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db MAGAZINE

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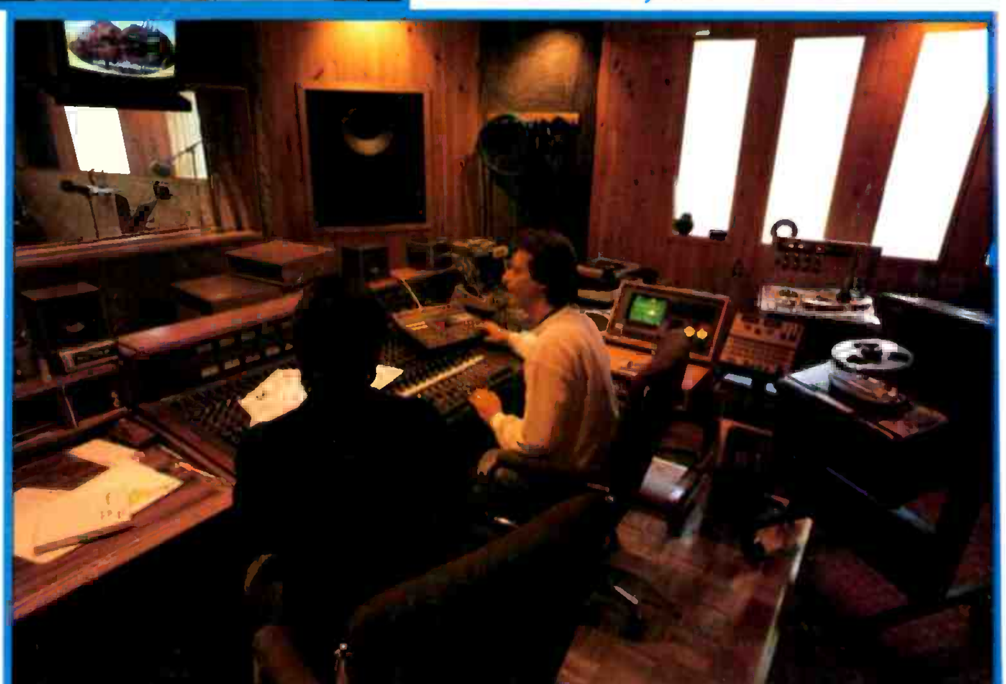
- Ed Learned Wanders South
- Lab Report: Tascam MSR-16 16- Tracker
- And the Usual Round of Stuff From— Battles, Bartlett & Barilla



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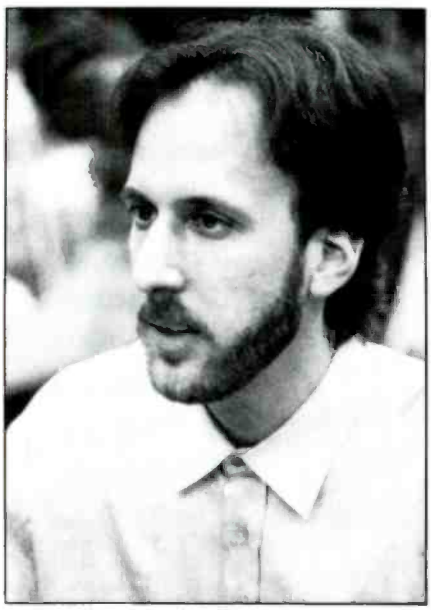

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About the Cover

- Streeterville Studios' Studio 3 (above) is better known as "the finishing suite." Technical director Steve Kusiciel is with the AMS AudioFile.

Universal Studios' "Backroom" (below) is a 48-track music scoring/post production room. Production mixers Bob Bennett and Mike Mason are at the MCI 628. The room is also equipped with an AMS AudioFile with 4 hours of memory.

Both studios are profiled in this issue.

db

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Calendar

• A four-week program, comprised of eight accredited graduate level courses in acoustics and signal processing, will be offered in June 1989 by **Penn State's Graduate Program in Acoustics** in cooperation with the University's Applied Research Laboratory (ARL) and the Research Center for Acoustics and Vibration Engineering (RCAVE). Courses offered include *Fundamentals of Acoustics, Underwater Sound Propagation, Digital Signal Processing, Electroacoustical Transducers, Acoustical Data Measurement and Analysis, and Intensity Technique.*

For further information contact: Dr. Alan D. Stuart, Summer Program Coordinator, the Penn State Graduate Program in Acoustics, PO Box 30, State College, PA, 16804, (814) 863-4128, or Mrs. Barbara Crocken at (814) 865-6364.

• The **APRS** (The Association of Professional Recording Studios) has now opened its bookings for stand space at the 22nd annual international exhibition for the recording industry, APRS 89. It will be held at Olympia exhibition centre, London, on Wednesday, June 7th to Friday, June 9th.

