XEQ-1A active crossover, which also provides the correct equalization for the low- and high-end components. This latter feature of the XEQ-1A is particularly useful. Should the prevailing conditions and/or music require it, bass and treble adjustment of the system can be made without upsetting the basic house curve on the graphics. Instant return to the equalized curve can also be achieved with the XEQ-1A.

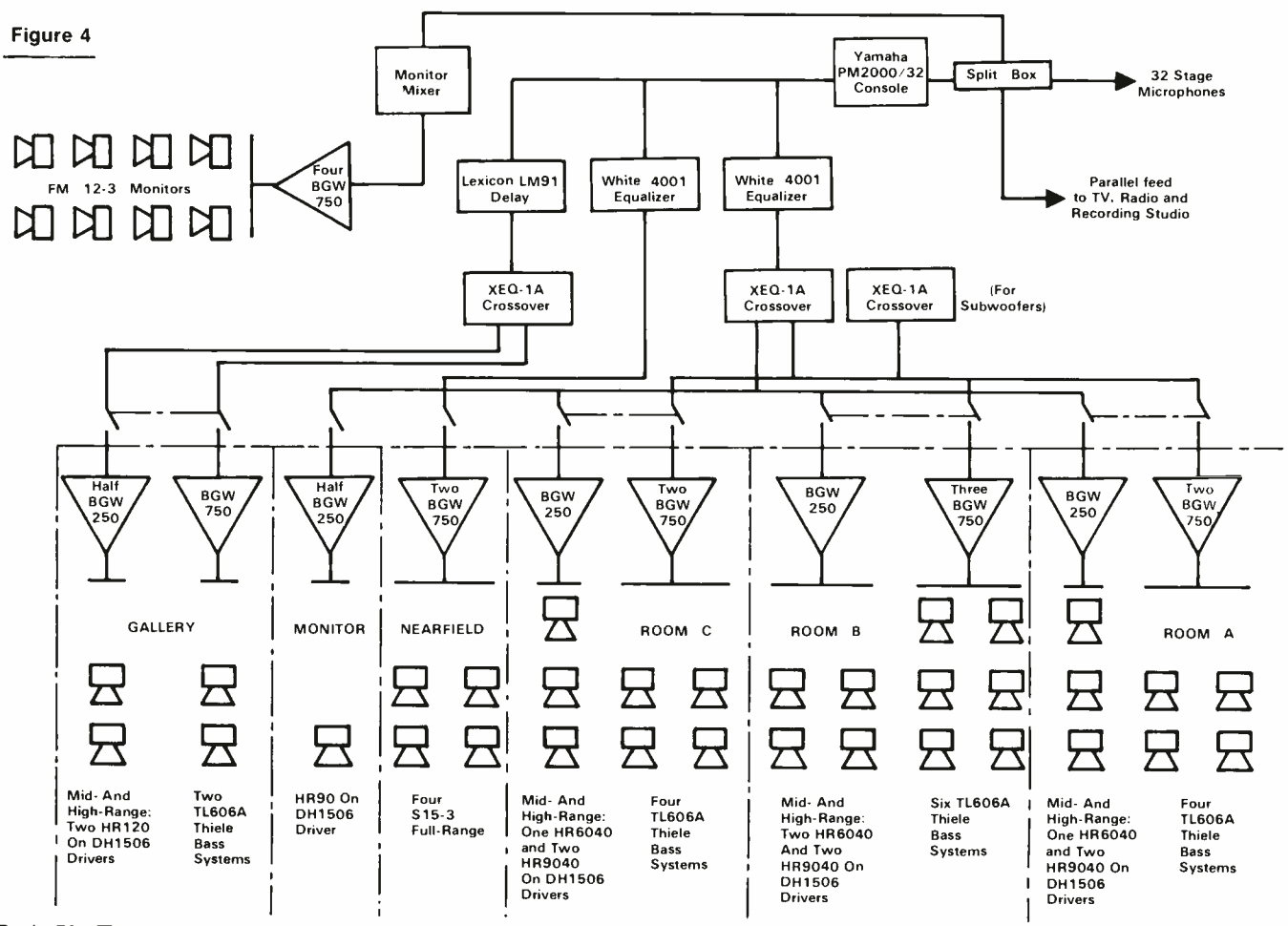
In addition to the hall area, there is also a separate sound system for a gallery located at the rear of the center hall, which is reserved for the press and guests. The main purpose here is to reinforce the house PA, and give just that little bit extra by the way of definition. Split from the main output of the console, the signal passes through a Lexicon LM91 single-channel DDL, and then through its own XEQ-1A active crossover. The output feeds two TL606A cabinets fitted with HR120 horns that are suspended just above ear level several meters in front of the gallery. For the system to be unnoticeable in operation, levels and delay time are set accordingly, the latter being between 42 and 45 milliseconds, depending upon the conditions.

In spite of security precautions in the hall, the well known "little fingers" act manage to get in and twiddle from time to time, with the result that the drummer sometimes appears to be playing in the gallery rather than on the stage — producing rather disconcerting psychological effects! As well as being annoying, this sort of tampering also means that time which could be better used elsewhere, has to be spent rematching levels and delay time.

Subwoofer System

One of the major differences with the previous year's system was the appearance of an experimental subwoofer installation. In the past the more rock-oriented acts commented that there was a certain lack of low-end "punch" to the PA sound, so it was in the cards that something would be done

Figure 4



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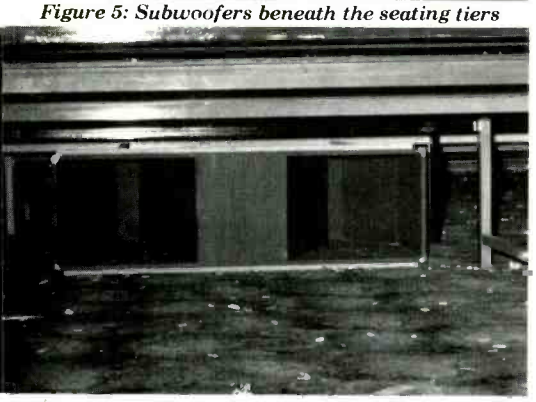
to improve the situation. As may be gathered, the main difficulty facing the use of a subwoofer system was the physical one of where to put it! Conventional positions would have brought howls of protest from the TV crews, and locating the subwoofers under the stage would have upset the studio — as well as the monitor mixer, and possibly the musicians. (The drummer might have liked it, though, as an "ideal" bass drum monitor.)

In fact, under the stage would have been an ideal position, but for the fact that it is a temporary structure, and thus rather light, as well as being a very good low-frequency resonator. However, the audio industry very rarely lets a good idea go to waste. Inspiration came from the direction of Britannia Row Productions, who was placing its subwoofers for the Pink Floyd's "The Wall" system around and underneath the audience area. O.K., the idea sounds good, but how to implement it? All the existing E-V bins suitable for the job were too large to be fitted under the raised seating in the audience area. Since the cabinet under consideration was the TL4050 twin 15-inch, folded horn, the logical step seemed to be to halve it and make a horn for one speaker; the result being the now officially named TL2025.

Figure 5: Subwoofers beneath the seating tiers



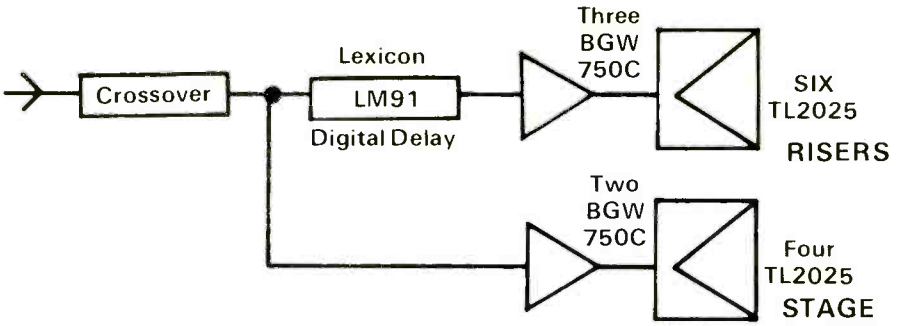


Figure 6: Electronic configuration for the 10 TL202S subwoofer cabinets

Ten subwoofer cabinets were used in all: two being stacked horizontally against the wall each side of the stage; and a pair for each of the three halls, the cabinets again being placed horizontally on the floor beneath the tiers, with the horn mouths facing towards the rear (Figure 5). For better integration with the main PA system, the signal to the riser speakers was delayed by around 38 milliseconds through a second Lexicon LM91 DDL. Although a third XEQ-1A has been installed for use with a subwoofer system, the actual crossover used was a Tapco CPX, which features a continuously variable crossover frequency facility, and thus made experimentation easier than working with fixed crossover points (Figure 6). Best results with the sub-bass were around the 100 Hz point which, unfortunately, coincided with the resonant frequency of the risers, and gave part of the audience an

unexpected vibro massage during several high-energy acts! However, room still exists for experimentation.

On the positive side, the sound system's low-end response is improved immensely by the subwoofers, although at times some engineers were getting just a little *too* over enthusiastic. As a personal comment, I must confess to liking the idea of treating the subwoofers as a separate system, with the crossover point overlapping with the main system, since it allows the sub-bass to be brought in more for effect than as a permanent situation. The TL606's are quite happy down to 55 Hz if need be.

Power for the sound rig is provided by BGW amplifiers throughout, and consists of four 250A's for the horns, and fifteen 750C's for the bass bins and full-range cabinets. The amplifiers and complementary equipment are all housed in well ventilated racks to the side of the mix position, making for easy access for on-the-spot adjustments.

House Mixing Console and Outboard Effects

Sound mixing is carried out on a Yamaha PM-2000 console equipped with 32 input channels and an 8-by-8 output matrix, as shown in Figure 7. (In the case of the Casino system though, only matrix #1 output is used, meaning in essence the PM-2000 operates as a 32/8/1 desk.) Being quite popular in the PA field, the Yamaha PM-2000 needs no introduction, although because the company decided to use fixed frequencies for the EQ instead of sweeps, I do wish that they had chosen more music-related ones! Otherwise, the console does its job very well.

Outboard equipment tends to consist of what visiting engineers may bring in, and this can vary from floor to ceiling rack cabinets and road-cased tape recorders (Spliff Radio Show), to a simple compressor or pair of ears. With Mountain Recording Studio being located in the same building, however, there are usually a couple of UREI 1176 compressor-limiters around, on loan for general limiting duties. One is usually connected across the main output of the desk, to come into operation at around 103 dB SPL. One of the studio's EMT plates is also patched in for additional reverb, and last year Mountain was rather nice about letting the house sound engineers borrow a Publison pitch shifter.

Those of you who have checked out the room layout diagram will have at once noticed that the mix position is far from ideal. Indeed, Montreux must be one of the few concert-sound systems to have a monitor cabinet installed for the house mixing engineer. Since the mix position is not in the direct line of fire of any of the horns, an HR90 cabinet fitted with DH1506 driver has been installed

Figure 7: Yamaha PM-2000 house console.

directly in front of the mixer. The obvious place to mix (if one cannot use the ground floor) would be in the Press Gallery, but this means giving up seats (though the TV crew manage to get two large follow spots set up bang in the center), and it really comes down to a question of priorities.

Monitor Mix Position

Stage monitor mixing is done from the side of the stage to stage right. Mixes are achieved with an Interface Electronics 24/8 monitor board feeding four BGW 750C amplifiers and eight E-V FM12-3 monitor speakers (Figure 8). In addition, for the Jazz Festival there is usually a supplement of FM12-3's, plus any redundant S15-3 cabinets that may be lying around. (The latter are often used as keyboard monitors.) The other extra that is wheeled out for the Festival is the drum monitor — strictly for drummers who are hard of hearing!

This consists of an E-V twin 15-inch, Thiele-tuned bass cabinet with a second cabinet housing an EVM15L and "white horn" (as the horn from the Eliminator V is known), driven by a DH1506. The system is bi-amped with one channel of a BGW 750C for the bass, and one channel of a Crown D150A amp for the mids and trebles. As might be

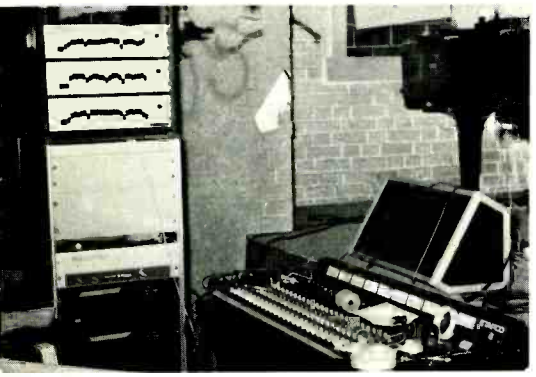


Figure 8: Interface monitor mixer and associated amplifier and equalizer rack.

expected, crossover unit is an XEQ-1A. (In case you are wondering, the other channel of the Crown is used for the monitor mixer's own monitor cabinet, used to check the different onstage mixes.)

Needless to say, the drum monitor setup can get *very* loud if necessary, and I must admit that I wouldn't like to have it 18 inches from my ears, as some drummers do. With a program as varied as that found at Montreux, monitor requirements can — and do — differ very widely, often during the same evening. These can vary from no monitors at all to "everything you've got — and more if you've got it!" with SPL's peaking up between 125 to 130 dB. (So who needs a front-of-house PA system?)

Operating Crew

Now that we have an idea of the system, what about the people who run it? Since the 1977 Festival the crew has remained pretty much the same, with Tom Durell from Disney on house, and Chris "Snoopy" Penycate on monitors. As Tom had other things to do for Disney while in Europe last year, his colleague Greg Watkins was in charge of house PA for most of the time. (Except when he was replaced by this writer for two days so Greg could be a tourist.)

From the stage crew's point of view, things have improved immensely over the last years with the appearance of a professional stage manager in the shape of Alex Higgins. Higgins and his crew, who are mostly of Anglo-Saxon origins, manage to get together each year for the Festival. Changeovers have been cut to the bare minimum, and it is very rare for the musicians to be waiting on the crew — the other way around . . . yes! Working hours are long, often beginning early in the morning with sound checks for the evening's program, and going right on through to the end of the concert with just a pause before the start of the evening's proceedings.

One of the problems facing any festival is the length of the program, and Montreux is no exception. No matter how high the quality of the content, a program that is too long is, quite simply, too long! Special guests and "surprises" do work sometimes, but more often than not are an indulgence for the artists, and a pain in the backside for the audience. This is especially true, I feel, when the concerts are held indoors, and the conditions of comfort may leave something to be desired. (At least outdoors one can get up and walk about!) Though things have improved a lot over the last years (due in no small part to the efficiency of the crew) one is left with the impression that a little more blue pencil could be used in planning programs. After all, you're *not* really in a state to fully appreciate a duo of John McLaughlin and Chick Corea playing a very special kind of music, after over four and a half hours of concert time.

It has often been noted that one of the prime qualities to be an audio engineer is the ability to get on with people, and this has certainly proved true at Montreux. While the sound system crew are in charge, they are often obliged to concede their places to the personal engineers of whatever act may be appearing. Such people can range from experienced professionals, to unenlightened amateurs who claim they know it all. Add to this the sometimes hostile treatment from personal managers, and the situation can call for a lot of patience. As has often been noted, the more professional the contacts, the less hassle there is and, in the majority of instances, this is the case at the Montreux Casino. However, it has been known for the sound crew to be refused stage passes for "important" acts. Paraphrasing a well known touring axiom, this became "No pass; No sound," and the problem was quickly solved.

More usually difficulties arise when people try running the system while not being familiar with it, instead of giving cues to the resident house or monitor engineer. Such behavior translates into the system sounding lousy, or not powerful enough, or whatever else excuse is available at the moment, when the real fault lies in the *way that the system is being used.* One of the sad facts of life at Montreux is that it is not a rock and roll venue, and as such high energy — in this case watts and SPL's — performances are just not on. Though one sympathizes with performers used to playing loud, an effort often has to be made to reduce on-stage levels in order for a workable balance to be made over the PA sound. Which is not to say that the PA could not counter-balance, as it will produce 130 dB SPL nicely, but the problem would be no audience left to listen to it!

In a normal configuration one could say that if the on-stage instrument amps are filling the hall, then leave them to it and reinforce the rest. At Montreux however, there is the additional problem of what to do for the side halls. While fully capable of handling separate mixes for center and sides — and, indeed, this would be the ideal configuration — the time needed for this sort of sophistication would just

Figure 9: Typical stage layout and miking for big-band performances. Figure 10: Microphone placement and general stage layout for Rock, Fusion, and Jazz-Rock acts playing the Montreux Casino.

Figure 9

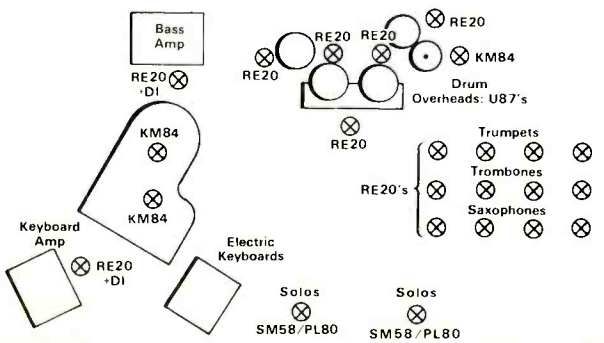
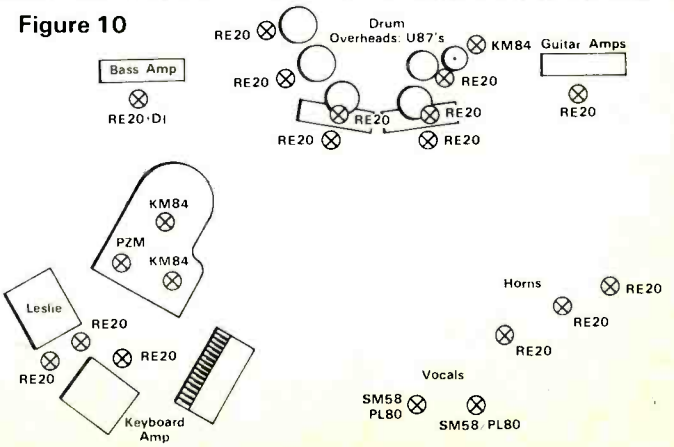


Figure 10



not be available.

Nevertheless, in spite of the difficulties the system does sound *good*. The smoothness of the E-V HR Series horns is very striking, and the absence of "honk" a welcome change. Though possibly not as punchy as the average rock 'n' roll PA rig, the Casino system is fully capable of giving a good account of most types of acts. It's simply a matter of having to become familiar with the hall and the system — which would seem to me to be good concert-sound practice. Also, with the introduction of the subwoofer system, even the most hardened rock and roller should find enough bottom end.

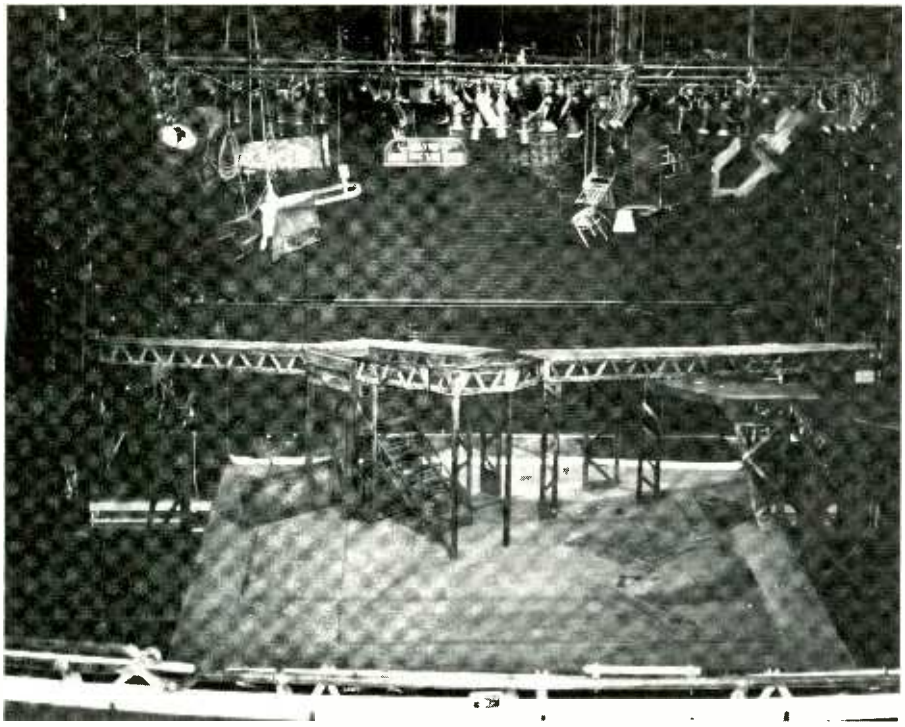
Collaboration is also very necessary between the PA engineering crew and Mountain Recording Studio, the latter under the auspices of Dave Richards; it is Dave who decides which microphones will be used on stage, and where. From the point of view of recording live albums, this makes a certain amount of sense, but at the same time is not without its problems. While certain mikes are fine for the studio, they are not very well suited to stage conditions, with high ambient levels plus the omnipresent risk of feedback (the latter especially for monitor mixes).

To avoid a forest of microphone stands on stage, the same mikes are used for the studio, radio and TV mixes, and the PA system. (With the exception of the radio feed, which is a simultaneous stereo mix from the studio, everyone derives their own mix from parallel splits of the stage mikes.) On electric instruments a fair sprinkling of D.I. boxes is used, so often the house mixer can get around most problems; typical mike layouts for a big band and rock concert are shown in Figures 9 and 10. However, if all of the 32 mike lines are not in use, it is not unknown for extra microphones to be put on just for the PA. As has already been pointed out, with so many situations going on at the same time, there have to be compromises, and it is rare for there to be any friction among the people involved in the audio side of the Festival. (Also in the spirit of compromise Snoopy, the monitor mixer, solved the problem of the television crews continually covering up and obscuring the EV labels on the cabinets by branding one of the cameras for the rest of the Festival.)

* * *

As can be seen, Montreux Jazz is an event requiring a high degree of teamwork from those involved. In spite of the same moans and groans each year, every Festival still sees the same faces who have come "to see what it'll be like this time!" Recording, PA, TV and radio; which is the most important? That sounds like the $64,000 question. Whatever the answer, everyone involved in handling sound for the Festival does it all very well. □□□

Set for "Jungle Of Cities". Speakers are purposely masked from the audiences view.



AUDIO for THEATRICAL PRESENTATION

by Regge Life

Amid the controversy that abounds within the legitimate theater, regarding the extent to which sound is being reinforced, mixed and reproduced for the audience, it is my hope that this article will provide realistic and cost-effective solutions to some inherent problems.

In designing sound for the theater, one is faced with providing reinforcement not only for an orchestra, but also for the human voice, without access to ideal mike placement in the orchestra or on stage. The system must be whisper quiet and, in most cases, hidden from the audience. Miking is either incidental, or by means of a wireless system. Problems of feedback generally occur in situations where incident miking is used in an effort to keep voices above the orchestra. Most outboard devices used in concert-sound reinforcement can be utilized to minimize this feedback when incident miking is the preferred means of reinforcement. But, unfortunately, in addition there are problems of sound loss when performers move across the apron of a stage, and a "tunnel-like" effect in an incident miking situation. As a result, wireless systems have emerged as the new wave of theater sound reinforcement. Although a wireless microphone enables the human voice to be kept discretely above the orchestra, it is the acoustical character of the wireless that has led to some restricted use. Also, problems constantly develop when using multiples of wireless microphones.

The frequency ranges encountered in the theater start around 800 Hz with the string bass in the pit, to well above 10 kHz on percussive and electric instruments. The transient differences in the vocals of most musical theater presentations create an equal amount of havoc for the designer. Traditionally, theatrical songs are composed in a far more emotional and eclectic style than pop songs, and this leads to wide differences from verse to verse and the need to constantly ride gain. It is generally a big job, and each system has to be tailored to the theater being served.

Since producers wish to provide audiences with top-quality sound, an audio engineer or technician may soon find himself being asked to design a sound system for use in a theater. Because of the nature of the new Broadway musical, there is already a need for professionals with experience of recording and reinforcement. The wireless microphone opened the door to serious considerations about reinforcement, imaging and reproduction, and what has followed is rooted in multitrack equipment and a multitude of outboard effects. The theater space is not unlike and sometimes identical to a concert hall, but with its own particular set of problems.

In a recent *New York Times* article, entitled "The Surrender of Broadway to Amplified Sounds," author Harold Schonberg came down very strongly against the use of amplification. "The technology is wonderful, but the sound is generally awful," he remarked. Much of the article's focus concerned the current production of *The Pirates of Penzance*, a Gilbert and Sullivan revival at the Uris Theater, New York, and which recently won a Tony Award. The producer, Joseph Papp, and designer, Don Kettler, chose what is basically a rock reinforcement system. Schonberg didn't like to see a classic updated in this fashion, and the article aired his displeasure.

Reaction from theater sound designers to Schonberg's article ranged from hostile to supportive. In my opinion, Mr. Schonberg, a classical music critic, prefers a "traditional" approach to classic work, and likewise the sound should be a part of that concept. Some of his observations, however, were very realistic in noting that most of our musical talents do not have the power to fill the larger theaters without reinforcement.

Sound design for the current production of *The Pirates of Penzance* is a choice of both producer and sound designer, and stands without explanation. This point will be pursued in greater depth later in the article, but nevertheless the question remains: How does one design a system for the theatre

— the author —

Regge Life is president of Lifecycle Productions Inc, a full service audio and video company. Beginning as a musician, he performed with many acts, notably Patti LaBelle (back in the days of Patti LaBelle and the Blue Bells), Hugh Masekela, and the Nu-Band (now A&M artists Atlantic Starr). He began engineering with Oscar Brown Jr.'s musical, *Joy*, in 1970, and ran the show until it closed. His work for Lifecycle this year includes acoustic design for the Brooklyn Academy of Music's Playhouse Theater, and extensive work with the Negro Ensemble company. In recent seasons, Lifecycle has served the following New York theater companies: The Chelsea Westside, The Whole Theater Company, Charles Ludlum's Ridiculous Theater Company, The Lion Theater Company, INTAR, The Women's Interart Theater, and the Frederick Douglass Creative Arts Center.

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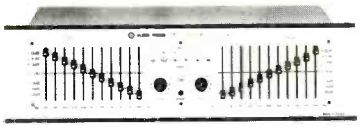

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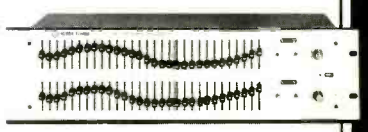

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DESIGN FOR THE BROADWAY-SIZE THEATER
The Brooklyn Academy of Music

I was contacted in September 1980 by the Brooklyn Academy of Music Theater Company, New York, (better known as BAM), as the company was about to begin its second season. Staffed with an impressive line-up of scenic and lighting designers, BAM sought to complete the package with high-quality sound. The Academy is the oldest performing arts center in America (established 1859), and their main stage was the playhouse space that seated 1,100. A renovation had been carried out to accommodate the theater company, which included raking — setting the stage at an angle of between 20 and 25 degrees — and thrusting or extending the stage beyond the proscenium arch. It successfully created a greater intimacy with the audience, and aided the actors' projection in a large house.

While walking on the stage area, I concentrated on all the sounds that could be heard. A sound designer has to trust his ears above all else; after all, actors and audiences, don't show up with test equipment.

The designer should test the projection and reverberant quality of the stage area. Have an actor (or in my case the production supervisor) stand approximately center stage and speak in a normal but projected voice (what I mean is don't shout!). Stand at all four boundaries upstage and downstage (upstage being the area farthest from the audience), and listen to the ambient and reverberant quality of the voice. Then move from one area and, while listening, note any changes. You should then have the actor change position and project from each boundary, while you maintain a stationary center position. In addition, both of you should try listening to each other speak from every possible combination of areas.

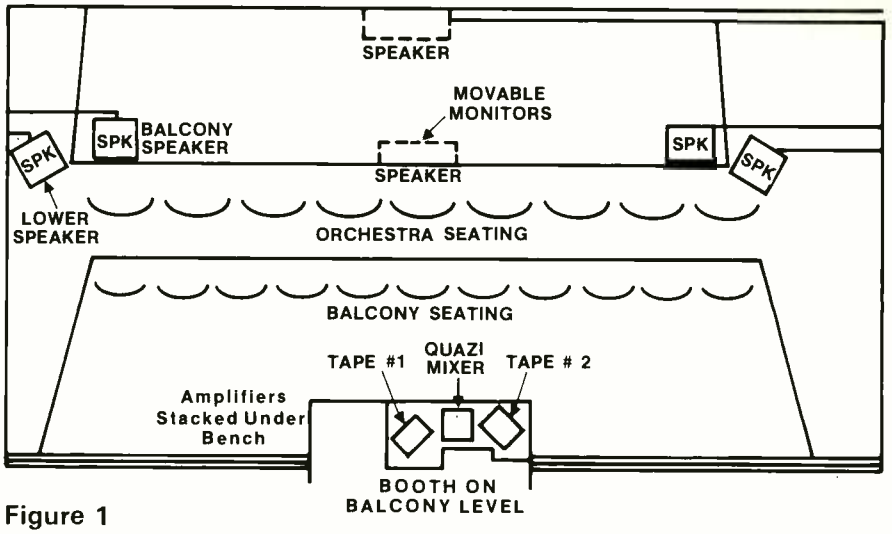


Figure 1

Particular attention was paid at BAM to the amount of resonance retained in the voice, how much reverberation was noticeable, and how well one could hear when more than one person spoke on stage. Standing on stage the sound in the theater was fairly live, and it was reasonable to suppose that once an audience occupied the now empty seats the reverberant quality would diminish.

At the orchestra level the sound was sharp and clear from any source near the apron of the stage, but as the source got farther upstage, reverberation built up due to the theater's enormous fly space above the stage area. The raked stage would aid my effort to project the sound from the upstage positions.

Even though the reverberant nature of the sound from upstage was not objectionable, nevertheless some miking needed to be considered to compensate for this problem.

The next section to be checked was the theater's most dead area: under the balcony. But here again, the renovation had worked wonders. The thrust of the stage, assisted by the angle of the rake, allowed actors to move downstage near the apron, and actually to be located at a lower angle to the audience than was previously possible with the proscenium arch. So, despite the expected loss in projection, the sound wasn't that discouraging. In any event, these aspects were noted for my sound plot and, if necessary, reinforcement speakers could be installed on delay lines to compensate for the sound loss.

Finally, the balcony seating was checked and, surprisingly, it was discovered that a good majority of the seating space was to be found in the balcony. Reverberation was most evident in this area, since the sound reaching to the balcony would leak into the fly space above the stage. Taking into account the fact that an audience would cut down reverberation — literally soak it up — and that actors would have a good deal better projection than the production supervisor helping out with these particular tests, it was

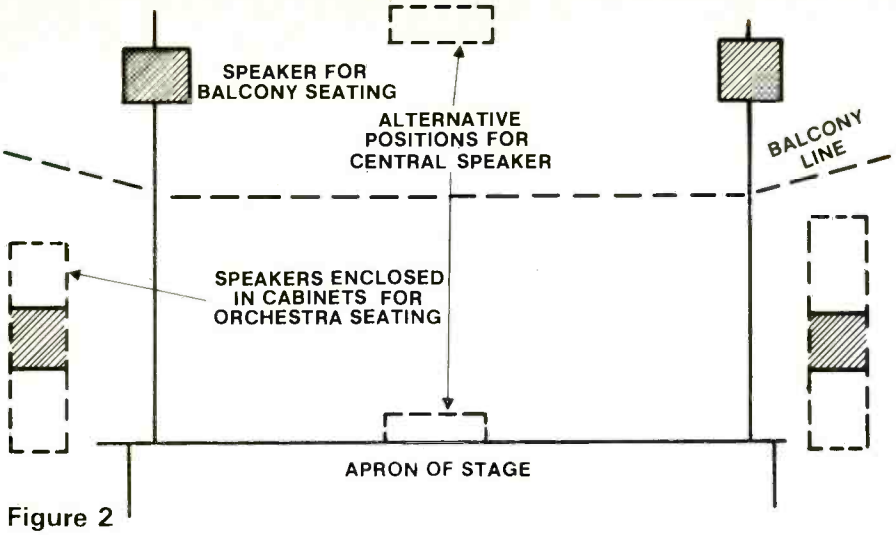


Figure 2

decided to trust the acoustics of the playhouse.

The theater had been built before the advent of microphones and, as a result, was designed to project non-amplified sounds. Consequently, there was a need to complement the natural acoustics of the hall, and not neglect any area of the audience. On the other hand, changes that needed to be made had to be subtle. For the most part, the theater company's plans called for a classical and a serious season. Therefore, the sound technology should maintain a low profile, but accentuate the theater's natural acoustical quality. The theater had to be filled with sound, but this needs to be done as close to natural projection as possible. Sound for the theater should capture the audience; enabling them to sense it viscerally as they engage themselves in the play.

The sound system employed the year before had been ruled inadequate because of three reasons: it possessed insufficient power to fill the theater; in our opinion it wasn't up to professional specifications; and it was also limited in the flexibility it offered. Judging from the system's equipment list, there already existed a useful guide to help us avoid making the same mistakes as the previous sound designer. Aspects that needed to be taken into account included selective miking in the upstage areas, as well as the possibility of an additional pair of speakers under the balcony on delay lines. My bid was submitted and, shortly after, approved. BAM commissioned Lifecycle Productions to design a system for the entire season.

Varied Repertoire

The BAM repertoire was to include a varied season from Shakespeare to Brecht. Three different composers were contracted to score music for the season, and their styles had as much contrast as the range of works to be performed. In addition, some re-budgeting had to be made, which eliminated the upstage reinforcement microphones. Also, the proposed system had to operate with the existing rack of amplifiers, since the theater found that it could not afford to buy or rent new ones.

The first step was to critically test the hall's acoustics. An environment had to be created that to a professional sound designer would suggest the room had been "tuned." Working actively with various consultants, Jim Taylor of The Taylor Shop, New York, was called in to test the playhouse. Providing a clear assessment of the room's acoustical quality, Taylor's tests with pink noise, reverberation-time measurements, and sound isolation showed that the hall

was extremely accurate, except for a dead area under the balcony in the orchestra seats. Our analysis of the balcony showed that a nearly identical sound character to that experienced in the orchestra seating could be achieved if a second set of speakers were hung just for the balcony area. At this point, we were proposing that three separate sets of speakers be provided for the system, but it was considered better having enough to be completely covered.

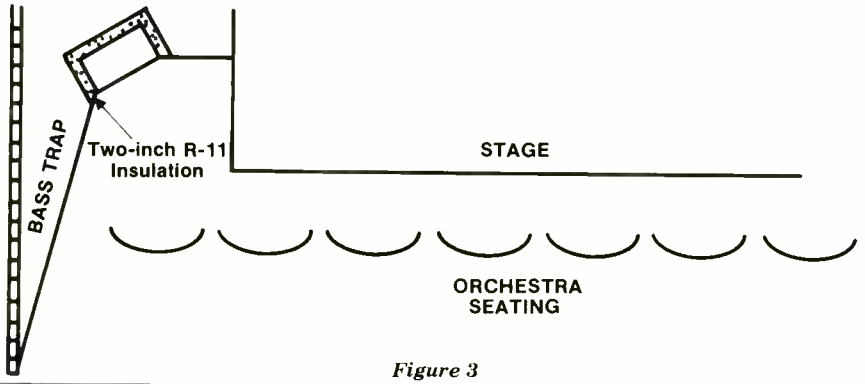


Figure 3

Equipment and Installation

After the test, Bob Bender of Masque Sound, a New York sound shop, was consulted on various types of equipment that could be acquired through purchase or rental. Working with spec sheets for all of the many options, our final choice was Otari ARS 1000 7½/15 IPS, replay-only tape machines, and Altec 9849 loudspeakers fitted with 414-8C 12-inch bass drivers and 32B horns. The Otari ARS 1000 is an excellent machine that features motion-sensing controls and 10½-inch reel capacity. It is equipped with a sensor that was lifted to prevent the tape from stopping unnecessarily. Being an instant-start machine, cueing presents few problems, and the half-track format reproduces studio master tapes to the fullest. The Altecs had been used previously on another job (see later section on *Star Treatment*), and they were found to be both accurate and efficient; in addition, they possess an extremely identical nature when used in multiple-pair applications. Power was supplied by two Bozak 80-watt stereo amplifiers for the main system, and a third Bozak 40-watt mono amplifier for the on-stage cue monitoring. The amps were fairly old and had not been worked on in quite a while, but with some careful alignments of the amplifiers' internal presets, they proved somewhat quieter.

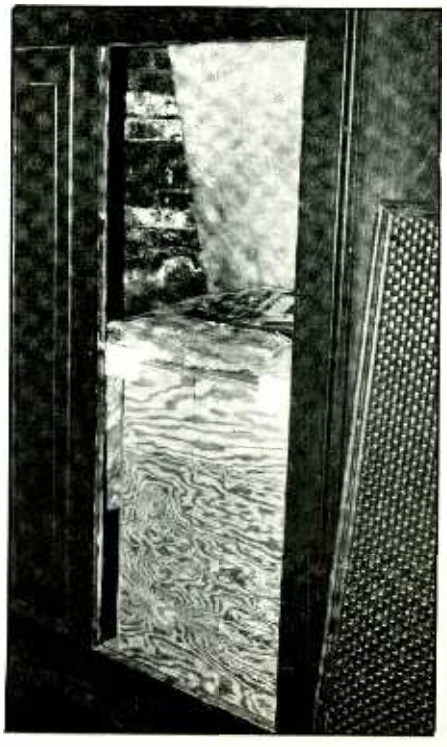
Choice of mixing board was a difficult decision to make, and is where the need for a cost-effective console comes into play. Too many of the boards that I was accustomed to in smaller theater operation would not be suitable for the BAM installation, and the use of a larger multitrack console would not be cost-effective. Due to the fact that there was to be on-stage singing during two of the productions, cue monitoring became a critical factor. Unfortunately, the majority of mixing boards made for the theater are designed simply to play back pre-recorded information.