More details are available from: APRS, 163A High St, Rickmansworth, Herts. WD3 1AY, England. Telephone (STD 0923) 772907, (International) +44 923 772907.

• The New England Conservatory is offering summer music workshops. The course titles are *Musicians and Technology, Electronic Music Workshop*, and *Controlling Your MIDI*. The workshops will take place in June, 1989. For more information, contact Mary Street, Director of Summer School, (617) 262-1120, ext. 283.

• Synergetic Audio Concepts of Norman, Indiana, announces a series of special three day classes to be held at the Syn-Aud-Con farm in Southern Indiana. These classes feature the opportunity for gaining direct learning experiences for specific objectives in fundamental audio systems measurement and analysis.

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DC area (Rockville, MD): October 26-27

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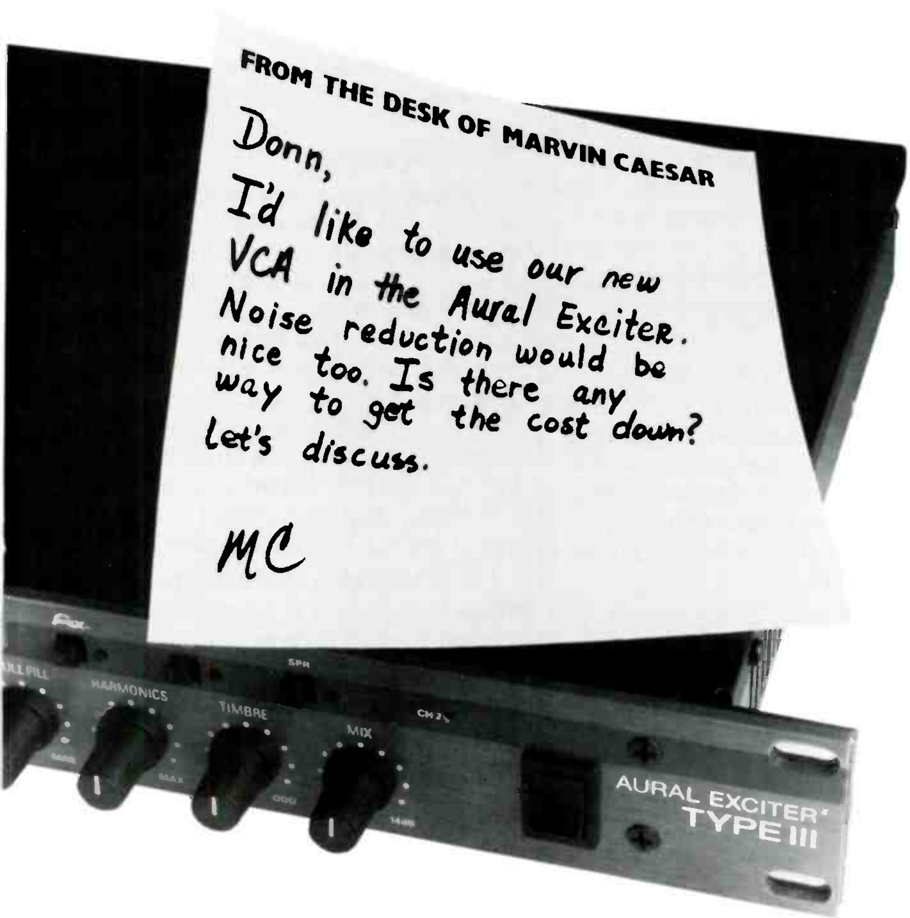

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db Visits the Windy City

UNIVERSAL RECORDING: THE HISTORY OF AUDIO FOR VIDEO

MURRAY R. ALLEN

Universal first became involved with mixing audio for video in 1957. In these early days all audio was mixed in the film format. Sprocket holes were the only method of synchronization. On rare occasions, phonograph turntables locked to a sel-syn motor could also be used as part of the synchronization chain.

The playback machines were the old RCA iron heads. A 60-second commercial would be looped. This means the picture, the music, the dialogue, and the sound effects tracks would be made into 118-foot loops and be individually played-back and/or projected on their own machines. Always in sync, the commercial would go around and around until the mix was perfected. Even the 35mm film stock on the master recorder was looped. When a perfect mix was attained, this same mix could instantly be played back in sync.

One of the problems with this method had to do with the inability to just fix one section of the commercial. The entire spot had to be mixed in total each time.

By 1970 we converted to film recorders and projectors that backed up. This made it possible to fix a tiny section at will without having to remix the entire commercial. Film recorders always have had the ability to punch in and out without leaving a hole. They were way ahead of regular tape recorder technology in this discipline.

In 1971 Ampex introduced the RA 4000 system. This system allowed the user to lock two MM-1000 (16-track) recorders together with an Ampex two-inch quad video recorder. There was no time code (it had not been invented yet). The tracks would have to be physically lined up, as in film. Once they were lined up they would run back and forth in sync.