Faced with the dilemma of too much or too little, a choice was made somewhere in between: the Allen and Heath 8-in/4-out Quazi mixer. The board was heavily modified to provide direct outputs from every channel, and far more flexibility than its original design possessed. The Quazi mixer offers four main program outputs and two separate cue sends.

The selected equipment was my personal preference for clean reliable sound within the available budget. Alternative mixing boards equally suitable to theater-sound applications include the Yamaha PM-1000 and -2000, and Midas TR-Series.

A multi-cable had been installed previously at the theater, and provided sufficient lines for program and monitoring. There were some initial problems with this cable because

Figure 4: Detail of speaker box showing air cavity and insulation.



communications were also being run through it, but they were overcome to produce satisfactory results. The process was one of trial and error: we tried different combinations of speakers and monitor lines, until a combination with the lowest amount of noise and hum was found. Ideally, it is advisable to keep all intercom and communication channels separate from program and monitor lines. Cable runs in most theaters are relatively consistent with cable runs used for concert sound; BAM had a distance over 200 feet from the booth to the stage. Figures 1 and 2 illustrate the theater's sound set-up.

Speaker Placement and Cabling

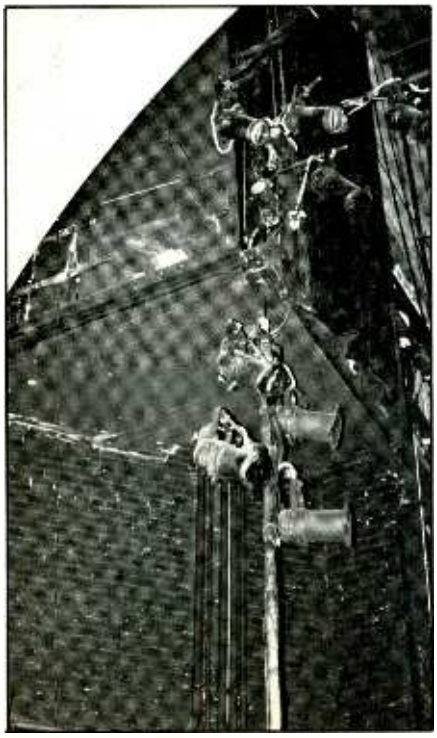
The hall was designed to house two speakers at orchestra level, mounted in recessed boxes stage right and left. Beautifully finished in oak, the wooden facade surround extended into the audience. An air cavity was left between the brick and the facade. Utilizing these existing positions with the air cavity proved advantageous, and the resultant sound appeared omnipresent. Because of the bass trapping, the natural wood facade also aided in frequency separation and clarity, and reduced the need for large amounts of equalization (Figures 3 and 4).

The speakers were installed in an innovative fashion. Support boxes were built to raise the speakers in the cabinet themselves. Using the original factory packing and some Owens-Corning R-11 insulation, the sides and the top of the speakers were packed to reduce resonance, the reason being that the 9849 is a much stronger speaker, and heavier on low-end response. The speakers were also raised slightly to compensate for the dead area underneath the balcony. This was a gamble which we hoped would work, since the use of delay lines can sometimes get tricky. Monitoring the sound beneath the balcony, we discovered that the gamble had paid off, and the sound perspective was accurate for any seating position. The natural acoustical ambience of the hall compensated for any small amount of delay.

Our analysis that called for mounting a second pair of Altec 9849's for the balcony seating area was also checked. Speakers were suspended 35 feet above the apron line of the stage, and positioned 40 feet apart. The pair of balcony 9849 speakers had been dressed for cosmetic appearance, and for another reason that will be covered later in the article. During the test, it was found that the balcony sound duplicated the imaging of the speakers in the orchestra. Some slight adjustments needed to be made to the crossover fitted to the back of each speaker. Altec 9849's tend to carry bass well, but are prone to clipping on the highs. Having achieved the desired sound, they were angled slightly and aimed more towards the center to achieve a true balance. Figures 5 and 6 further illustrate the mounting and placement technique.

The system proved to be functional

Figure 5: Stage right speaker hung from balcony to insure imaging with stage left.



for the entire season, which raises the point that when designing for the theater you must take into account a complete range of music. Different types of equipment will be better suited for particular musical forms, but it is not wise to limit yourself. A board that facilitates the cleanest and most efficient mixing is ultimately the best choice. Be especially careful not to get pulled into buying PA-style sound reinforcement gear. Many of these systems are simply too noisy and ill suited to acoustic or classical music, and you can be certain that there will be music of this sort at some point in the season. Meeting with each composer prior to installation is the best idea.

Keep in mind that the sound system must be like the modern studio: able to accurately reproduce any mode of music. In the case of BAM we only experienced some difficulty on *Jungle of Cities,* a play set in industrial Chicago, and which was the final show of the season. Since the director insisted on using the on-stage monitors for 90% of the sound reinforcement, the monitor system had to be repositioned, and its power-handing capability increased to facilitate his artistic choice.

System Adjustments

Prior to the opening of each show, during technical rehearsals, subtle changes were made in equalization to suit each passage or cue. Since we wanted to achieve a natural hall ambience and overall sound, in this case EQ was used only for small adjustments to suit the individual tastes of each composer: a 4 dB boost in the midrange, which is fixed at 700 Hz, and extending the 10 kHz range approximately 2 dB for woodblocks and other percussive instruments.

Because the ability existed to separate orchestra program from the balcony program mix, some small adjustments in gain were made — along with midrange EQ adjustments to balance the balcony with the orchestra seating.

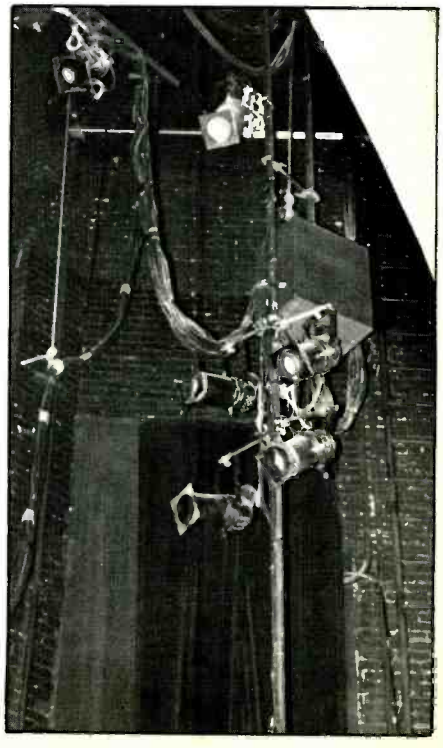
The system proved easy to use, mainly because of the fact that all the power amplifiers had been properly aligned and adjusted at the start. A whisper-quiet system is *mandatory* in the theater, and without proper alignment the noise level would have been unbearable. You can't even afford a hiss.

In fact, this aspect completes the point made earlier about cosmetically treating each speaker that can be seen by the audience. If the audience is unaware of speaker positioning, your work to achieve proper imaging will be aided. In too many cases, sound is incorrectly imaged and this disturbs the audience. The sound surrounded the audience at BAM, however, and was never obtrusive or over amplified. The hall architecture — in the style of grand old theaters — was perfect for our purposes.

DESIGN FOR THE SMALL THEATER OR CABARET

Whereas sound design for the BAM

Figure 6: Stage left speaker, along with lighting, mounted on pipe.



Theater Company was applicable to the standard Broadway-size house, recent changes in the musical theater format, including cabaret, have created a new set of problems for the sound designer. One should aim to make use of the technology at hand to boost the musical perspective, in order to provide ample reinforcement without going overboard. Two current Broadway productions further illustrate this point: the previously mentioned *Pirates of Penzance*, and the tribute to Duke Ellington, *Sophisticated Ladies*. Both of these productions are musicals in which the emphasis is strictly on the music; in the case of *Sophisticated Ladies*, the book was discarded completely. Such shows are run more like a concert, and the demand is to make everything sound like today's stereo recordings. This has become the model for the current Broadway musical and, as illustrated, it becomes a challenge for all of those involved.

In the design for *Star Treatment* by Jack Heifner (author of the comedy play *Vanities*), we were faced with a combination of a straight play accompanied with music written by Janis Ian. The set, which turned on a 270 degree revolving stage (Figure 7), was designed and work plans drawn up prior to my being commissioned to design the sound. Construction didn't allow for a space at the core of the revolve to circulate cable. Faced with this dilemma, it became apparent that connections to the microphones would have to be made another way. Having met with the set designer, some ideas were presented for running cable along the walls. The set designer was able to make some last minute changes in materials to support the weight of the cabling, and a method devised that would allow the cables to wrap gently when the straight part of the play (set in an apartment) was being performed, and unravel in the correct positions for the cabaret section.

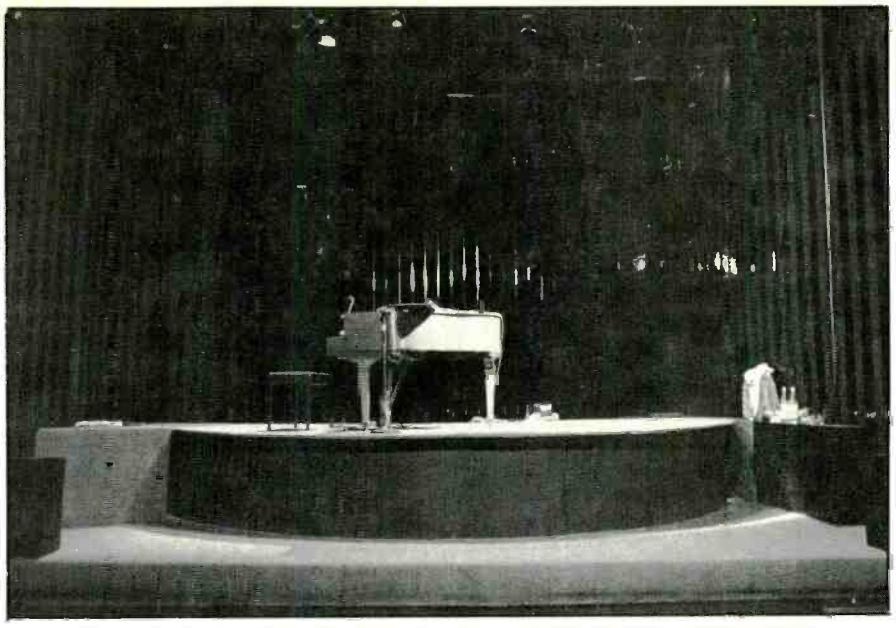


Figure 7: Set for "Star Treatment." Note revolve platform and speakers laid like furniture on stage right and left.

First, the cables were tightly wrapped from the booth, and then run carefully along the tops of the set's false walls. Cables were positioned through glides about 4 inches wide, and given slack once they reached the stage. Finally, a

ring followed by smaller glides enabled the cables to separate in position without tangling (Figures 8 and 9).

Ideally, the cables should have been run as part of the revolve. Also, a larger core pivot would have permitted all cables (or your multi-cable) to be installed without interference to the revolve.

Equipment

The house was typical of off-Broadway and cabaret theaters: with less than 300 seats they are extremely "intimate." Careful reinforcement was deemed necessary to prevent over-amplification of the proceedings. Janis Ian's music was slightly folksy, and required only a piano and two vocalists. If a lot of reinforcement had been specified, on-stage monitors would have been necessary but, unlike BAM, the producers were tightly budgeted. As a result, it was decided to take it easy on the reinforcement, and try as much as possible to achieve a good mix for Janis' music.

Regarding choice of microphones, the sound design called for Shure SM-58's for vocals: one for the show's lead, June Gable; and another for the pianist, who sang and was also an actor in the play. The first microphone tried on the Yamaha Baby Grand piano was a Crown PZM. Although the PZM has been used with great success for incident miking, in this application of reinforced sound it was felt that its use detracted from the piano's acoustical nature, because of the need to work in close to the instrument. A Shure SM-53 was tried next, wrapped in sound absorbent material. With the lid of the piano on the short stick, the SM-53 was positioned about 6 inches above the hammers, directly over the strings and slightly off-center. The resultant sound — softer and less defined — was deemed to be much more consistent with the design concept. By the way, all mikes were shock mounted on their stand mountings.

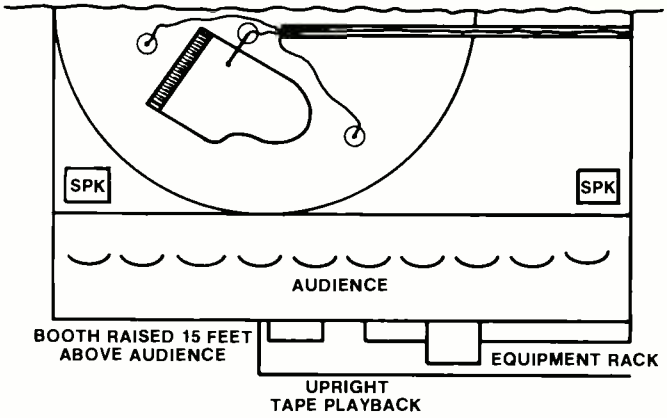


Figure 9

The cabaret atmosphere was a fortunate plus in the design, because it helped eliminate the need for special miking. Theater sound as we know it today relies heavily on miking. In musicals, there is an abundance of wireless microphone systems, and even straight plays are utilizing incident miking. As far as wireless systems go, it's a matter of taste and the environment you have to control. Unfortunately, too, no system is foolproof. I'm not too fond of using wireless in the theater; I find the absence of true sound image, and non-directionality of the close-miking technique due to the use of wireless systems, to be generally offensive. True imaging would almost demand that there be speakers clustered cross the entire proscenium, or from the apron. Due to the fact that wireless pick-up almost suppresses any ambient quality in the voice, you are still faced with the problem of using equalization or some outboard to make the voice sound as natural as possible. In my opinion, there still is no one ideal system, and designers and engineers are working in a compromising situation.

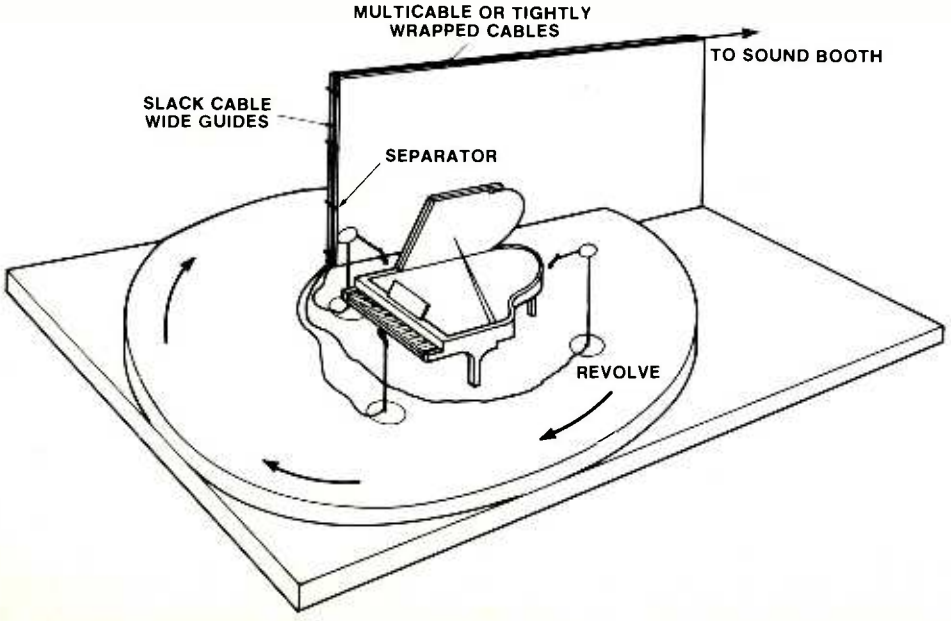
If mikes *are* necessary, my compromise is to use incident miking. In short, a high-quality microphone (condenser or electret-condenser) is placed in a "mouse" — a bed of sound-absorbent material that allows the mike to lie onstage. Placed on the apron or any other part of the stage, the mike is directed toward the sound source. To overcome the problem of imaging when the resultant sound is reproduced over speakers, a nice trick is to use a pair of PZM mikes mounted back to back on a piece of absorbent foam. (For further details on various applications of PZM's for stage and studio environments, see Steve Barker's article, *Recording and Sound Reinforcement Applications of Pressure Zone Microphones, R-e/p February 1981 issue, page 80 — Ed.)*

This particular method solves some problems of imaging, and ultimately allows you to use less reinforcement at each individual mike. The use of one mike per area requires the engineer to run each mike so high that a performer seems to drift in and out of reinforcement. Judge for yourself as you monitor the show, but don't give up until you get it to sound right.

By using the vocal mikes for incident miking during cabaret sequences, a simple Shure M-67 mixer, coupled with an add-on Tone Controller unit for EQ, was found to be adequate for mixing. The entire system was driven by an Altec 1569 80-watt tube amplifier and, as was previously mentioned, the speakers were Altec 9849's; their high effeciency and ability to take the punch of live reinforcement is not easily rivalled.

The show was run hot, tending to punch up the musical numbers that greatly aided the show's pacing. Very little equalization was used, with the result that the live sound never felt *too* reinforced. The theater's brick walls provided a crispness, and the industrial carpeting in the seating area tended to attenuate the high-end. While I don't usually advocate making a show sound louder to make it better, in this case it did manage to keep things lively and involve the audience.

Figure 8



CONCLUSION

Within both of these illustrations of design concept, it's important to gauge the equipment — simple or state-of-the-art — to the theater. *Star Treatment* is a good example of the use of simple reinforcement, whereas shows like *Sophisticated Ladies* and the *Pirates of Penzance* represent complex reinforcement. The designer must gauge the show, the theater, and the probable audience to the equipment he chooses. Less tends to be more in the better houses but, as was mentioned earlier, be careful not to design a system for only one show if it will be used for an entire season.

Plan to have a mixing board that can handle at least two monitor sends, but be wary of complex equipment. In some theaters you will find that the union technician who will run the show is not always a sound engineer.

Imaging is most important; plan to arrange the speakers so that the sound source cannot be detected easily. Remember, the less the audience is aware of where the sound is coming from, the better it appeals to them. (Here we may begin to appreciate the fundamental difference between concert sound and theater reinforcement.)

Advance planning is a must. You have read numerous times the word "ideal," and that is because the sound designer and then the engineer are never in ideal conditions. Working in different legitimate theater houses can be like building a new studio from project to project; the environment is never the same. In the case of the typical off-Broadway black box, it is best to work with designers to ensure proper placement of single or clusters of speakers, and then hide them!

If you are lucky enough to be called in when a new theater is being built or renovated, encourage the architect to plan for sound. See to it that architectural considerations are made for speakers and monitors and, above all, please make sure that two discrete channels are created for communication and program feed. You need to take the approach of building a controlled listening environment within the theater, and architects are not averse to your input if it is made during the early planning stage. Physical beauty of a theater can be no compromise to acoustical satisfaction.

Keep in mind that the producer in the theater is the first person the sound designer has to please, then the audience. If the audience is displeased you'll know it, and the producer will remind you anyway! So, initially gain their trust. As designer you make the decisions regarding equipment, equalization, alignment and sound volume. The producer generally wants things louder, but you're the one who will know how to give him that without blowing the whole system sky high! ■

Ray Parker, Jr. | Supertramp's Rick Davies | George Duke | Chick Corea

Personal Use Studios...
notes on individual motivations & requirements.

by Jimmy Stewart

Freedom, flexibility, and the ability to work in an environment tailored to individual tastes, has prompted many musicians to invest in their own recording studios. After all, a commercial facility has to cater to varying tastes; 50% of the customers may like it, yet 50% may not. A musician first creates an environment to suit his or her own creative tastes whereas, by and large, a commercial studio tends to create a studio design from the engineer's standpoint.

Having their own studio available to them 24 hours a day provides artists with the space to comfortably create, rehearse, listen, study, learn, experiment, and produce in their choice of location. A studio represents a large investment if it is done properly. However, being able to charge studio time back to the record company for each project has made it a sound investment for many recording artists; an investment that pays off.

In the same way, a studio that a musician constructs for his or her personal use — and not to be confused with a "high-tech" facility that is intended to be hired out on a commercial basis — invariably reflects on the artist's personal musical reality. A studio put together by a rock band that's heavily into multitrack technology will be somewhat different in feel and atmosphere to one created for a jazz artist. Nevertheless, there are fundamental requirements of any musician-owned studio, even if it is simply somewhere that an artist can leave his instruments set up for an indeterminate length of time, so that he or she can hit the record button whenever divine inspiration strikes.

To find out more about the particular technical and creative requirements of musicians that have built their own recording facility, *R-e/p* visited the studios of *Chick Corea, George Duke, Ray Parker, Jr.*, and members of the band *Supertramp*. As we discovered, each of these artists has their own individual musical needs and technical requirements.

RAY PARKER, JR.'s "Ameraycan Studio"

Ray Parker, Jr. is a multi-talented instrumentalist: he plays guitar, piano, drums, bass, and sax, and is a compulsive songwriter. Parker has played on many studio dates as a rhythm guitarist, earning the name of "Skank" for his funky playing. The man also knows how to make hits. He has garnered a total of four Gold albums: *Radio*, *Rock On*, *Two Places at the Same Time*, and *A Woman Needs Love*.

Parker's facility, known as "Ameraycan Studio," is run by his long-time friend Steve Hallquist. While Parker likes to engineer his own product, when producing other acts — including Cheryl Lynn and the group Brick — Hallquist runs the console.

Seating himself behind the console in his studio, Parker began to reminisce: "I couldn't find an engineer who would give me the sound I heard in my head. Their attitude was, 'Oh yeah, I understand what you mean,' but after I left they would do it their own way. And usually it didn't sound like the sound I needed. I knew the record company wouldn't like it if *I* didn't like it.

"I then met Steve Hallquist, who is articulate and knows a lot about equipment. He's a good engineer, and got me into my first multitrack recorder: an MCI JH-16 16-track, which I put in my small house in North Hollywood.

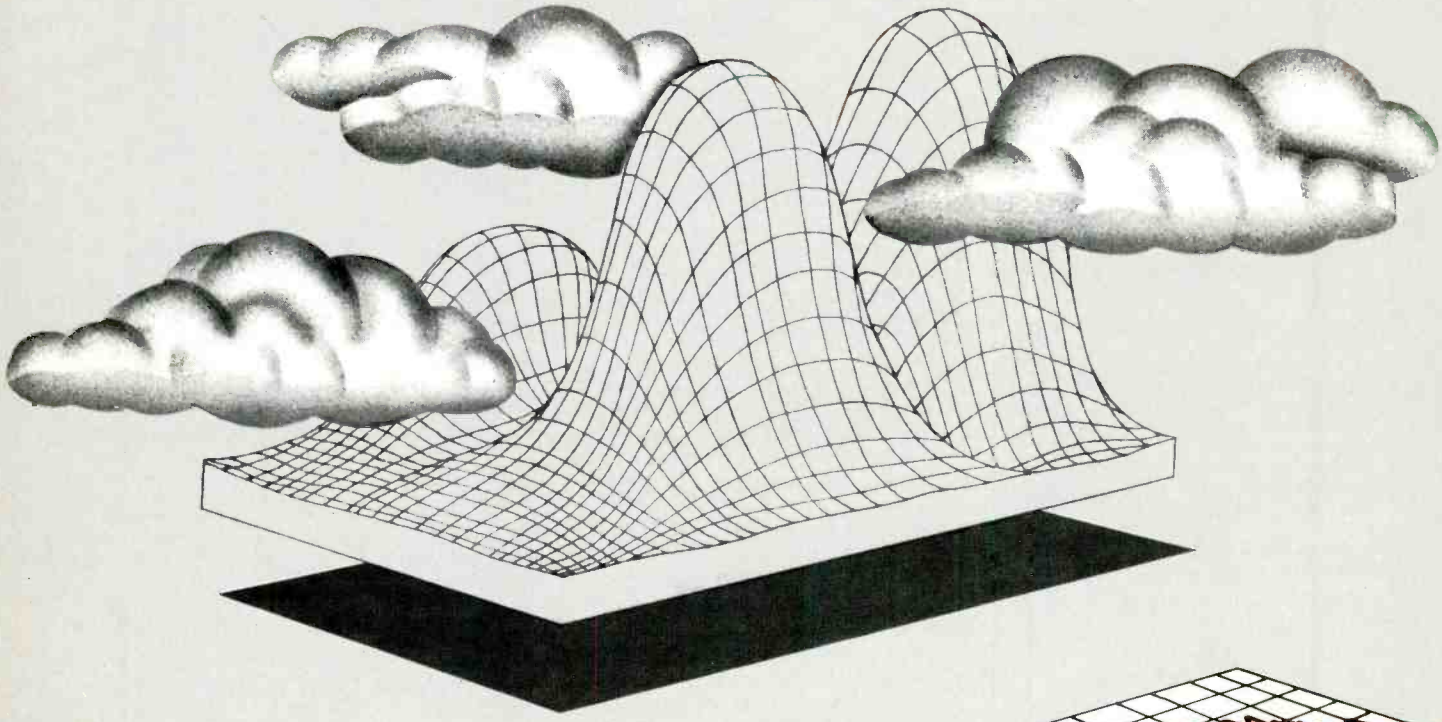
"There were wires all over the place and no air conditioning. Everything was miked with a Shure SM-57. I just stuck this mike in front of anything I wanted to record, and most of the time I plugged the electric instruments directly into the 16-track or the console, without a direct box.

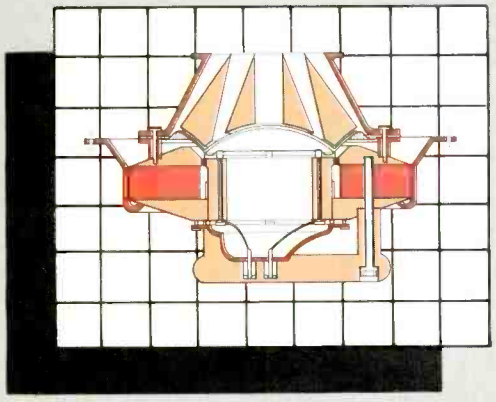
"Leakage problems were solved by recording each instrument separately. I

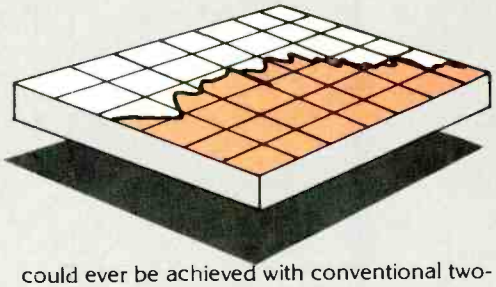
Ray Parker, Jr. (R); Steve Hallquist (L) at the MCI JH-636 console of Parker's AMERAYCAN studio control room

Photography by Kathy Cotter

NOW IS THE DAWN OF A NEW DECADE IN SOUND ENGINEERING







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M4 opens a whole new range of sound system applications, first and foremost of which is the design of a three-way system from three-way components

Until M4, three-way systems could only be designed using two-way components. While these adequately covered the upper and lower frequencies, they required a crossover point at that precise point in the midrange decade where the human ear is most sensitive, not a particularly desirable solution when true sound reproduction really counts.

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For additional information circle #62

as bound and determined to beat the gineers in the commercial studios, ho had never cut hits and cost me a lot of money. I kept thinking about the electronics that I had studied in school, and my fascination with tape recorders and gadgets. If I got stuck with any recording problems I could visit my friend Reggie Dozier at the ABC Studio Complex; he used to let me work out my recording ideas late at night, while he slept on the couch. Anything I needed to know, I could wake him up and ask.

"I cut three an one-half albums at this house and I liked it, except for the lack of air conditioning. Later on, friends became a problem. I'm a social guy, and friends would hang around the house all the time. Also, there was the wiring problem; someone could possibly trip, and even be electrocuted.

"That's why Ameraycan Studios was built. Scott Putnam helped us design the studio. He did an excellent job; not one thing had to be changed. This studio was built to create music, and cater to the musician. We are geared for making hits here."

The control-room hardware centers around an MCI JH-636 32-input transformerless, automated console. As Parker explained "This console is like a Ford . . . it runs all the time. I love the automation and its affordable price. If I want to rent the studio out my profit margin is higher.

"In making records, people get carried away in what they want to do, instead of what is economical. This board cost $50,000, and will do the same thing as a $150,000 console will do. I can't really see the purpose in spending that extra money to get all those problems. I've been in studios with all that real expensive hardware, and they're sitting there working on fixing this thing. And by the time they get through, I guess the sound was great, but we musicians lost the groove! What good is having a great sound if you don't have a good feeling performance? I need equipment that's going to work *all* the time.

"When I had the MCI machines in the house, they were already tested out. There was dust in the machines like you couldn't believe. The console was so full of dust it had changed colors, and I had poured everything down it you could imagine. Yet these machines passed their test with flying colors."

The multitrack is an MCI JH-114 machine equipped with Auto-locator and full remote control. Stereo mastering machines comprise an MCI JH-110B with remote, and a JH-110A. Control-room monitors include JBL 4343's and 4311's, as well as Auratone 5-C cubes.

On the choice of monitors, Parker commented: "I started out with JBL's at the house, and that's what I am accustomed to listening to." Monitor amps are by BGW, a McIntosh and Crown D-75 amps being used for the cueing system.

Parker in the studio . . . note the mirror faced gobos, used to emphasize the brightness of his guitar parts

Steve Hallquist put the processing equipment together for Parker. Apart from the normal complement of compressor-limiters, Ameraycan Studios is equipped with an Eventide Harmonizer®, Lexicon Prime-Time digital reverb, as well as an AKG BX-10E dual spring reverb.

Regarding choice of mikes, as Parker explained, "Steve and I like to use a Crown PZM™ on the piano. We also have the normal assortment of mikes, but we are not overstocked. I like to use a U-87 for my vocals and, for gang-related vocals, the C414."

When recording guitar Parker likes a "hard" signal, and doesn't use an amplifier. Instead, he takes the guitar through a direct box. There are also times when he may go directly into the multitrack machine, and bypass the board.

When composing, Parker prefers using the studio. He will lay down a track with a Maestro-King drum machine, which helps to get him into the groove. Depending upon the process, he may lay in next Fender Rhodes, guitar, or bass. He then starts to work with the lyrics, and the demo tune is completed.

When Parker is tracking he works like a scientist: "I am a compulsive worker; everything becomes important to what I am doing: the color of the walls, lights, and so on. If there is anything that changes in the studio, it kills my creative process. This studio allows me to get into right-brain technology, thereby placating my left brain to get more of Ray Parker into the product."

Did being a studio guitarist also help Parker in the creative process?

"I learned what everyone else was doing wrong," he remembers. "Many people seemed more concerned with what was going on technically in the studio, than with the music. I have turned that process around within my own studio."

"After all, a studio should cater to the people making the music, not vice-versa. See, when it gets vice-versa, you're talking about a record that sounds good, but has no chance of making the charts. No one will like to hear it, because it's not saying anything — it lacks feel."

Whenever possible, Parker prefers to work in the control room in front of the monitors. There are mike inputs located around the control room, plus outputs for the cue system; he even records vocals in the control room. When working on a project each track is pretty well positioned during the tracking session. By the time Parker has completed the song, mixing it is pretty straightforward.

"This studio is my creative quality," he concludes. "My ideas are so spontaneous, and everything is so important to what I'm doing. The studio allows me the time to take the song and different sounds apart so I can lock it in the way I want to. If I had to be in another studio while renting time, and changing studios every week, it would take me too long to get used to the equipment and the monitoring systems. And that can interfere with my creativity."

One is left with the impression that Ray Parker has really turned the technical aspect of recording around, and that it works for him. As he offered: "I like to create my own sound, and will continue to wear these two hats: artist and engineer. I don't like creating 'format music.' This studio is a tool for composing and producing hit records."

The recording process may vary, but the results are the same: a good, hot sound with a lot of feel.

SUPERTRAMP's Studios

Supertramp has achieved Gold Record status in nearly every country

. . . . *continued on page 98* —

Digital Update

MITSUBISHI AND AMPEX ANNOUNCE SUPPORT OF 48 kHz DIGITAL SAMPLING FREQUENCY

Signalling a major breakthrough in inter-format compatibility, both Mitsubishi and Ampex have joined the growing number of digital equipment manufacturers — including Studer, Sony and 3M — in support of 48 kHz sampling rates for PCM recording. According to Lou Dollenger, national sales manager for Mitsubishi's digital audio division, "Our design engineers examined these rate reductions very carefully, and they felt no sonic compromises would be necessary." He added that Mitsubishi equipment formerly used a sampling rate of 50.4 kHz, the industry's highest.

A team of factory engineers is being assigned the task of retrofitting Mitsubishi's existing recorders around the world, Dollenger said. The newest Mitsubishi recorder, the X-800 32-channel unit, will be equipped with a 48 kHz clock when it arrives in the U.S. this March.

Acceptance of the 48 kHz standard by the major manufacturers will simplify the introduction of digital recordings into a number of different areas, Dollenger offered, particularly broadcasting. "While this new standard still does not allow unrestricted tape compatibility, it certainly is an important step toward our goals in this area," he concluded.

Ampex has long proposed a common frequency for all applications except European radio broadcasting, which has already established itself at 32 kHz. This latter frequency is felt to be too low for professional recording, however, since it cannot support a 20 kHz bandpass. The company feels that the choice of sampling frequencies should attempt to minimize the need to convert from one rate to another, whether by analog or digital means.

Several techniques have been devised to digitally convert between rates having arbitrary ratios, preferably small integers. Taking into account the need for compatibility of sampling rates for digital television and the professional recording market, a sampling rate for the audio channels of a digital video recorder would ideally form an integer number of samples per frame for PAL/SECAM, NTSC, and film frames. Ampex offers that a 48 kHz sampling rate produces an integer number of samples in PAL/SECAM and at film frame rates; for NTSC, it produces an integer number of samples per five frames. As a result, Ampex intends to preferentially pursue and promote the use of 48 kHz in both audio and video professional products, while continuing to accommodate other, less favorable, sampling frequencies. ■■■

around the world, and is known for its eclectic style. The group's songwriter/vocalists are keyboard-player Rick Davies, and keyboard- and guitar-player Roger Hodgson. Davies is a self-taught musician with a great feel for the street, as opposed to Hodgson, who is trained in basic music and has an "ethereal" approach. John Anthony Helliwell plays wind instruments, and also doubles on vocals. Heavily into R&B and jazz music, Helliwell acts as front man for the group and loves to play music. Bob Benberg, the only American in the group, is drummer, and Dougie Thomson the bass player adds a lot of energy to the group.

Ian Lloyd Biseley acts as the band's monitor engineer, and was responsible for setting up their studios. Because of recording commitments, Supertramp was unavailable for interview; nevertheless, Biseley served as spokesman for the band.

So what made members of Supertramp decide to build their own recording facilities?

According to Biseley, "Supertramp has progressed to the point where each member needs to do his own thing. It makes working together easier by having our own individual studios. A lot of artists are now making this investment for the freedom and flexibility of having a working environment tailored to their own taste, because it pays for itself with the end result: good creativity."

Supertramp's Rick Davies (R) with Ian Biseley (L) at the modified API console in the guest house studio

Rick Davies had his studio, located in Encino, California, built from the ground up. It comprises an extension to a guest house. Biseley has installed patch panels for mike inputs and cue outputs in all of the rooms of the guest house. As a result, any or all of these rooms can be used to achieve the isolation necessary for recording. The control room hardware comprises an Ampex MM-1200 24-track machine; an API board; and Tannoy monitors powered by McIntosh amps. Microphones available include Sennheiser MD421, AKG C414, Neumann U87 and U47 models.

Roger Hodgson is currently having a four-bedroom house next door to his existing house remodelled for his new 24-track studio, located in Nevada City, Northern California. The house will

. . . openness and brightness are keynotes of Davies' recording environment.

also be used as sleeping quarters for the group when they record at the studio. At present Hodgson has an eight-track demo studio in his house. The studio is fitted out with an Amek-TAC 16-in/4-out board, and two Otari two-tracks. Planned for his new studio is a Studer A-800 24-track machine, a pair of Studer A-80 two-tracks, and two Studer cassette decks. The new studio will feature a modified Trident Series 80 board. Biseley plans to change buffer amps and varying points of the equalization section to make the Series 80 sound like the older Trident A-Series consoles.

John Helliwell's studio, Biseley says, is a cross between a music room and a studio. Comprising a simple 8-track room located in a guest house, Helliwell uses an Otari 8-track, coupled to a 16-in/4-out Bi-Amp console fitted with an extensive patchbay. Outboard effects include a rack-mounted MICMIX Master-Room reverb, and an MXR digital reverb and delay unit. Mastering duties are handled by a Pioneer two-track, and a Nakamichi cassette deck.

"We get good sounding tapes out of here without spending any money at all," Biseley offers.

Bob Benberg wanted a good studio without having to spend a lot of money. His studio, which is located in his home, features a Neotek board, an Otari MTR-90, and an Ampex two-track. Benberg chose the Neotek for the quality and price, Biseley says. Tannoy Super-Reds with electronic crossovers serve as monitors in the control room. Mikes include AKG C414, U87 and a few U47 tube mike models. A collection of Shure SM-81 microphones used by the band on the road are also kept at Benberg's house for use in his studio. Reverb comprises an EMT-140 plate and a MICMIX Master-Room. Additional outboard gear includes a Delta-Lab DL-2 digital delay line, and two UREI 1176 compressor-limiters.

"Bob likes toys — he loves to experiment with sound," Biseley explains.

At one stage Biseley offered to build bass-player Dougie Thomson a studio on his boat. But, as Thomson commented to him at the time, "Why do I want a studio on my boat when I've got four studios, and four different sounds

George Duke at the Soundcraft Series 3B console of his LE GONKS - WEST office/listening room/studio . . .

to choose from?"

Supertramp is a very close-knit group, but its members lead varied life-styles. With their individual studios, band members can express more of their individuality and, as far as making music is concerned, the facilities become a tool for their composing. This new way of working has given them a great deal of energy, Biseley expressed. In the past, Supertramp has had a reputation for spending months in the studio recording an album, which can become very expensive in session time.