THE UNIVERSAL STORY

We did our first location and what might be the first video sweetening session in history in 1972. We went to Evansville, Indiana, to record Margaret Whiting with the Cavalcade of Big Bands. We recorded the show on fourteen tracks. On track 16 we recorded vertical sync. Back at Universal we mixed down to a four-track recorder, copying the same vertical sync to track 4. Using the external code source on a Magnatech 92C resolver we locked the four-track to house vertical sync and transferred the mixed program to two-inch video tape. The editor then physically lined the track to picture and married the two together. This show (The Big Band Cavalcade) is being aired every Christmas on PBS. It still sounds good.

Since 1972 we have come a long way.

The one most important rule that Universal has followed throughout all its growth is that whatever is added in the way of equipment and services must meet the highest standards of the professional audio field. Nothing has been added just for hype.

Today Universal lists among its clients every other recording studio in Chicago, a couple of majors in New York and L.A., just about every major advertising agency in the world, all of the big five record companies as well as most of the major motion picture companies in Hollywood. Also, let's not forget the blue chip corporations and publishers. Universal also services those smaller winning companies that are nipping at the major's heels. It is impossible to turn on prime time TV and not hear several tracks recorded at Universal.

Universal is comprised of two film mixing theaters, four 32/48-track music scoring and video post-production studios, five 16-track music scoring and video post-production studios, one 8-track media studio, two Synclavier tapeless studios, four film/video transfer suites, one optical transfer suite, three tape copy rooms, a search and storage room for music libraries, and a zillion sound effects. Universal has enough space in its tape vaults to store over 25,000 sessions, meaning 32-track sessions.

PIECE BY PIECE

Universal has also been a leader in the area of digital recording. After eight year's experience in this field, it proudly boasts of owning 20 digital recorders encompassing several 32-track digital machines, a 16-track and 4-track digital machine, and several 2-track digital machines from such manufacturers as 3M, Sony, Otari, Mitsubishi, and New England Digital. In fact, Universal was the first studio in the world to develop a system whereby a digital recorder could talk to a Necam computer.

Let us take Universal apart piece by piece to see why it is the ultimate as a one-stop recording studio. Let us start with the film-mixing theaters.

Theaters One and Two are placed back-to-back, separated by a large machine room. Both theaters have multitrack ADM consoles with total machine control at the console. This includes start/stop and high-speed

Figure 1. Scoring in Universal's studio A.

shuttle controls, advance/retard controls, vari-speed in sync using Universal's own invention called Magwarp, all in/out 8-track record functions and BTX Softouch controls for non-sprocket/video synchronization. The consoles have four-band equalizers on every channel strip as well as programmable graphic equalization. Both theaters have 35mm, 16mm, and full-screen video projection. Naturally each theater has echo, several digital delay and digital reverb systems designed by AMS and Lexicon. There are also dip filters and compressors in both theaters as well as JVC 8250 video recorders. As a master recorder both theaters use an 8-track New England Digital Direct-to-Disk system.

The machine room contains twenty-seven Magna Tech high-speed dubbers and recorders. These machines have a total of fifty-seven heads giving the client the option of 6-track, 4-track, 3-track, 2-track, and/or monaural play/record. If so required, Universal can give the producer seven 4-track recorder/dubbers at the same time. These machines run at 24 fps, 25 fps, 30 fps or whatever speed the client may desire by the use of Magwarp. They are capable of looking at 50 Hz, 60 Hz, or 59.94 Hz as a sync source. In the 16mm mode they have both edge and center track. Naturally, Dolby units are available (on special request up to 96 channels). The digital arsenal has a Mitsubishi X-400 16-track digital recorder and a Sony BVH-2800 digital C format video recorder.

Figure 2. Tom Miller working with the SSL console in studio B.

Universal also offers computerized ADR using both 35mm, 16mm film and/or video. Another popular format is the synchronization of digital non-sprocket equipment with the film dubbers either for playback or record. The final master mix can be on film, audio, digital, U-matic, C-format video or hard Winchester Disc. By the way, about 50 percent of the national commercials aired on the last summer Olympics were mixed in these theaters. ADR for Top Gun, The Hoosiers, Manhunter, and Crime Story are just a few of this division's many credits.

The four film transfer rooms are capable of handling 480 different combinations from 50 Hz, 60 Hz, stereo Nagra center track to standard or drop frame time code looking at either line frequency, 60 crystal, or 59.94 house sync to 24 fps, 24 fps to 30 fps, or whatever other speed is desired using Magwarp be it 35mm 6-track, 4-track, 3-track, mono, or 16mm edge or center or both.

The optical transfer facilities are the only full-time operation of its kind in Chicago.