"When we work in our separate studios," Biseley offers, "We have the opportunity to get the structure and the feel of the tunes real tight. The new album we are working on now will the best demo-ed album we have ever done!"

GEORGE DUKE's "Le Gonks-West Studio"

George Duke's recent album, *The Clarke/Duke Project*, was conceived at his own studio, Le Gonks-West. As master musician of rare versatility, Duke has evolved with the best in Jazz and Rock; he now chooses to express himself in the R&B idiom. His previous album, *Reach For It*, went Gold, and he has also produced Taste of Honey and Sea Wind. Engineers with whom he has worked closely include Tommy Vicari and Eric Zobler.

Duke has remodeled the maid's quarters of his house, which is located in the Hollywood Hills of California. He related the story behind setting up his recording facility: "This room started out totally as a listening room. There was just going to be a big couch in the center of the room, and I was going to put up a couple of JBL 4311's on the side, and use it to listen back to mixes I had done at other studios. From that idea, the room has changed into what it is now.

"Having a studio in the house allows me a chance to come downstairs and record in the middle of the night if I want. I don't have to worry about getting bumped out by another act. I can charge my own studio bills back to the record company. It is also good for me headwise, because every day that I come down here I learn something new about electronics, and about this room. So it is a good learning experience for me to figure out what makes all this stuff tick.

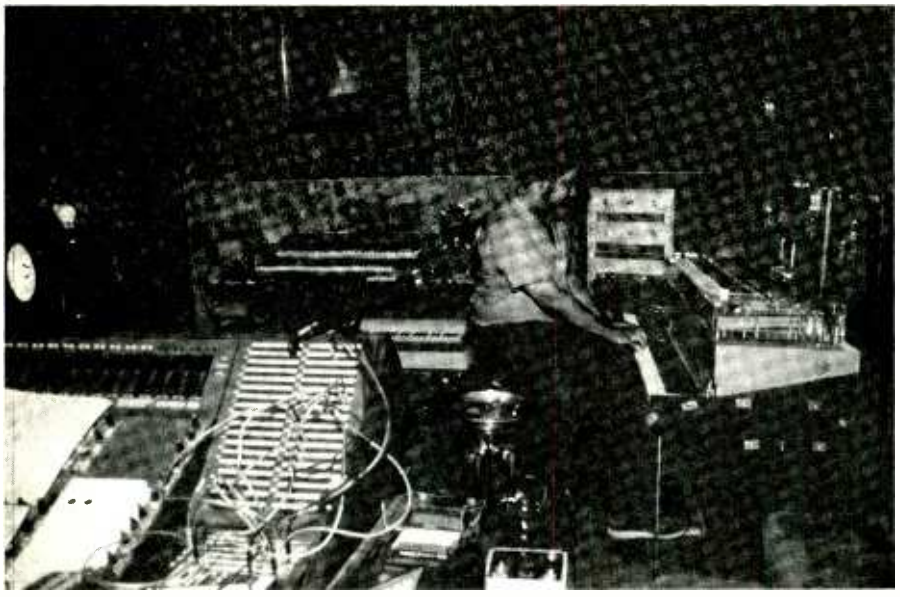
"It gives me a chance to experiment with sound that I never felt I had enough time to do in a regular studio, because I was worried about dollars. Being at home I can experiment as much as I want, because the only thing I am paying for is an engineer. And if I want to send the engineer home, then I'll just experiment by myself.

"I think the records I make sound good, and I have two good engineers that are very much interested in sound. That's why I have this room: it allows me a chance to really seriously get into making good records."

Le Gonks-West features a Soundcraft Series 3B transformerless console which, Duke says, has a clean sound, and is easy for him to organize in his mind when working. Recording duties are handled by a 3M 79-Series transformerless 24-track, and a new Ampex ATR-100 two-track that, according to Duke, "has a lot of punch to it."

A pair of small Yamaha speakers sitting on the console are used for monitoring when Duke is tracking in his office with the door shut, which reduces any sound leakage in the recording area. For its principal monitoring system, the studio has UREI Time Aligned™ speakers, powered by Bryston amps.

A good selection of outboard gear is available, including dbx Model 160 compressor-limiters, and the new



Keyboards and Synths . . . the other side of George Duke's "office"

Valley People Kepex™ II expander/noise-gates.

As yet, George Duke has not mixed an album at Le Gonks-West, but the facility is said to be ideal for tracking and overdubs. On albums prior to *The Clarke/Duke Project*, basic tracks were recorded at outside studios, and then the tapes brought back to Gonks-West. The next stage, Duke explained, would be to clean up any rough parts: "I will bring a tape in, listen through and make sure that what I heard in the other studio was what I'm hearing here, and then make any necessary bass or keyboard repairs. The direct box I like to use on bass and keyboards is made by Westlake Audio; it's an active DI box, and gives us a very clean sound.

"Then I'll usually add necessary guitar parts next. If I need a particular kind of 'grunt' guitar sound, we'll go from doing most of it direct, to putting the guitar amp back in the jacuzzi. I have a line run back there, which makes a great effect for live sounds, hand claps, or bizarre guitar sounds.

"I might consider doing horns next, depending on what's happening. More recently, I've started putting the vocals at this stage. Why? Because I don't want to cloud the tracks, and make the sweetening or overdubs work around the vocal line. I like to use a U89 for the vocals; it has a bright airey sound in the mid-range."

In the past Duke has used Westlake Studio B to record horn overdubs, with the Sea Wind horn section. With the latest album, however, he planned to track horns in an upstairs room at his studio.

"I decided this time that I was going to try, because it's home, and I'm not going to lose any money anyway. I'll take a bath if it is not going to work! Because of the acoustic piano overdubs I've gotten, and the ambience I can achieve, the horns should sound real good up there.

"Most of the recording work here is done in my office. I have two acoustically-treated doors which, when closed, result in very little sound getting through. I do all my vocals in there, and any small instrumental things like glockenspiel, single horn overdubs, or

flute. That really does sound amazingly good. It's got a lot of wood, cork and, of course, my Gold records hanging on the walls.

"Having my own studio fits my career plan. And, economically, it's working for me. Just think of all these other guys giving all that money to studios to improvise and work out their material. Nowadays, you can't afford to do it.

"All I have to do is turn off the refrigerator, unplug the phone, and close the door."

Summing up the process of building a studio, Duke offered that "people have a certain instinct and use street sense to decide what to do with their career. For me, when I made my first Gold record, I decided to invest my money in a studio, because I thought that was the best investment I could make at the moment."

Chick Corea at the Trident Series 80 console with Bernie Kirsch at Corea's MAD HATTER studio

CHICK COREA's "Mad Hatter Studio"

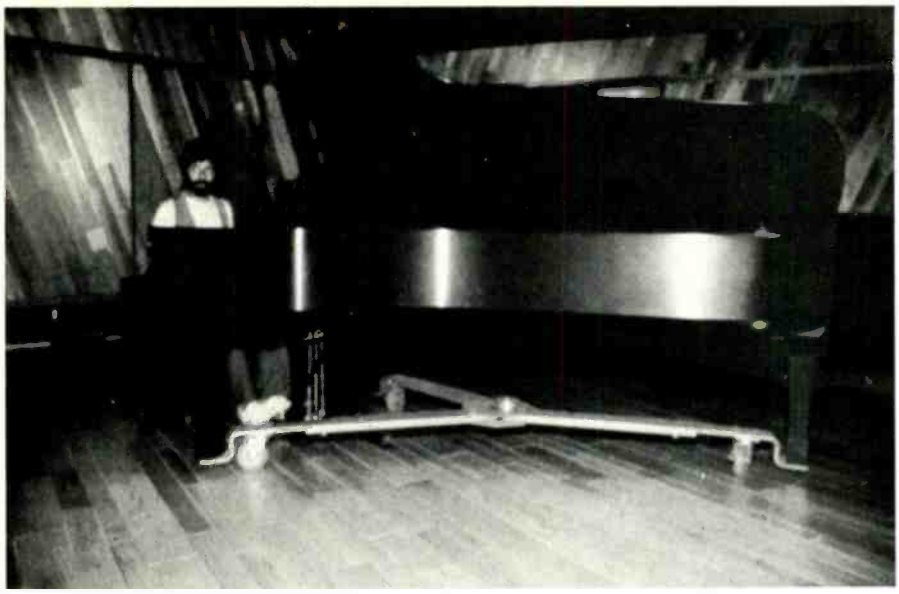
As a keyboardist and composer Chick Corea is skilled in acoustic and electronic styles. A true contemporary musician, he has reached Gold success with *Return to Forever*. Corea's other albums, such as *Friends, Music Magic,* and *The Mad Hatter*, are considered by many as classic musical experiences. His latest album, *Three Quartets*, was recorded at Corea's new studio, Mad Hatter.

The recording area at Mad Hatter is laid out rather like a concert stage, with oak floors and full ceiling, in the center of which is a skylight. Corea outlined his thinking behind the studios's design: "Building a studio was something I had been thinking about ever since I started getting really interested in the whole technique of recording. I guess that about 10 to 11 years ago. The first five or six years I was making records I had absolutely no technical interest at all in the recording process. I used to go into the studio and play all my music; I think I got it down enough to know that I should wait until the guy behind the window said 'Go.'

"But when I worked with Stanley Clarke on *Return to Forever*, that band was very much into the technique of recording. I then met [engineer] Bernie Kirsh in 1975, and made *The Leprechaun* with him. It was during that period that I was starting to become fascinated with the recording process, and started collecting recording gear.

"The turning point came when I put a 13-piece band together in 1978, and we decided to make the in-performance sound of the band really happening. Bernie and I enlisted the assistance of Mark Levinson, who was a designer of home audio gear, and also an old friend of mine. So we got Mark to come in and help us put some stage monitoring together for the four string players. We came up with a system that included some very, very high-quality microphones — B&K and Schoeps — and some pre-amplification that Mark designed. Each string player had his own system with a Klipsh speaker, as did my keyboards and the out-front soloist. After the brief tour was over I had all this equipment, including a portable Stellavox 15 IPS machine made in Switzerland.

"We had so much gear after the tour that we set it up in my living room, and began to fool around with it. It came to a point where we decided that if we rented a 24-track we could virtually make a record in my living room. At that time we had purchased one of Mark Levinson's ML5 modified Studer two-tracks, so all we had to do was rent an Ampex multitrack machine. We had all the microphones, amplification and a good two-track, so we made two records in my living room: Gail Moran's first solo album; and my record called *Tap Step*."



MAD HATTER's recording area . . . acoustically designed and built for brightness

Once Corea had been bitten by the recording bug, he turned to the professionals for help in turning his dreams of building a home studio into reality. As he recalled, "We called up Sierra Audio to do the construction, and Tom Hidley to handle the initial design for us. Actually, we worked hand in hand with them, because we didn't necessarily want that exact turnkey studio. Having ideas of our own, we worked hand in hand with our experience, and what we wanted to get from a studio."

Mad Hatter is equipped with a Trident Series 80 console, chosen for its cleanliness, flexibility and simple layout. For a multitrack machine, they chose a Studer A-80 24-track. According to resident engineer Bernie Kirsh, "We wanted a Studer because of the way it handles the tapes, which is very important to us — the transport on it is incredible. Mark Levinson initiated the idea of modifying the electronics, and went through all the cards, designed and changed components; he did some other work that allowed us to have a quieter recorder. We don't use any noise reduction here, and we can really put a lot of level on the tape; the electronics handle it."

Until recently, the studio has been using an electrostatic monitoring system. But, as Kirsh pointed out, "they are rather unrealistic for the type of recording that we're doing, because they won't deliver the kind of punch you get with conventional speaker systems. However, they do have a beautiful musical quality and 'air' about them.

"We wanted to keep that acoustic reproduction that this electrostatic system was giving us, yet have something that was realistic for a studio setting. In the past we had done some on-the-road work with McCune Sound, and they used a concert sound system developed by John Meyer that had a real nice quality to it. Looking into it a little further, we heard that John was developing a studio monitor system so we went up to the Record Plant in Sausalito where it had been set up for demonstration. We took some acoustic piano and electric music, listened to the speaker system, and found that it was just what we were looking for; it played acoustic music well, had a beautiful warmth to it, and was an accurate reference."

Kirsh is particularly fond of the Schoeps range of microphones: "They can be used where you might use a Neumann U87 or an AKG C414, and are nice and smooth. The Schoeps has an openness and a clear top end that I prefer over the 87's. You can add a pad to it if you need it, which just screws on the microphone body."

"I did an album for Gail Moran, which we recorded at Chick's home, and I used some Schoeps on the drums for basic tracks. Everything else was covered by a B&K — a nice-sounding microphone. I have not heard a better microphone for vocals in our environment; you can put your mouth right up to it and it doesn't sound that way. It has great intimacy and clarity."

Chick Corea gives every appearance of being pleased with the way his studio turned out: "When I'm working in here I lose track of the hours. When we get into an album project it is more than just a one-day thing. I like to work after humanity has gone away to sleep, and things are very peaceful."

"Yeah, it's very comfortable here," Kirsh continued, "and we have always had our priorities right. Which is that all of this technique — the building of the room, and the recording process — is a servant to creating music and art. I think the main reason why this studio feels real comfortable is because that's the way it is operated: to serve the musician."

□□□

New Products

NEW DYNAFEX SINGLE-ENDED NOISE REDUCTION SYSTEM FROM MICMIX

The Dynafex Model D-2B noise reduction system does not require any encoding or decoding, allowing the unit to reduce noise on virtually any audio source. Although compander-type noise reduction systems have been available for some time now, such systems do not remove noise present in the original signal; they only prevent further noise build-up by the tape recording process. The Dynafex, however, is said to be capable of reducing noise on the *original* material. The device reduces noise by continuously analyzing and responding to the frequency content and amplitude of the signal.

The Dynafex is a two-channel system capable of reducing noise by up to 30 dB or more. Each channel has a Threshold control that determines at what signal level maximum noise reduction occurs. A hard-wired Bypass switch is also provided for each channel. The unit can be adjusted for reference levels of +8, +4, 0, or -10 dB, to accommodate a wide range of professional equipment.

In the recording studio, Dynaflex may be used on a per-channel basis as a sophisticated noise gate/noise reduction system, as well as in a mixdown situation just prior to the mastering tape machine. Adding further to its versatility, says the manufacturer, is the fact that it can be used to reduce noise on tapes where no noise reduction has been used, or to further reduce noise on tapes after they have been decoded from compander-type systems.

Being a single-ended system, the Dynafex is capable of operating on-line in broadcast applications, as well as in the production room. According to the manufacturer, this unit will also greatly reduce noise on cart machines, announcer's microphones, VTR audio tracks, phonograph records, as well as motion-picture soundtracks.

MICMIX AUDIO PRODUCTS
2995 LADYBIRD LANE
DALLAS, TX 75220
(214) 352-3811

For additional information circle #69

NEW ENGLAND DIGITAL ANNOUNCES NEW OPTION FOR SYNCLAVIER II SYNTHESIZER

The Digital Analysis/Synthesis Option (Sample-to-Disk™) allows the user to add *real* instruments to the Synclavier® II. The sampled data is recorded onto a Winchester Disk at a sampling rate up to 50 kHz for a minimum of 100 seconds, to a maximum of 24 minutes recording time. The 50 kHz sampling rate ensures high-fidelity digital recording.

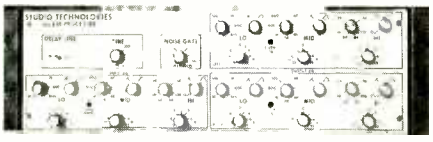
The user can then analyze and modify the recorded timbre using sophisticated software techniques provided by New England Digital. Complete dissection and reconstruction of the sampled sound can be specified. This finished timbre may then be triggered from the Synclavier II real-time keyboard for use in recording and live performance.

NEW ENGLAND DIGITAL
P.O. BOX 546
WHITE RIVER JUNCTION, VT 05001
(802) 295-5800

For additional information circle #70

STUDIO TECHNOLOGIES LAUNCHES NEW REVERB PROCESSOR

Designed to enhance any reverberation system, the new processor contains analog time delay, three-band parametric equalizers with continuously variable Q, two noise gates, and the exclusive Stereo Stretcher™.



Configured so that the variable time delay and one parametric equalizer are in the send circuit to the reverb device, each of the stereo returns from the reverb unit is routed through a parametric equalizer, the Stereo Stretcher, and finally a special noise gate, which follows the reverb decay down to a -60 dBm level then gently kills to a -90 dBm level. A noise gate threshold control allows the user to select a reverb time, while the noise gate will shorten the "tail out" of the reverb to a more normal 3.5 seconds. This effect can be used to "fatten" certain sounds, such as vocals and horns.

Many other new effects are possible using the Studio Technologies Reverb Processor. For example, the use of much more high-frequency boost on the reverb returns is possible, because the noise gate will eliminate any noise not masked by the signal. The Stereo Stretcher widens the stereo spread of the reverberation, thereby creating a wall of sound stretching beyond the speaker boundries. This effect is made frequency selective by using the two parametric equalizers in the circuit.

Recommended user price of the Reverb Processor is $1,500.00.

STUDIO TECHNOLOGIES
6666 NORTH LINCOLN AVENUE
LINCOLNWOOD,IL 60645
(312) 676-9400

For additional information circle #71

AMPEX UNVEILS UNISYN SYNCHRONIZER FOR UP TO 16 TRANSPORTS

Designed with a universal interface, the UNISYN comprises a modular system that can be expanded by adding one synchronizer for each controlled transport to support any post-

production requirement. One UNISYN is required for each recorder, and the system can lock up to 16 transports in sync. The unit's software automatically adapts to transport dynamics for optimum cueing and fast lock up. The synchronizer can interface with audio and video recorders without special external interfaces.

UNISYN is described as the first synchronizer to provide wide range, continuous variable speed lock — the slave can lock to the master recorder even when the master speed varies dynamically from 0.25 to 2 times the play speed. Synchronization of a slave recorder can be controlled to 0.01 frame accuracy.

Each slave recorder can locate and synchronize 10 separate segments in one continuous pass of tape on the master recorder, to provide significant time savings in post-production. Eight optically isolated time code controlled outputs assignable to 10 events are said to make UNISYN a truly versatile synchronizer. The unit offers the choice of time code or machine tach pulse cueing, providing additional flexibility and time saving.

UNISYN is available in two versions: control panel for manual control, and digital interface for integration into post-production and editing systems. The control panel can be remoted with a 20-meter cable.

Professional user price of UNISYN is $6,450; delivery is expected in August.

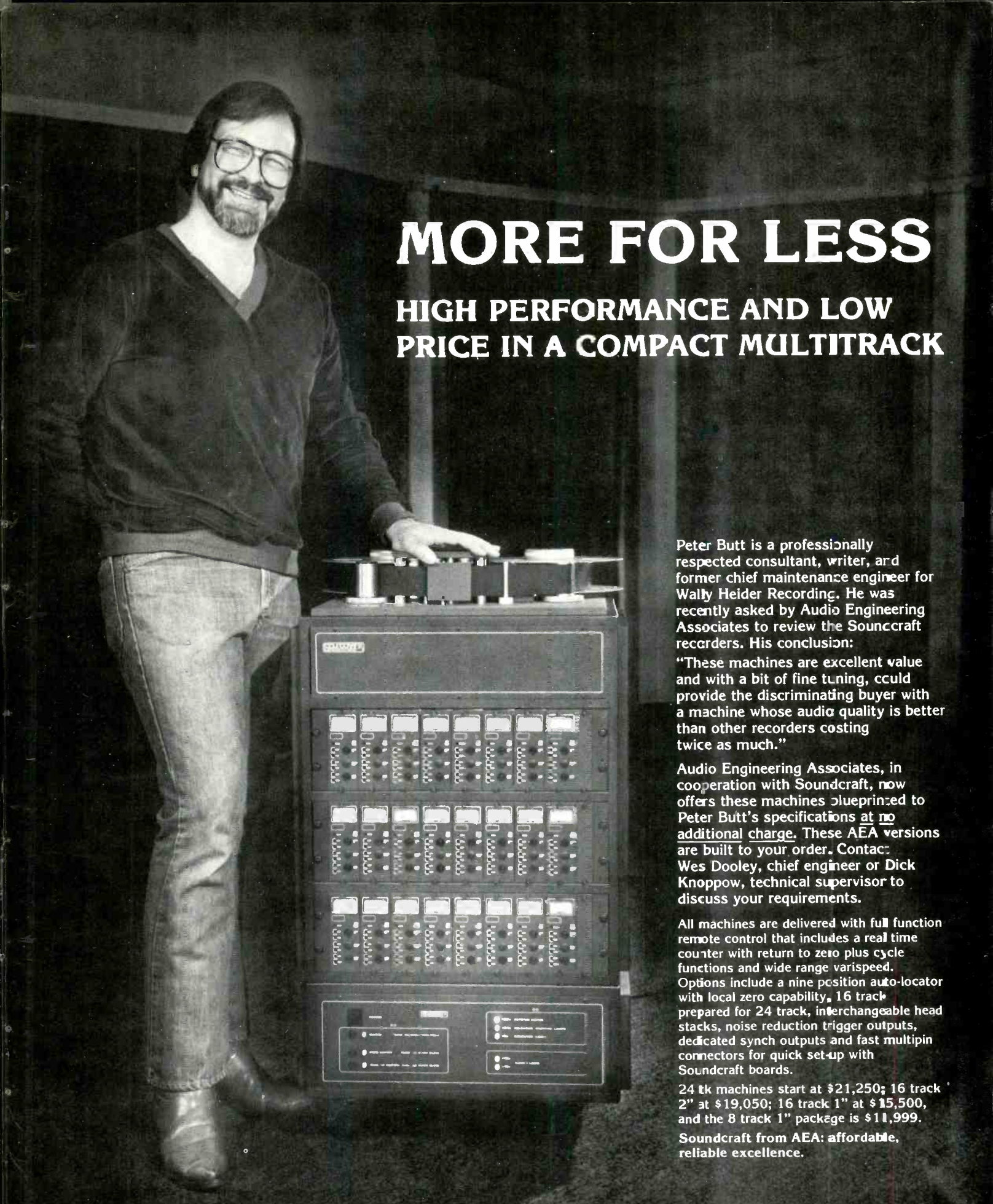
AMPEX CORPORATION
401 BROADWAY
REDWOOD CITY, CA 94063
(415) 367-4151

For additional information circle #72

SERIES 30 TAPE MACHINES FROM TASCAM

All three new models — the 2-track Model 32, 4-track Model 34 and 8-track Model 38 — are based on a similar transport design, and accommodate up to 10½-inch diameter reels. Models 32 and 34 both run ¼-inch tape at 7½ or 15 IPS, while the Model 38 runs ½-inch tape at 15 IPS. Full sync recording is provided on all machines, including the two-track.

New high-torque, slotless DC reel motors are said to minimize wow and flutter, exhibit less start-up noise, and have an extended service life. Smooth record/play motion is further assured by a polyurethane rubber pinch roller,



MORE FOR LESS

HIGH PERFORMANCE AND LOW PRICE IN A COMPACT MULTITRACK

Peter Butt is a professionally respected consultant, writer, and former chief maintenance engineer for Wally Heider Recording. He was recently asked by Audio Engineering Associates to review the Soundcraft recorders. His conclusion:

"These machines are excellent value and with a bit of fine tuning, could provide the discriminating buyer with a machine whose audio quality is better than other recorders costing twice as much."

Audio Engineering Associates, in cooperation with Soundcraft, now offers these machines blueprinted to Peter Butt's specifications at no additional charge. These AEA versions are built to your order. Contact Wes Dooley, chief engineer or Dick Knoppow, technical supervisor to discuss your requirements.

All machines are delivered with full function remote control that includes a real time counter with return to zero plus cycle functions and wide range varispeed. Options include a nine position auto-locator with local zero capability, 16 track prepared for 24 track, interchangeable head stacks, noise reduction trigger outputs, dedicated synch outputs and fast multipin connectors for quick set-up with Soundcraft boards.

24 tk machines start at $21,250; 16 track 2" at $19,050; 16 track 1" at $15,500, and the 8 track 1" package is $11,999.

Soundcraft from AEA: affordable, reliable excellence.

AUDIO ENGINEERING ASSOCIATES
1029 North Allen Avenue • Pasadena, CA 91104 • (213) 798-9127

which resists wow-inducing flat spots, reduces tape slippage, and has a longer useable life.

A four-digit fluorescent LED tape counter is tachometer driven for consistent indications regardless of the tape position or reel size; the counter is complemented by an accurate zero search function. A remote punch-in/out jack accepts the optional RC30p foot switch, for hands-free inserts and overdubs.

The Model 38 features a precision diecast head mounting block and a plug-in head assembly for field replacement with minimal or no realignment; head angle can be adjusted while tape is in motion. A manually retractable shield minimizes noise for critical recording or playback.

All models have improved high-density permalloy record/sync and repro heads, which typically deliver 40 Hz to 22 kHz response in either mode at 0 VU level; head life is claimed to be 20% to 50% greater. Flip-up head covers facilitate editing and maintenance.

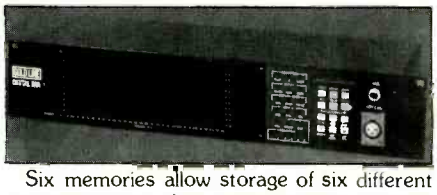
Recommended retail prices: Model 32 $1,300; Model 34 $1,700; Model 38 $2,750.

TASCAM
7733 TELEGRAPH ROAD
MONTEBELLO, CA 90640
(213) 726-0303

For additional information circle #74

GOLD LINE INTRODUCES MODEL 30 REAL TIME SPECTRUM ANALYZER

The heart of the Model 30 third-octave analyzer is a quartz crystal-controlled filtering system called "Switched Capacitive Filtering," a technique that results in filters which do not drift in bandwidth, center frequency or flatness from one filter to the next. Switched capacitive filters will withstand hard usage, the abuse of the road, and do not require recalibration due to ageing or temperature variations that are inherent in all analog filters. Spectrum analysis is accomplished by use of a powerful microprocessor operating at 6 MHz.



Six memories allow storage of six different response curves. Any response curve or group of curves may be displayed as an average, or as a weighted average. A choice of "Flat," "A," or "User Defined" weighted curve may be employed. Extra weighting may be applied to any of the stored curves, or groups of curves. The Model 30's memory features allow sophisticated equalization for sound reinforcement to be accomplished readily without guesswork.

The Model 30 also incorporates a psuedo randomly generated pink noise source. Future optional facilities are provided for precision noise gating for RT60 measurements. Other optional features are planned, such as computer interfaces for IEEE 488 and RS232 busses, X-Y chart recorder interface, and non-volatile storage of response curves using battery backup.

GOLD LINE
P.O. BOX 115
WEST REDDING, CT 06896
(203) 938-2588

For additional information circle #76

AUDIO & DESIGN F601 SUPERDYNAMIC LIMITER

The Superdynamic has been designed to meet the challenge of the digital era. As a mastering limiter for recording applications, front-ending PCM units, satellite links and broadcast transmitters, the new unit offers a 100 dB dynamic range (threshold-to-noise), and 100% transient control.

Facilities include: precision stepped attenuators and controls; 20 dB make-up gain; side-chain pre-emphasis (0, 25, 50 or 75 microseconds); a unique music/voice ratio control (VO) system; transient clipper; symmetrical/asymmetrical (AM) output; output filtering (AM/FM) option; logic function tamper-proof inhibit; and dual function metering.

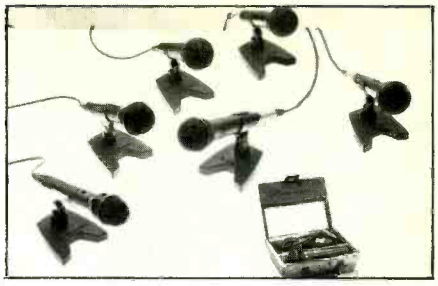
The F601 system is available as a unity-gain, single-channel (F601-R), or stereo/dual mono (F601-RS) package. Inputs and outputs are both electronically balanced (with provision for transformer option).

AUDIO & DESIGN RECORDING
P.O. BOX 786
BREMERTON, WA 98310
(206) 275-5009

For additional information circle #77

PRO-SERIES MICROPHONES FROM AUDIO-TECHNICA

The PRO-Series microphones — until now consisting of only two models — have been expanded with the addition of four new dynamic types, each provided with an on/off switch. The PRO2, a high-impedance unidirectional



vocal/instrument model, will be offered at $62.00, suggested list; while the PRO3, a unidirectional high-impedance vocal type, carries a suggested list price of $97.00

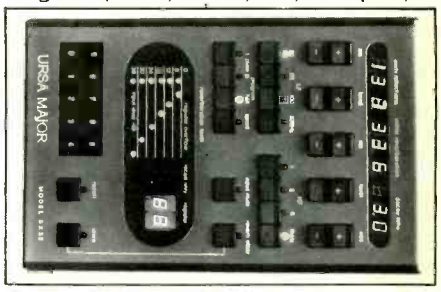
The PRO4H unidirectional high-impedance vocal microphone is provided with a separate cable with XLR/A3F connector and phone plug output; suggested list price is $125.00. The PRO4L, also a unidirectional vocal type, is a low-impedance model, and carries a suggested list of $125.00.

AUDIO-TECHNICA U.S., INC.
1221 COMMERCE DRIVE
STOW, OH 44224
(216) 686-2600

For additional information circle #78

URSA MAJOR 8/32 DIGITAL REVERBERATOR REMOTE UNIT

The new remote unit provides all the controls and displays of the mainframe front panel in a box that measures 5- by 9- by 1½-inches. The 8/32 now also features capability for comuter automated mixdown interface. Early Reflections, Initial Reverberation, Decay Time, and Low- and High-Frequency Decay are pushbutton adjustable within each of four basic Programs (Plate I, Plate II, Hall, and Space).



Non-volatile registers can store 64 complete patches or user-derived programs for immediate recall. LED displays show all settings in use, and a readout of the input and reverberation levels. The system will synthesize reverberation over a wide range of natural and unnatural spaces, including decay times up to 20.0 seconds.

URSA MAJOR, INC.
BOX 18
BELMONT, MA 02178
(617) 489-0303

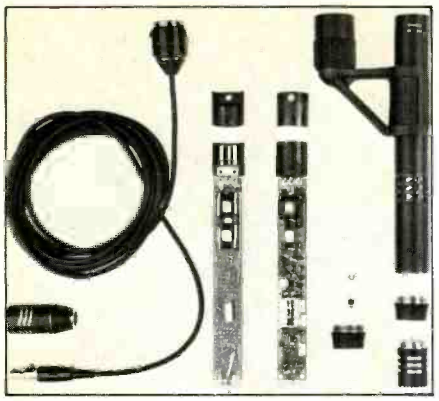
For additional information circle #79

AKG C460B PRE-AMP FOR CMC CONDENSER MICROPHONES

The new C460B microphone pre-amplifier is compatible with existing CMS Series mikes.

Standout features include:

- Wide dynamic range with low self-noise.
- Minimum distortion throughout the dynamic and frequency ranges.
- Stability of specifications even under extreme conditions, such as connection to excessively long cables.
- Minimum current consumption when connected to any powering voltage from 9 to



52V.

To provide a means of connecting CMS capsules to the C460B, an elastic adapter has been designed. Its mechanical construction is said to ensure considerably lower sensitivity to handling noise than encountered in a rigid joint. The outer case is made of an electrically conductive rubber material to ensure a low-impedance connection between the housings of the capsule and the microphone pre-amplifier.

THD at 1 kHz is a quoted 0.01% at maximum SPL level; equivalent noise level 15 dB SPL, according to IEC 179/A-weighted; dynamic range 125 dB; and maximum SPL for 0.5% THD over full frequency range 140 dB SPL.

The C460B pre-amplifier has been designed to anticipate future developments. Therefore, according to AKG, it not only satisfies fully the stringent requirements of digital technology, but offers better distortion figures than 16-bit PCM equipment. Since it is compatible with the existing CMC modular system, it can be used with a great number of capsules and accessories.

AKG ACOUSTICS, INC.
77 SELLECK STREET
STAMFORD, CT 06902
(203) 348-2121

For additional information circle #80

THREE NEW ACCESSORIES FOR REVOX PR99 TAPE MACHINE

The console frame, made from heavy guage steel, allows operation in three positions: horizontal, 30-, and 45-degree angle. The tilt position can be changed in seconds, without tools, by turning the large, side-mounted knobs. An optional lipped utility shelf may be mounted atop the frame. The console's recorder housing is built around welded steel side members, and

accepts the standard 19-inch, rackmount PR99 with no modifications.

The transportation-case version of the PR99 utilizes the same rugged housing as the console, with the addition of a snap-on front cover and heavy-duty, flush-mounted handles. The case if finished in thick, scuff-resistant vinyl. By replacing the handles with a mounting bracket, the case may be placed in the console frame.

The 19-inch rackmount monitor panel, which may be fitted in a 3½-inch space at the top of the cabinet console housing, includes an amp card, 6-inch oval speaker, volume control, and track selector switch. Installation is accomplished by simply mounting two screws, and plugging in the 7-prong DIN connector.

Professional user net for the console is $330.00; utility shelf $65.00; transportation case $225.00; and monitor panel $130.00.

STUDER REVOX AMERICA, INC.
1425 ELM HILL PIKE
NASHVILLE, TN 37210
(615) 254-5651

For additional information circle #81

FET-200 MOS-FET POWER AMPLIFIER INTRODUCED BY ASHLY

Describing the power MOS-FET as a new device which, in time, will render the conventional transistor obsolete in audio amplifier applications, Ashly president Bill Thompson says, "It's a good idea whose time has come. It's exciting to see such a genuine advance in component technology as the power MOS-FET. It enables us to produce a superb power amplifier with simple circuits. MOS-FET's require almost no drive current, so there is an absolute minimum of high-power components. They are so rugged that we have

still not failed one in our prototyping process."

Occupying only 3½-inches of rack space, and providing 100 watts per channel at 8 ohms (175 watts into 4 ohms and 350 watts into 8 ohms-bridged mono), the FET-200 is a fully complementary, push-pull amplifier with totally discrete, high voltage, wide bandwidth electronics; an approach that is said to assure low noise, low distortion, and excellent transient response.

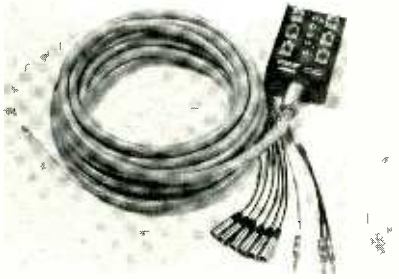
Inputs are bridging, active balanced (transformerless), equipped with both ¼-inch jacks and XLR-type connectors of both sexes to facilitate linking several amplifiers together. Stereo, mono, and bridging modes are user-selectable by means of rear panel switches. Two 10-segment, 27 dB-range LED meter displays provide an easily viewed indication of power levels.

ASHLY AUDIO, INC.
100 FERNWOOD AVENUE
ROCHESTER, NY 14621
(716) 544-5191

For additional information circle #83

NEW MEDUSA POWER SERIES MULTICABLE SYSTEM FROM WHIRLWIND

Designed to let the user reap the maximum benefits from the newer powered mixing console systems, the Medusa Power Series system is described as being the first to combine both microphone and high-powered speaker cable into one custom, patent-applied-for multiple wiring system.



This new format enables the user to organize all of the cables into one main snake running from the stage back to the powered mixing board, resulting in a cost-effective, total wiring system that can reduce set-up time and eliminate messy wiring.

The new Medusa Power Series also features 14-gauge speaker wire, nickel-plated brass connectors, a choice of 6, 8, or 12 inputs, and is available in a variety of custom-designed configurations.

WHIRLWIND MUSIC, INC.
100 BOXART STREET
ROCHESTER, NY 14603
(716) 663-8820

For additional information circle #84

ROLAND SDE-2000 DIGITAL DELAY

Featuring delay times adjustable from 0 to 640 milliseconds, the SDE-2000 contains a large capacity memory system (16K-Byte) and a high-resolution (12-bit) A/D converter, to provide both low-noise operation and high-fidelity sound with full bandwidth.

Delay times are selected by two switches that automatically vary the delay times either up or

down in 1 mS increments. A Double (×2) switch enables any selected delay time to be doubled (up to 640 mS) for quick access. The LED display provides visual indication of the actual delay time set for precise, repeatable performance.

The unique Modulation section on the SDE-2000 provides two waveforms: the standard sinewave is available for doubling and flanging effects, while the new triangle waveform is used for the popular chorus effect. Other modulation controls include Rate and Depth, and an on/off switch which enables the modulator to be switched in and out easily for special effects.

The SDE-2000 has a recommended retail price of $1,150.00.

ROLAND STUDIO SYSTEMS
2401 SAYBROOK AVENUE
LOS ANGELES, CA 90040
(213) 685-5141

For additional information circle #85

JBL 2425 HIGH FREQUENCY COMPRESSION DRIVER

Equipped with a pure titanium diaphragm in combination with JBL's exclusive diamond-pattern surround, the 2425 was engineered with the benefit of laser interferometry in the company's Northridge, California-based research and development facility.

Described as an ideal device for both high-power sound reinforcement and custom studio monitor installations, the 1-inch throat diameter 2425 incorporates JBL's unique ferrite magnet structure, for extended frequency response with minimal distortion and greatest efficiency. The motor assembly also includes a 1¾-inch, edgewound aluminum ribbon voice coil.

Power capacity is a quoted 70W continuous program at 800 Hz, 12 dB per octave slope; and 100W continuous program at 1.2 kHz or higher, 18 dB per octave slope. Frequency range is 800 Hz to 20 kHz.

JAMES B. LANSING, INC.
8500 BALBOA BLVD.
NORTHRIDGE, CA 91329
(213) 893-8411

For additional information circle #86

dbx ANNOUNCES MODEL 180 TYPE I NOISE REDUCTION SYSTEM

The Model 180 provides two channels of encode electronics and two channels of decode electronics, and is intended for use with professional two-track tape machines.

According to Lance Korthals, dbx director of marketing and sales, professional products, "The new noise reduction system will produce a stereo master tape which fully preserves the dynamic range of live music and is completely free of audible tape hiss, as well as distortion due to tape saturation."