POST-PRODUCTION FACILITIES

Now to Studio A. This is the largest room in the middlewest. It measures 68,000 cubic feet. It can handle the Chicago Symphony Orchestra comfortably. Studio A is a 32/48-track music scoring/video post-production room. This room has been used to score the Blues Brothers film, record stars from Stevie Wonder and Frank Sinatra to The Police and Tom Waits plus Grammy-winning albums and Clio-winning spots. Phil Ramone has been quoted as saying, "If Studio A could talk, we would hear the history of modern music." Bruce Swedien claims that Studio A is one of the four best rooms in the world.

The console is a Neve 8048 with Necam 96. The digital tape machines are an Otari DTR-900, a Mitsubishi X-850, a Mitsubishi X-86 and X-80 plus a Sony DAT recorder. The analog machines are a MCI 24-track machine plus several Ampex ATR-102s and ATR-104s. Videowise there is a JVC

Figure 3. Film mixer Bill Reis at the controls.

8250 with all machines looking at a BTX Softouch system. There are two live room chambers, two EMT 140-S plates, a Lexicon 224X with Larc plus a Lexicon 480L and an AMS RMX-16 for echo/reverb. In addition there are about seventy-five additional pieces of processing equipment from Pultecs to TCs to help get any sound desired. Musical instruments include a Bösendorfer 9.5-foot Imperial grand piano with a 97 note keyboard, a Fender Rhodes, a DX-7A, a celest, a Hammond B-3 organ with Leslie, a tack piano, several drum sets to choose from and a complete set of tympanies.

Universal probably has the greatest collection of microphones anywhere. Available is almost any microphone ever made from several C-12s and U-47s through today's 414s. Of course any synthesizer required is available as well as 16mm and 35mm full-screen projection.

Studio B, like Studio A, is a 48-track music scoring/video post-production room. This studio has recorded from Nat King Cole to Prince and from Sir George Solti to Little Wally and his Polka Kings. The console is an SSL 6000 series G with Total Recall. This studio has the same digital and analog machine components as Studio A. It also has a dedicated JVC 8250/BTX Softouch package similar to Studio A. For echo there is a live room chamber, a stereo EMT plate, and another Lexicon 224X (the 480L and the RMX-16 are also available on request). The piano is a 7-foot Steinway. There is a choice of one of five Fender Rhodes or many drum kits to fill out instrument needs. Of course there are all microphones and processors as well as 16mm and 35mm full-screen projection.

The Backroom is another 48-track music scoring/video post-production room. One of the 24-track machines in this room is convertible to a 16/8-track recorder or a quad video layback machine. For C format laybacks this studio has its own C format machine with a TBC for excellent picture reproduction. There is also a dedicated JVC 8250/BTX Softouch video system. As this studio is used a great deal for foreign language recording and overdubs, the studio cue has six separate channels and a stereo mixer at every earphone position. The console is a MCI 628. The other tape machines include Ampex 102s and an Otari 4-track machine. This studio also has an AMS AudioFile with four hours of memory. It has its own 7-foot Steinway piano. For echo there are EMT plates as well as its own Lexicon 224X. All of the above-mentioned goodies from Studio A and B are also available. The Backroom has recorded from Gary Coleman to Ray Parker Junior.

The Synclavier Studio has a 32 out, 64 voices (FM and Polyphonic sampling), 200 track sequencer, 32 megabytes RAM, 320 megabytes disk storage, Direct-To-Disk, 2 in/8 out MIDI, 2 gigabyte WORM and Music Printing Synclavier System from New England Digital. The console is a Harrison. Digital tape machines include Mitsubishi X-850 and X-86. Analog machines include an Otari MTR-90 II 24-track and an Ampex 102 recorder. The same JVC 8250/BTX Softouch video package is dedicated to this studio. For echo/reverb there are Lexicon PCM 70s, Yamaha REV 7s, as well as the same peripheral equipment found in all the other studios. This studio also has a 90 square foot overdub room.

Next to our film theaters is another Synclavier studio that is used for sound effects and foley. This room, called

Figure 4. Universal's Backroom film theater.