The 180 System is designed for installation between the console or mike mixer, and the line level inputs to the tape machine. It is described as compact and light in weight, and can easily be taken into the field on remotes or location jobs. The separate encode and decode electronics permit decoded monitoring off tape of the signal being tested.

dbx, INC.
71 CHAPEL STREET
NEWTON, MA 02195
(617) 964-3210

For additional information circle #87

EAW MID-BASS CABINET WITH NOVEL THROAT DESIGN

The new MR-102 Mid-Bass Reproducer from Eastern Acoustic Works uses a true exponential horn to provide high acoustic output at low distortion. Key to the exceptionally clean performance of the MR-102 is said to be a proprietary displacement plug that maintains a true exponential expansion rate throughout the horn's critical throat region, and eliminates the distortion heard as harshness or "honkiness" that plagues most existing horn-based systems.

Suggested for use in the 200 Hz to 1.2 kHz range, the MR-102's 3 dB points are 150 Hz and 1500 Hz, and it delivers 107 dB SPL at one meter and one watt input when loaded with EAW's standard LF-301-R 12-inch driver. This choice of driver and crossover points relieves the neighboring mid-range compression driver of much of its work-load, and reduces or eliminates mid-range driver failures in most systems.

The MR-102 Reproducer is packaged in laminated birch hardwood for mechanical ruggedness in the field, as well as freedom from resonances. All cabinet voids are filled with anti-resonance foam. The unit measures 19- by 29- by 25-inches deep, and weighs 110 lbs. with driver installed; unit price is $586.00.

EASTERN ACOUSTIC WORKS
59 FOUNTAIN STREET
FRAMINGHAM, MA 01701
(617) 620-1478

For additional information circle #88

LEXICON UNVEILS MODEL 224-XDIGITAL REVERB

The new unit combines all the features of the standard Model 224, and adds 15 kHz

bandwidth, more processing power, and more flexible effects. Model 224-X's full bandwidth is said to make most programs sound smoother and more natural, while opening a new generation of effects. High-quality reverb or effects can be used alone in a mix without the "direct" for a whole new sound. A variable bandwidth control allows 224-X to reduce its bandwidth with a natural 6 dB octave slope to better mimic the effects of absorption in air.

A new feature, "Dynamic Decay," provides the intelligence to automatically switch to a different reverb time when the music pauses or stops, allowing long sustain or decay without muddying continuous music. A paging system allows the six sliders on the remote panel to be redefined to control additional features, while retaining the familiar pattern for most-used reverberation adjustments.

All the features of the standard Model 224 reverb have been retained in the 224-X, including the capability for inexpensive field program updates; true stereo operation (two inputs and four outputs); reverb times from 0.6 to 70 seconds; 36 user program storage registers with single button recall; and the most natural synthesis of concert halls, chambers and plates available.

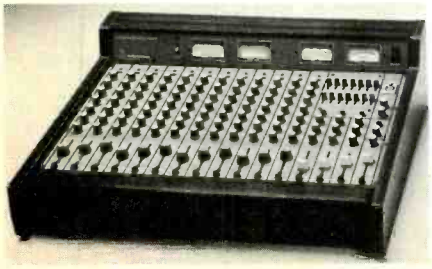
The 224-X does not replace the widely used Model 224; Lexicon will market and support both models as a series. The 224-X, priced about 40% higher, becomes Lexicon's top of the line digital reverb product, and is intended for those who require the "ultimate" in reverb capabilities and performance. Model 224-X will be priced at $12,000, with initial deliveries to be made in March 1982.

LEXICON, INC.
60 TURNER STREET
WALTHAM, MA 02154
(617) 891-6790

For additional information circle #89

AUDIO-TECHNICA INTRODUCES TWO NEW STAGE CONSOLES

The ATC820 and ATC1220 are 8- and 12-channel stereo consoles that feature phantom-power on all microphone channels. Each channel has its own high-, mid- and low-frequency equalization section. The program output is equipped with a dual-channel graphic equalizer, while the monitor output has a mono graphic equalizer.

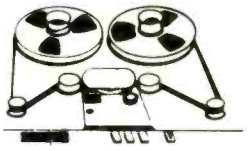


An important feature in the new consoles, according to professional products national sales manager, Charlie Winkler, is the provision of transformer-balanced inputs and outputs that reduce the possibility of extraneous noise pickup. Additional features offered by both models include filters to eliminate "popping"; a talkback system which allows the board operator to provide directions or musical fill-in through the stage monitor output; accurate, easily read peak level meters; line as well as microphone inputs; direct outputs; effects and monitor busses, built-in headphone amplification; and stacking inputs.

The ATC820 8-channel stereo console will be offered at a pro net price of $1,495.00, and the ATC1220 12-channel model will carry a suggested price tag of $1.995.00.

AUDIO-TECHNICA U.S., INC.
1221 COMMERCE DRIVE
STOW, OH 44224
(216) 686-2600

For additional information circle #90



SIMPLIFYING THE AUDIO SYSTEMS INTERFACE: The TUCHEL CONNECTOR

by Larry Lamoray
MCI, a Division of
Sony Corp of America

Those who have ever installed a console with input/output connections made via "christmas trees," screw terminals, telephone-type punch blocks, etc., or have tried to troubleshoot system problems when using such connectors, are well aware of their shortcomings. The frustration of trying to neatly dress wires while ensuring reliable connection, or isolation of hum-producing ground loops and intermittent problems, can rattle even the best of technicians or engineers. Use of XLR-type connectors certainly simplifies much of the troubleshooting process, but they are expensive.

Enter the "Tuchel"* connector. This connector, now being utilized by some U.S. equipment manufacturers, has been widely used in many audio applications throughout the European broadcast community for several years now. It combines high reliability and convenience with comparatively low cost.

The Tuchel connector currently used for multichannel connections is a 30-pin type with an asymmetrical pattern per DIN specifications (DIN 41622). The electrical connection is made via wiping, low insertion force contacts of nickel/gold or hard silver plating (gold-plated contacts are also available). Protective shells for the mating half are made of rugged high-impact polyamid plastic with integral strain relief, metal shells also being available.

Since the connector is configured in 10 rows of three pins each, it lends itself to audio applications. Several manufacturers have mutually agreed on its wiring configuration to provide easier interface between different pieces of equipment. Among these are: MCI, who use the Tuchel as standard connectors on their JH-24 Series multitrack recorders, and JH-500D and JH-600 Series consoles; Harrison Systems, Inc., who offer them as an option on all of its recording consoles (standard on the model MR-2 when ordered less its optional patchbay); and Leunig GmbH, who use the Tuchel on the Param automated equalizer. This wiring configuration provides for standard connectors each including four line inputs and four line outputs, eight microphone inputs; or connection to peripheral equipment within a small area (0.75 by 2.7 inches), and is fast becoming a de facto standard for the audio industry.

Use of Tuchel connectors offers many advantages.

• The standard pin configuration allows construction of cables prior to system installation, therefore shortening labor requirements on site.

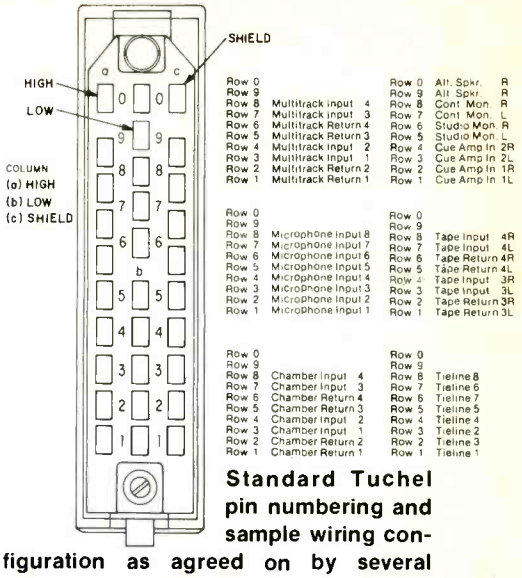
• Crossing connections to facilitate easy troubleshooting becomes a matter of merely unplugging and replugging cables; in fact a whole system can be connected or disconnected in minutes.

• Transferring equipment between studios or interfacing extra equipment becomes far easier.

• Several equipment installers report connection of outboard switcher networks, remote patchbays and various other pieces of outboard gear far easier due to use of Tuchel connectors.

• For those manufacturers or installers performing high volume connections, Tuchel pins are available with solder tabs, crimp, or wire-wrap termination. A complete line of hand and mass termination tools are also available.

As equipment requirements of the audio industry continues to grow, use of a standard connector and wiring configuration can only make the job of equipping studios a little easier.



Standard Tuchel pin numbering and sample wiring configuration as agreed on by several manufacturers.

Footnote:

*Tuchel-type connectors are available in the U.S. from Bunker Ramo and Contact Electronics, Inc. "Tuchel" is a colloquialism derived from Ampenol-Tuchel Electronics GmbH, a division of Bunker Ramo.

■■■

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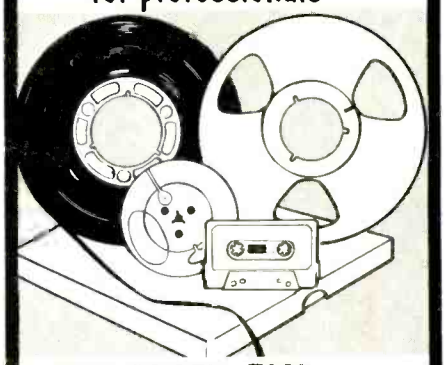

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☐ **ELECTRIC LADY STUDIOS** (New York City) has completed acoustic modifications to Mixing Suite 'C.' Project was finished in two weeks by Alangrove Builders Ltd., of London, England. *52 W. Eighth Street, New York City, NY 10011. (212) 477-7500.*

☐ **JAY ROSE/SOUND, INC.** (Boston, Massachusetts) has added a SMPTE-based dialogue relacement system for video and mag film. The computer-controlled system complements two fully equipped production studios, music and effects library, and duplication facility. *480 Boylston Street, Boston, MA 02116 (617) 247-1141.*

☐ **NATIONAL VIDEO CENTER & RECORDING STUDIOS** (New York City) has installed an Audio Kinetics Qlock 310 SMPTE time-code synchronizer in its 24-track studio. The new unit will be used to lock up a 24-track recorder to either 1-inch or ¼-inch VTR's. *460 West 42nd Street, New York City, NY 10036. (212) 279-2000.*

☐ **NORTHERN LITES AND SOUND STUDIOS** (Georgia, Vermont) has recently upgraded its facility to 8-track with the addition of a TEAC Tascam 80-8 and DX-8 dbx noise reduction package. **Tracy Lord** is the studio's owner/engineer. *36 Cedarwood Terrace, Georgia, VT 05468 (802) 893-1220.*

☐ **OMNI RECORDING STUDIOS,** (Morristown, New Jersey) has expanded to 16-track operation. Upgrade hardware includes an Ampex MM1000 and an Eventide Harmonizer. Other studio facilities include Big Reds, Little Reds, and Auratones; Crown and Marantz amps; Neumann, Sennheiser and Shure mikes; Steinway Grand piano, Slingerland drums, and guitar amps. *44 Abbett Avenue, Morristown, NJ 07960. (201) 539-8804.*

☐ **PRESENCE STUDIOS** (West Haven, Connecticut) tell us that the facility recently "went a little nuts," and purchased an Ampex ATR-800 2-track with remote, two Dolby 361 Type A noise reduction units, a Neuman U87 microphone with shockmount, an Oberhiem OBX-A Polyphonic Synthesizer, two dbx Model 903 noise gates, and a Eumig FL1000YP cassette deck, according to chief engineer **Jon Russell**. *17 Enfield Street, West Haven, CT 06516 (203) 397-8682.*

☐ **SELECT SOUND STUDIO** (Kenmore, New York) has moved into new, expanded facilities designed by **Lakeside Associates** of Mission Viejo, California. Equipment highlights include: Lakeside's monitor system, powered by Crown PSA-2 and DC 300 amplifiers, and equalized by White 4001's; MCI tape transports fed from an Allen & Heath Syncon Board (28 × 28, parametric equalizers, 14 stereo subgroups); EMT 240, Lexicon Prime Time, Kepex II, and UREI compressor/limiters for signal processing; and a full complement of microphones, including Neumann U87's, Sony C37's, Sennheiser 421's and 441's, various AKG's and miscellaneous others. The studio was entirely built by Select Sound in-house staff and local labor, with lots of personal assistance from designers Carl Yanchar and Steve Fouce from Lakeside Associates. *2315 Elmwood Avenue, Kenmore, NY 14217. (716) 873-2717.*

☐ **SOUND SHOP** (New York City) has reopened Studio D, which has been refurbished and new equipment added. Included in the modernization are a Foley stage for producing live sound effects, and a new ADR synchronized dubbing system with custom-built console. Sound Shop provides full sound services for both film and videotape, according to studio president **Emil Neroda**. *304 East 44th Street, New York City, NY 10017. (212) 573-6777.*

☐ **TROD NOSSEL RECORDING STUDIOS** (Wallingford, Connecticut) recently installed three Pultec EQP-1 tube parametric equalizers, a Pultec EQH-2 tube equalizer, a Hycor 4201 equalizer, and a UREI 530 stereo graphic used for room EQ. *10 George Street, Wallingford, CT 06492. (203) 269-4465.*

Southeast:

☐ **MUSIC MILL** (Nashville, Tennessee) has installed a new Trident TSM 24-track console, complete with Valley People Fadex 65K automation, which offers programmable muting and nine assignable VCA groups. The console features graphic equalization and effects filters on each input, plus shelving equalization on each monitor channel and echo return. The monitor section is a separate mixing section located to the right of the inputs. Each monitor channel can be assigned to the stereo mix, thereby providing a total of 52 inputs for mixdown. Owners **Donny Canada** and **Harold Shedd** immediately used the new console to record and mix the new album by **Alabama**. *1526 Laurel, Nashville, TN 37203. (615) 254-5925.*

☐ **SOUND EMPORIUM STUDIOS** (Nashville, Tennessee) has recently purchased a Studer A-80 half-inch, 2-track mastering system. According to studio president, **Jim Williamson**, "The system is the newest innovation in analog recording. Compared to quarter-inch mastering, the half-inch machine reduces tape noise and enhances all facilties of the analog signal." *25 Music Square East, Nashville, TN 37203. (615) 244-1060.*

☐ **LITTLE CARIBOU STUDIO** (Orlando, Florida) formerly the Demo Factory, has recently completed remodelling and installation of a Neotek Series I console supplied by Valley People. Installation was performed by Tim Trott Audio, Sanford, Florida. Studio owner/producer **Brian Little** is presently engaged in a local album project. An Eventide Harmonizer™ and AKG BX-5 spring reverb were also installed, along with an Otari MX-5050B mastering machine. *Orlando, FL.*

South Central:

☐ **THE BOB O'NEILL SOUND STUDIO** (San Antonio, Texas), designed by **Woody Smith** of Abadon/Sun Pro Audio, recently celebrated its grand opening. Studio owner **Bob O'Neill** reports that the equipment purchase includes a TEAC Tascam 85-16 16-track recorder, Soundcraft console, JBL monitors, Crown amplifiers and Ampex mastering machines. *3014 Broadway, San Antonio, TX 78209. (512) 824-7698.*

Midwest:

☐ **PYRAMID AUDIO** (South Holland, Illinois) is moving to a larger full-service audio complex. The new studio will feature a 24-track Otari MTR-90, Neotek Series III console, and a full complement of microphones and outboard gear. Complete sound system/recording equipment rental and leasing, rehearsal studios, showcase/presentation room with video facilities, equipment sales and service, and sound system design and installation will also be available. The new facility is scheduled to open in mid-March. *16240 Prince Drive, South Holland, IL 60473. (312) 339-8014.*

☐ **CHICAGO TRAC,** (Chicago, Illinois) recently upgraded its facilities with the addition of a new Otari MTR-90 multitrack and Studio Technologies Ecoplate reverb unit. Studio managers are **Reid Hyams** and **Tom Kee**, and chief engineer is **Al Ursini**. *2656 N. Wayne Avenue, Chicago, IL 60614. (312) 525-6565.*

☐ **AUDIO VILLAGE RECORDING STUDIO** (Bloomington, Indiana) has installed a new transformerless MCI JH110B mixdown machine, Kepex II and Gain Brain compressor/limiters, and Delta Lab digital delay. In addition, the studio's ADS 91011 monitors are now augmented with UREI/Altec 604's, and a McIntosh tube amplifier is available, as well as the 250W solid-state amp. *1000 West 17th Street, Bloomington, IN 47402. (812) 332-7475.*

☐ **QCA RECORDING STUDIOS** (Cincinnati, Ohio) has recently added a Harrison 28/24 MR-3 console featuring Allison automation, transformerless inputs, and 28 channels of true parametric equalization. Studio manager is **Jim Bosken**. *2832 Spring Grove Avenue, Cincinnati, OH 45225. (513) 681-8400.*

☐ **MAUER BROTHERS' RECORDING** (Milwaukee, Wisconsin) has opened a new 8-track featuring TEAC Tascam 8-, 4-, and 2-track recorders fed by a new Neotek Series 1E 16/8 mixing console. Monitors are custom designed using JBL components powered by SAE amplifiers.

— continued overleaf . . .