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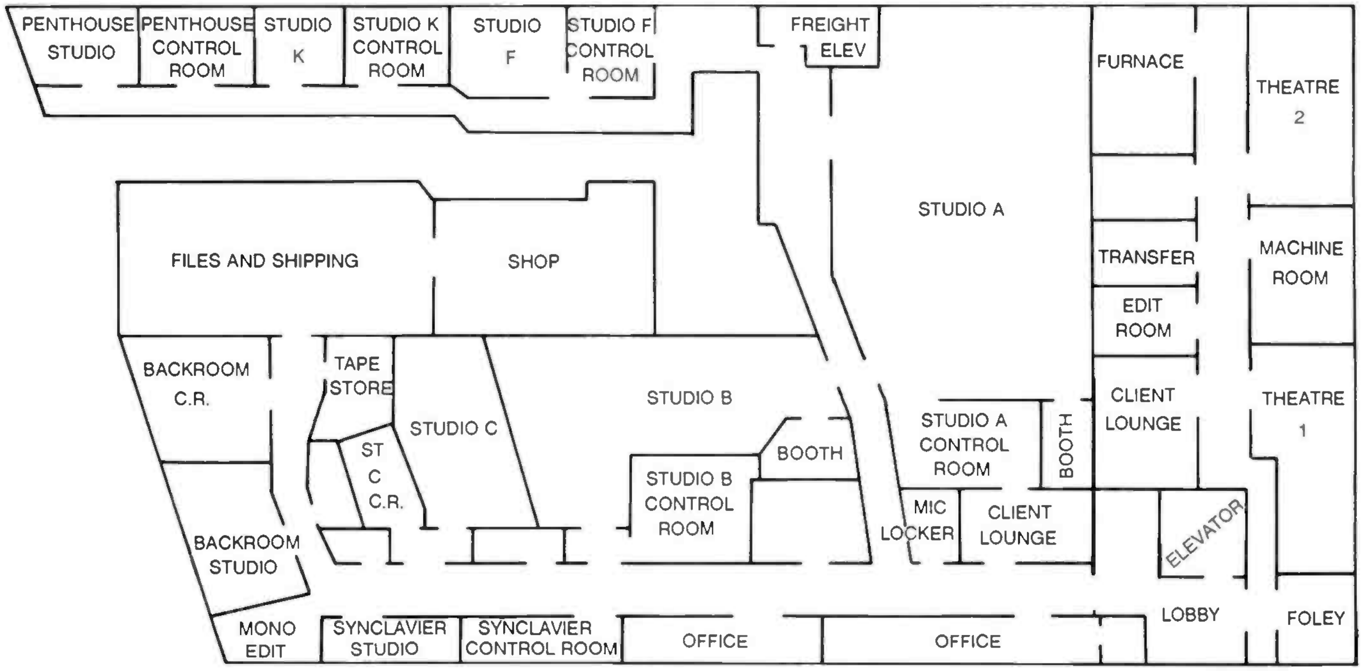


Figure 5. Overall floor plan of Universal Studios.

Sync-2, is just a scaled-down version of the above Synclavier studio with a 16-output system. This system has the new digital time compression from New England Digital. As an added feature this room also has a Sony BVH-2800 for digital audio on C-format layback. Programs created in this room can be locked in a matter of seconds to either film theater so as to avoid transfer and generation loss. Among its list of credits is foley for the Omnimax production of "The Great Barrier Reef."

THE 16-TRACKERS

Among the 16-track studios there are three in particular to discuss. Studio C is one of the "hottest" rhythm rooms in town. The console was built by the Universal staff. The piano is a 7-foot Steinway. For echo there is a live room and an EMT plate. Of course all the digital reverb units can also be used in this studio. Dozens of national commercials have been recorded and mixed in this studio. The Peabody award-winning radio dramatization of Ulysses was recorded in this room.

The next studio is the Penthouse. This room features a Ramsa 8616. This studio has a full complement of dedicated processing equipment along with VHS and U-matic video playback. Thirty-six of the Golden Books Music videos were recorded and mixed in this studio.

Another 16-track studio is Studio 8. This room is a media/video post-production studio that features a Ramsa WR-8428 console. This studio has another JVC 8250/BTX Softouch video system for sweetening. The studio is large enough to record a large string section or 20 voices. The echo is another stereo EMT plate plus a Lexicon 200 and a REV7. This studio has its own film transfer facilities as well as high-speed Magnafax machines for multiple copies.

To finish off its complement of studios there are two more 16-track rooms and an 8-track studio used for media production. Each has its own set of multi-track machines plus 2-track and mono machines. These studios also have their own peripheral package as well.

Just about every studio including these studios has its own compact disc player and most now have DAT players. For a complete list of equipment and artists who have recorded at Universal write or call. The address is Universal Recording Corporation, 46 E. Walton Pl., Chicago, IL 60611. The phone number is 312-642-6465. They will probably send you one of their award-winning brochures.

Equipment is not everything. The people who work at Universal are very special, too. Among the 41 persons employed at Universal, there will be someone that will speak just about any foreign language that is heard in the recording business. Twenty are accomplished musicians and most are college graduates. The tech department has four people with BSEE degrees and one with a masters degree. The number of gold records, Clios, Grammy, Oscar, and Emmy nominations and other awards won by people at Universal's staff ranges into the hundreds.

If you can't make it to the AES, NAB, or SMPTE conventions, drop in at Universal. It is probably the next best thing.

HARPO STUDIOS

CAROL A. LAMB

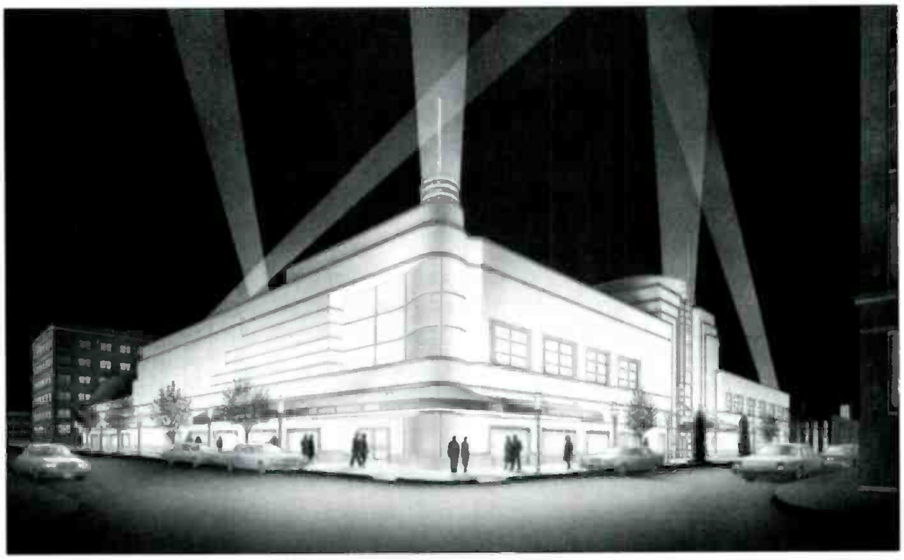
• One of the most exciting projects to hit Chicago within the past year is the purchase of a Chicago film industry landmark, the old Fred Niles studios on the Near West Side. For the past twenty years, it has been the filming location for hundreds of commercials and many feature films.

Now, Oprah Winfrey, the major investor, and her partners, syndicator King World, and manager/lawyer Jeff Jacobs, plan to invest over 10 million dollars including the reported purchase price of 4.5 million dollars.

Harpo Studios, as it is now known, is an 88,000 square foot television and film complex one mile from downtown Chicago. Oprah has commenced extensive construction and renovation to create state-of-the-art television and film stages designed to encourage more local and national production.

Harpo Studios will include, among other things, three film soundstages. Stage A is 10,000 square feet, and Stages B and C are 6,000 square feet each. These were built exclusively for film production, and are being renovated by the Chicago architectural firm of Nagle Hartray in a joint venture with interior architectural designer Pam Anderson. These soundstages will continue to be rented out for commercial and feature films, as well as to shoot Harpo's own productions. Jacobs also hired a well-known local video consultant to create a customized state-of-the-art television operation.

The complex will also include the corporate offices for Harpo Productions and Harpo, Inc., along with a screening room, a lounge, a gym, and a kitchen to cater meals. These facilities will be available to everyone: producers, actors, and crew. The studio will have a post-production editing facility to do promos and segments. It will handle 1-inch, 1/2-inch, and 1/4-inch Betacam.



A new 100x100 square foot TV studio and technical facility is being built to house *The Oprah Winfrey Show*. The TV studio is scheduled for completion in the latter part of 1989, and Oprah's first show will originate shortly thereafter from the new facility. The studio will have the capacity for a daily audience of 125, who will be able to park around the block in a Harpo-owned lot. When Oprah's set is not being used, the studio can be rented to outside producers.

Many phone calls are being received by Harpo Inc. from various editorial houses and recording studios that are all vying to be a part of the complex. The decision has yet to be made as to whether suppliers will be invited to rent space in the complex, or if Harpo Studios will initiate its own support facilities. Either way, there will be more opportunities for Chicago filmmakers to whose growth Oprah is personally dedicated, says Jacobs on her behalf. In addition, studio rentals are not to be only entertainment-oriented. Jacobs is looking to influence advertising agencies to use the facility, and to persuade them how cost-effective producing commercials at a world-class studio in Chicago would be.

Harpo Productions already has four TV and film projects in development, and Jacobs foresees the studio regularly producing network shows. The stages will be used for three ABC prime-time specials from Harpo Productions. A long-term use of the studio involves the "R&D Network" of Roger and Michael King's King World. The "R&D Network" is a consortium of stations to fund shows and play them for testing purposes on the network.

AN ADDED VIEW

Murray Allen and Universal Studios is the creative consultant for all aspects of audio production for Harpo Studios. When **db Magazine** spoke to Murray about his involvement, he said that it was one of the most exciting things that he's ever been involved with, and that it is a tremendously important project. Oprah's desire, according to Jacobs, is to open her company doors to local talent and have the studio used as a "creative center." Oprah's commitment goes further than to create a state-of-the-art studio complex; it encompasses all of Chicago.

• We would like to credit Screen, a local Chicago magazine, and editor/author Ruth L. Ratny, for much of the information on this page.

JIM DOLAN

As a multiple room facility that services the record, film, TV/radio and industrial clients, we have the opportunity in our operations to blend creativity and quality with speed and efficiency. This combination helps create answers to the challenges that reflect our diverse and constantly changing audio business.