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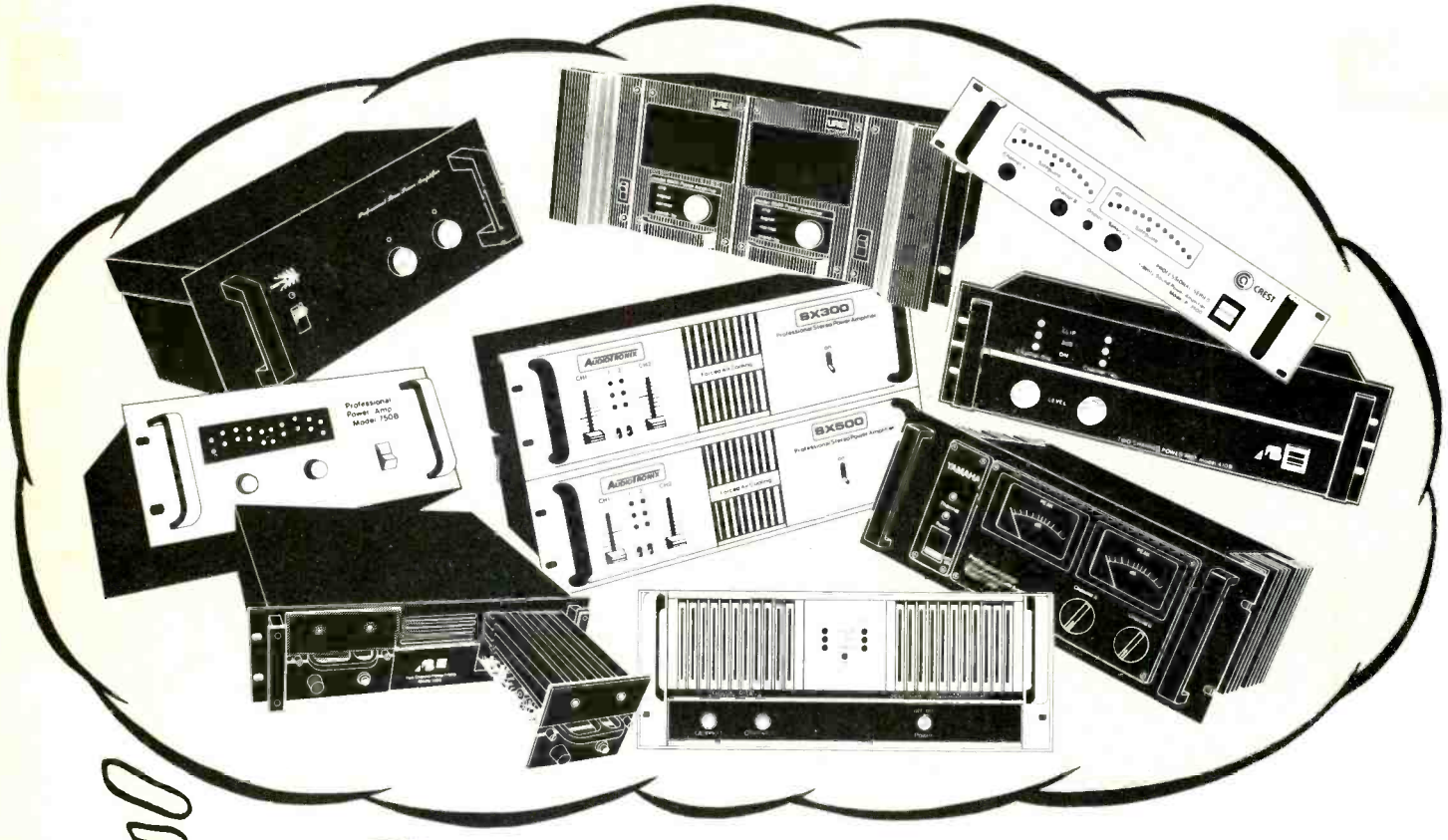

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STUDIO FACILITIES EQUIPMENT PEOPLE UPDATE

Midwest . . . continued —

Outboard equipment includes an Ursa Major space Station, MXR and NEI equalizers, and Roland Stereo Phase. Mikes are by AKG, Shure, E-V, Audio-Technica and Teac. Owners are **Mark** and **Charlie Mauer**. *10533 West Silver Spring Drive, Milwaukee, WI 53225*

Mountain:

☐ **ROSEWOOD RECORDING COMPANY** (Provo, Utah) owner **Guy Randle** reports the installation of a Tangent 24-channel console, an Orban de-esser and a Symetrix compressor/limiter. A nine-foot Baldwin grand piano has also been added to the studio's new facility. *186 West Center, Suite 2, Provo, UT 84601. (801) 375-5764.*

Southern California:

☐ **ARTISAN SOUND RECORDERS** (Hollywood) has had a major redesign of the building to accommodate an acoustically "live" room by Sierra/Eastlake. The new Studio C control room is said to be the first in the world to utilize the Pioneer T.A.D. 1601A high-compliance woofer. Sierra/Eastlake worked closely with Pioneer's Bart Locanthi in development of fitting the new transducers to the Sierra/Hidley system. Joining mastering director **Gregory Fulginiti** is new chief engineer **Jim Stern**. *1600 N Wilcox, Hollywood, CA 90028. (213) 461-2751.*

☐ **GOLD STAR RECORDING** (Hollywood) has replaced its custom board in Studio B with a Trident TSM console, in an effort to "attract independent engineers and their projects to Gold Star." Studio A still houses a Gold Star Custom board, designed by owner **Dave Gold.** *6252 Santa Monica Boulevard, Hollywood, CA 90038. (213) 469-1173.*

☐ **RECORDING SERVICES COMPANY** (Studio City) recently added an Audio Kinetics Qlock 310 SMPTE time-code synchronizer to its list of rental equipment. *10824 Ventura Boulevard, Studio City, CA 91604. (213) 766-7191.*

☐ **SFX MUSIC/SOUND DESIGN** (Los Angeles), an audio/visual facility offering frame-accurate time-code audio editing, sound synthesis and electronic music, has moved to the Lion's Gate complex. *1861 S. Bundy Drive, West Los Angeles, CA 90025. (213) 820-8728.*

☐ **TOPANGA SOUND RECORDERS** (Topanga), the former Keys Sound Recorders in Islamorada Florida, has changed its name and relocated to California. Owner **Christopher Dinneen** alsoannounced the acquisition of a Synclavier II digital synthesizer. *Topanga, CA (213) 455-3461.*

☐ **THE SOUND CHAMBER** (Pasadena) studio manager, **Dick McIlvery**, announced the upgrading from 16- to 24-tracks, with the addition of a Stephens 821A 24-track tape recorder, and expansion of its Tangent 3216 console to 24-tracks. Also added is a MICMIX Master Room XL305 Reverb, MXR Digital Delay, MXR Flanger/Doubler, Orban Stereo Parametric Equalizer, Moog 12-stage Phase Shifter, and dbx compressers. *27 South El Molino, Pasadena, CA 91101. (213) 449-8133.*

☐ **MOTIONPICTURE RECORDING** (Hollywood) is said to be the first new dubbing company to be set up in Hollywood in the past ten years. Owner **Garry Ulmer** chose **Everything Audio** for technical and acoustic design, and equipment interface. The computerized console was built to Ulmer's specifications by Amek in England. High-speed projection system and film recorders and reproducers were built by RCA Photophone. *7060 Hollywood Boulevard, Hollywood, CA 90028. (213) 462-6897.*

☐ **THE SUNSET SOUND FACTORY** (Hollywood), formerly The Sound Factory, has re-opened for business following its purchase by

Sunset Sound. Studio A, the room responsible for many Platinum and Gold disks, is currently being renovated. **Richard Goldblatt**, has been appointed general manager of the facility, and is directing current alterations. "We intend to keep Studio A basically the same as it was," he says. It would be foolish to change something that had a good track record. We are completely renovating the API console, to bring it up to first-class technical specs. We are replacing the present tape machines with Ampex multitracks, and adding the option of room EQ." Goldblatt also intends to review the control room acoustics of Studio B, and change whatever is required to bring them to acceptable standards. *6357 Selma Avenue, Hollywood, CA 90028. (213) 467-2500.*

Northern California:

☐ **LIKEWISE PRODUCTIONS** (Berkeley) is now operating as a full 16-track facility. Equipment includes a SoundWorkshop Series 20 board, TEAC Tascam 85-16 recorder with dbx, Tannoy monitors, Lexicon DDL, MICMIX Master Room reverb, and a large assortment of microphones. Staff engineers are **Jeffry Holt, Greg Spinosi, Matt Brady**, and **Randy Statman.** *P.O. Box 5447, Berkeley, CA 94705. (415) 654-3112.*

☐ **HYDE STREET STUDIOS** (San Francisco) new Studio D features a 40-input B Series Trident board, an Otari MTR-90 24-track, and an MTR-10 2-track. Other new equipment includes an Eventide H949 Harmonizer, a pair of Teletronix LA-1A tube limiters, and two Telefunken Elam-251 microphones. **Don Kruse** has joined Hyde Street's full-time maintenance department, handling all types of repair duties, along with his speciality, Otari tape recorders. *245 Hyde Street, San Francisco, CA 94102. (415) 441-8934.*

☐ **THE AUTOMATT** (San Francisco) has completed renovation of the Automatt Complex with the addition in Studio A of a new John Meyer A.C.D. monitor system and expansion of the control room. Owner **David Rubinson** says: "With Studio A being one of the finest rooms in the country, having the A.C.D. John Meyer Monitor System permanently installed in the control room was a natural for the Automatt and our state-of-the-art system." *827 Folsom Street, San Francisco, CA 94107. (415) 777-2930.*

☐ **RECORD PLANT** (Sausalito) has completed of a quarter of a million dollar remodeling and hardware update program in Studio B. Thirty percent additional floorspace is now available, along with adjustable surface treatments to allow for tuning of room reverb characteristics. Hardware acquisitions include a 40/32 Trident TSM console with Melkquist floppy disk automation, and a John Meyers A.C.D. monitoring system. Hidley/Westlake monitors also remain available. *2200 Bridgeway, , Sausalito, CA 94965. (415) 332-6100.*

☐ **BODACIOUS AUDIO** (San Mateo) is now using the Sony PCM-1610 digital mastering system in its mobile units, in an effort to improve the quality of radio rebroadcasts, or audio and video demo projects. The studio believes that the new Sony digital system will help in "capturing the elusive 'live' electricity of a concert." *4114 George Avenue, Number 1, San Mateo, CA 94403. (415) 573-573-5297.*

Canada:

☐ **OUCHARD STUDIO** (Nowal, Ontario), situated on a 100 acre farm near Toronto, is an 8-track facility equipped with a TEAC Tascam 80-8 with dbx noise reduction, and Tascam 4- and 2-track recording machines. Console is a Soundcraft Series II 16/8. Available monitors are Tannoy Ardens, JBL 4311's or Auratones. The studio is said to provide a good selection of keyboards, including a new 7-foot Heintzman Grand Piano, and a collection of vintage guitars. *RR#2, Nowal, Ontario, Canada L0P 1K0.*

☐ **ROUND SOUND STUDIOS** (Weston, Ontario) has recently equipped it's 16-track facility with new synthesizer equipment from Roland.

continued overleaf . . .

Canada . . . continued —

The system consists of the Roland Jupiter-8 eight-voice polyphonic synthesizer, with multi-voice programmable MC-4 Microcomposer, and the TR808 programmable rhythm unit. President **James Sutherland** says equipment will be used to further Round Sound's expansion into the audio/visual, film and advertising markets. *357 Ormont Drive, Weston, Ontario, Canada M9L 1N8. (416) 743-9979.*

England:

☐ **THE MANOR** (Shipton-on-Cherwell, Oxfordshire) is redesigning its control room to include a Solid State Logic 4000 E Series Master Studio System. Tom Hidley, who was responsible for the original acoustic design, will be supervising the control room update. Studio owner, **Richard Branson,** says "We bought the first UK-based B Series SSL console for Townhouse 2 in 1978, and when you look at the reaction of the producers and artists who use it, and the number of hits that have been cut on it, the choice was obvious." The new 40-input console features a limiter/compressor/noisegate/expander on each channel, and will also be fitted with SSL's Primary Studio Computer. *Shipton-on-Cherwell, Oxfordshire, England.*

☐ **SARM STUDIOS** (London) is installing a Solid State Logic 4000 E Series Master Studio system, complete with the SSL Studio Computer and Total Recall. SARM Studio's owner, Jill Sinclair says, "Computer control is the way of the future, and we want a control room which will handle not just our current techniques, but one that will take us into the Digital Disk era, with the capability to extend synchronization and computer control to video as well as audio transports." *Osborn House, 9-13 Osborn Street, London E1. 01-247 1311.*

Switzerland:

☐ **BLUE CEDAR** (Corcelles s. Chavornay), a new studio near Montreux, is equipped with TEAC Tascam 8- and 4-track recorders, and a Tascam Model 15 console. Monitoring comprises a Studer A68 powering E-V Sentry III cabinets. Effects units include a Great British Spring reverb, custom-built reverb, and Binson echo. Outboard gear includes ADR Stereo Compex and Scamp rack, custom stereo phaser, ADR voiceover limiter, ES-560 selective limiter, and an EXR Exciter. Tom Hidley of Sierra/Eastlake is said to have lent a hand in the specification of basic acoustics for the control room, and advised on the rest of the design. According to co-owner **Terry Nelson,** the studio expects to take delivery of a Studer C37 mastering transport equipped with CB Electronics record-replay amps during early 1982. *Corcelles s. Chavornay, Switzerland. c/o (24) 21 36 49.*

— AUDIO/VIDEO UPDATE —

Eastern:

☐ **VIDEO COMMUNICATION SERVICES** (Riverton, New Jersey) president **Frank Siegel** has announced that the company, which has been involved in broadcast commercials and industrial shoots up until this time, is now gearing up to do multi-camera music demos for acts in the Pennsylvania, New Jersey, Delaware, and Washington D.C. areas. A remote truck will handle two to five cameras recording on ¾- or 1-inch type C format. *208 Linden Avenue, Riverton, NJ 08077. (609) 786-1775.*

For additional information circle #104

☐ **NATIONAL VIDEO CENTER & RECORDING STUDIOS** (New York City) boasts completion of audio post-production of "Philadelphia Folk Festival 1981," employing their Audio Kinetics Qlock 310 SMPTE synchronization unit to mix the multi-track audio derived from live recordings of the festival. The Qlock 310 permits up to three audio machines to be enslaved, providing four interlocked units. *460 West 42nd Street, New York City, NY 10036. (212) 279-2000.*

Central:

☐ **KLUGE COMMUNICATIONS** (Milwaukee, Wisconsin) is adding video to its line-up of audio/visual post-production services by installing a 1-inch computerized editing suite. Full transfer capabilities to and from videotape in 1-inch, ¾-inch and ½-inch formats will be available. Full operation of the video facilities is expected by April 1982. *5350 West Clinton Avenue, Milwaukee, WI 53223. (414) 354-9490.*

Western

☐ **SOUND SMITH STUDIOS** (Portland, Oregon) has assembled, edited and mixed what **Ed Palmer**, national director of Professional Development for the International Television Association (ITVA) believes is the first industrial video show with stereo sound to be reported to the organization. Sound Smith used the facility's new BTX Shadow SMPTE Interlock system. Stereo music from records and tapes was transferred to a time coded ½-inch 8-track audio tape on an Otari MX-5050-8D with dbx. The music, sound effects and narration (also on coded tapes) were assembled on 2-inch time coded audio tape with an Ampex MM1200. The 2-inch tape was then mixed down to stereo, and transferred to the Otari. The final program was subsequently transferred to 2-inch video tape with an Ampex VPR 20 portable video recorder, and laid-back to the 1-inch video master on an Ampex VPR 1C at Magaurn Video. *426 N.W. 6th Avenue, Portland, OR 97209. (502) 224-7680.*

☐ **THE VIDTRONICS COMPANY, INC.** (Newbury Park, California) has purchased a total of five Sony Broadcast BVH-1180 VTR's for its videocassette duplication and post-production divisions. Vidtronics' recently formed videocassette division is claimed to be the largest in the world, with an annual duplication capacity of 7,000,000 cassettes to be reached this year. *Newbury Park, CA*

☐ **TRANS-AMERICAN VIDEO** (Las Vegas, Nevada) officially introduced "#740," a 40-foot unit, as the newest member of its mobile fleet, joining the company's 10- and 20-foot units. The new unit contains a Grass Valley 1600-3H production switcher, four Ampex VPR II-B VTR's (with Slo-Mo), portable 1-inch VTR's, and a Quantel DPE-5000 digitial effects unit. Audio equipment includes a Yahama PM2000 console, Telex cartridge machines, RTS intercom and communications system, and a 12-channel IFB. *3349 South Highland Drive, Las Vegas, NV 89109. (702) 733-2922.*

☐ **LION'S GATE FILMS** (Los Angeles, California) has unveiled the first working prototype of a VARMS (Video/Audio Recording and Mixing Stage). Designer John Storyk, executive vice president of Metropolis Studios, says, "The VARMS is a technologically advanced studio providing space for simulltaneous video/audio recording. It is the backbone for integrated format production, providing ultimate, true integration of film or video with audio. The design of the VARMS permits recording of broadcast-quality video simultaneously with digital audio, to provide state-of-the-art capabilities, that, among other things, are essential for video disk production." *Los Angeles, CA*

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book review

THE MASTER HANDBOOK OF ACOUSTICS

by F. Alton Everest

353 Pages; $15.00 (paperback) and $21.00 (hardback), including postage

F. Alton Everest's latest book is an easily readable, concise, and up to date presentation of the acoustical factors important to good listening conditions. Its title would lead you to expect a coverage of most if not all aspects of acoustics. Although this is not the case, it should not be taken as a harsh criticism. The subject of acoustics has *SOUND* as its common denominator; it encompasses such topics as speech, hearing, music and musical instruments, transducers, noise, ultrasonics, vibration, and more. These areas are well covered elsewhere.

The Master Handbook does deal, however, with such timely concepts as TDS™ (Time Delay Spectrometry), LEDE™ (Live-End/Dead-End) control room design, and PZM™ microphones. There is an entire chapter devoted to acoustical comb filter effects. The usual basics of frequency, wavelength, "the Decibel," and how the ear works are basically basic — enough for the unknowing to understand these important concepts, yet not too much for the initiated to be bored. Somehow it's always the easy things we think we fully understand that always get us.

The principles of sound propagation, indoors and out, echoes, reverberation, absorption, diffusion, and the ever-present room modes, are discussed in an easily understandable manner. Once again, it is *concepts* that are stressed — the math involved is only the minimum.

Old ideas with newfound applications, such as the Haas Effect, are explained, as are new developments, such as Schroeder's Mechanisms for obtaining diffusion. A chapter is devoted to some of the latest instrumentation for acoustical measurements, although the most important — Heyser's Time Delay Spectrometry — is given but a paragraph.

One chapter discusses the importance of quiet air conditioning. Low internally generated noise levels have always been a consideration in the design of listening rooms, control rooms, and studios, and will become even more critical as digital recording's 90 dB-plus signal-to-noise ratios come into play.

Most of the balance of the book illustrates how acoustic principles are applied to the design of a listening room, studio, or control room. Several of the current design theories are explained very objectively. The section on the Nippon Gakki's psychoacoustical experiments sheds some light on the basis for these seemingly contradictory theories. Unfortunately, one of the most important contributors to modern control room and studio design, Tom Hidley, of Sierra/Eastlake, is not mentioned.

The Master Handbook provides the essentials to understanding how rooms affect the sound we hear. The list of references is an excellent basis for more detailed information.

In a section on room proportions, Mr. Everest says: "The perfect room proportions have yet to be found. It is easy to place undue emphasis on a mechanical factor such as this. The reader is urged to be informed on this subject . . . and to be aware of certain consequences. But let us be realistic about it — all of the recording that has ever taken place has been done in spaces less than perfect. The point is that striving to upgrade sound quality at every stage of the process . . . is just good sense."

Carl Yancher

"THE MASTER HANDBOOK OF ACOUSTICS"
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news

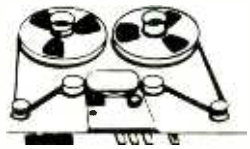
RECESSION MEANS "GOOD DEALS," ACCORDING TO SEATTLE PRO-AUDIO DEALER

The current recession may be a "good deal for people looking for good deals," says Tom Martin of Pro Audio Systems. "I'm hearing about some real bargins in used recording equipment these days."

Recently Martin announced plans to publish a monthly "Used Equipment Newsletter." According to Martin, the new publication is an effort by Pro Audio Systems, the Seattle-based MCI dealer, to "keep up with the changing needs of the professional audio industry."

Martin says he will use an in-house computer to keep an "up-to-the-minute" listing of "who has what." He will be able to instantly match customer needs with equipment consigned with him.

The newsletter will be sent free of charge to audio/video production houses, broadcasters, and video studios, as well as to recording studio engineers and producers.



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- Full-function channel remote and microprocessor controlled autolocator with 20 address memory.

The Studer A80 ½-inch 2-Track Mastering Decks

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For additional information circle #106