THE GROWTH PROCESS

Streeterville, which opened in 1969 as a 6500 square feet, three-studio facility, now has nine studios covering over 27,000 square feet (with three more planned for 1989/90). We've grown through major expansions in 1979 (two studios), and 1985 (three studios, two control rooms). The ability to expand in our location for the last twenty years has allowed planning and vision to work to the optimum. The result is a facility that has good flow pattern, intelligent systems, depth and versatility. The combination of planning and technological expansion has allowed us to be ready for continued growth.

A talented engineer and a digital audio workstation create a powerful team that can truly master all the challenges of a recording session. Whether it be original music or post-sweetening, an engineer can choose the best recording and processing approach either analog or digital offer, with creative freedom of non-destructive editing and experimentation and the speed of instant access.

The challenge of selecting a suitable digital audio workstation to meet the individual mixer's needs as well as maintaining flexibility and compatibility inherent in the design of our multi-room operations was the key to our growth. The ability to engineer work privately in the studio while networking audio throughout the facility was the combination of power and versatility we saw around the corner over two and half years ago.

During our investigative stages we were looking at ways of maintaining the highest quality audio through the whole production process while handling many sound management and production tasks. Other "had to be's" were the ability to follow tight synchronization to picture via timecode, access to at least eight independent digital tracks, cut and splice reel-rocking editing, intuitive and user-friendly interface, restoring and archiving that allows for multiple operators in multiple studios.

CHOOSING A WORKSTATION

After months of extensive research and analyzation, we decided on the AMS Audiofile. The Audiofile is a stand-alone workstation with control panel and CRT display linked to the rack of processing units and hard drives. The system simultaneously controls a total of eight digital tracks, each of which can be edited independently to the video frame, using digital reel-rocking to locate exact in and out edit points. In addition to analog ins and outs, Audiofile had digital (SDIF-2 Sony 1630 format) ins and outs. AES/EBU ins and outs will be available soon. Later this year we should be able to interface EDL lists coming out of the edit sessions for quicker audio build-up. We have in operation six of the eventual nine systems; one system is configured with two-hour stereo and two-hour mono configurations.

Individual systems can be linked via timecode to expand track capacity if needed. In terms of direct user-friendliness, not all digital systems are the same, some are a great deal easier to use than others. Audiofile is the most user-friendly system we've seen. It translated to our engineers who are used to switches and buttons instead of mouse-driven or touch-sensitive approaches. When you're involved with new technology that will ultimately change the technique of the engineer, we found it is very important to be sensitive to that transition of performance from old technique to new. We went with the buttons, which proved to be more effective from a business standpoint.

Figure 7. Streeterville studio 1 has a Neve 8128 48-channel with NECAM and a 4-machine lock-up.

Training in any facility is a never-ending process, so when we placed the order for our first system, we insisted on a demo unit being made available to us prior to delivery so that the training process could begin immediately. We dedicated a three-week period of group and solo tutorials to become knowledgeable of the operational philosophy of the Audiofile. We found this allowed for much quicker integration into sessions when our first four Audiofiles arrived. With a device this powerful, and with repetitive day-to-day use, expertise became cumulative.

Every time engineers use it, they not only step further into its power, but it also sets up another frame of reference of confidence, an awareness of quality, of the best utilization of its creative potential. It enables engineers to microscopically edit out an ugly "pop" or seamlessly merge two performances

Figure 8. Studio 2 has an SSL 4000ETR and a 3-machine lock up.

together—this ability creates great excitement and positiveness in our sessions that everyone involved feels.

A MYRIAD OF ADVANTAGES

There are two primary modes of operation on the system: sound assembly, and cut and splice editing. From an engineer and producer's point of view, the Audiofile opens up enormous avenues for production answers and creativity, as well as cost savings and value for the budget dollar. On a recent job, a client brought a music track from one producer and a vocal track performed in a different key and tempo by another producer to us with the task of putting the tracks together while maintaining the integrity of the performances. Previous techniques would have offered little, if any success without severe signal degradation. With the Audiofile, engineer Bob Kruger was able to shift pitch and re-line up the vocal tracks to create a finished master that sounded great and took only forty-five minutes to complete. What began experimentally quickly turned into something very real and of the highest quality because the digital domain preserved the master quality of the elements. The track was downloaded right to one-inch video clean as a whistle.

As we go through the learning process with the system's capabilities, so do our clients. Producers will develop new and different production techniques as they become more experienced with digital workstations and what they offer. This will change the preparation, budgeting and decision process because now all elements, with multiple variations can be evaluated quicker than ever.

Our "off-line" rooms give the engineer and producer access to experimental sound assembly, editing or looping. For assembly and preview to picture with multiple versions, the ability to experiment with practically limitless, non-destructive edits, puts more creativity, intuitiveness and verve into the whole production. Now recording and assembly are closely linked and mixing/finishing (mastering in the record biz) are also linked through automation and digital powers. Soundtracks can be prepared for mixing so precisely now that the finishing to the final format is easily accomplished. This is quality audio with efficient processes that is cost-effective as well!

Figure 9. Studio 5 with its Harrison 4032 with Auto Set and a 3-machine lock up.

CENTRAL ROOM

With seven studios and two "off-line" rooms, the networking approach revolves around a central digital core room (see associated article by Steve Kusiciel, Director of Technical Operations, Streeterville). This room houses the hard-disk drives or what the staff calls the "brains." The studios and off-line rooms all have terminals or heads. Using our in-house routing system, an engineer can patch any brain (storage module) to any head (control surface) in the facility. A new release in future software will allow for true sound file networking between separate systems. The networking also allows us to upload and back-up sound files in a background mode, using a dummy head that we have in the central core room. This gives good efficiency in getting into and out of the digital domain around sessions that are back-to-back as well as directing audio throughout the facility.

The feeling that there is no session or project that could be too complex or big for us to handle is natural because the creative juices flow more readily when there is access to tools as powerful as the current generation consoles, automation and digital workstations, but let us not kid ourselves—we're headed for the 90s and the 2000—all we can do is keep trying and forging and trying and forging...

BREAKDOWN OF THE STUDIOS

Studio 1

A full-service 48-track recording studio with open drum booth, solo isolation booth and live wooded area. Private lounges and showers off control room.

Studio: 47x22x14 feet

Control room: 25x16.5x12 feet

Console: Neve 8128/48 with 48 channels of Necam 2 automation

Multi-track machine: 2 Sony/MCI JH-24s with Auto Locator 3

Video: Sony 5800

Synchronizers: 3 Machine lock-up via Timeline Lynx/BTX for 48-track

Instruments: Yamaha 7-ft. 4-in. grand piano with MIDI and Sonor drums set (oak).

Studio 2

A full-service 24-track studio with controlled f=drum area, six-person isolation (max.) booth and special wooded live area.

Studio: 37x17x12 feet

Control room: 24x17x10 feet

Console: SSL 4040E with total recall (fitted with 32 modules)

Multi-track machine: Otari MTR 90 III 24-track

Video: Sony 5800

Synchronizers: 3 Machine lock-up with Adams-Smith 2600

Instruments: 6-ft. 4-in. Steinway piano with MIDI, Sonor Signature drum set

Studio 3—The Finishing Suite

A world-class audio mixing studio that is for finishing any project whether it be for records, commercials, or films. The "Suite" also has a recording booth that can be used for vocals, percussion, and small section overdubs of any sort (no grand piano).

Studio: 14x15x12 feet

Control room: 21x22x10 feet

Console: SSL 6048E with total recall and programmable stereo EQ from SMPTE code (fitted with 48 modules)

Multi-track machine: 2 Otari MTR-90s 24-track

Video: Sony BVU 800

Synchronizers: 5 Machine lock-up with Adams-Smith 2600

Studios 1, 2, and the Suite all have Lexicon 480L and 224XL digital reverb with Larcs.

Studio 5

A 24-track narration and sfx post-production studio that also services synthesizer sessions.

Console: Harrison 4032 with Auto-Set 1

Multi-track machine: Sony/MCI JH-24 24-track

Video: Sony 5800

Synchronizers: Timeline Lynx 3 Machine lock-up

Studio 6

A 24-track narration and sfx post-production studio that also services synthesizer sessions.

Console: Harrison 4032 with Auto-Set 1

Multi-track machine: Sony/MCI JH-24 24-track

Video: Sony 5800

Synchronizers: Timeline Lynx 3 Machine lock-up

Studio 7

A 4-track narration and sfx post-production studio with a noted high-quality signal path for voice recording.

Studio: 2-3 persons

Console: Custom Neve

Multi-track machine: Studer 80 MK II

Video: Sony 5000

Synchronizers: Q-Lock 3.10 3 Machine lock-up

Studios 5, 6, and 7 all have Lexicon LXP-1s and can supplement with Lexicon PCM 60s and Yamaha REV 7 and SPX 90s.

Figure 10. An 18-year-old Neve discrete console. It has been renovated with an extended patch bay and 4-track monitoring.

SIDEBAR

STEVE KUSICIEL

• When Streeterville first began to consider the installation of digital audio editing systems, we realized that in a multiple room facility, we would need the ability to operate with the minimum amount of disturbance to the clients. This need meant that some sort of a central location for the main computer bins was necessary. We chose a location based on the distance from the computer core to each control room. This ended up to be on the third floor in the center of the building.

We then ran three sets of data/control cables to each control room through our existing central patch cable troughs—one cable to provide the control surface to computer link, one to handle remote machine control, and one for a spare. These cables conform to the RS-422 standard so length and shielding requirements were not as critical as they could have been. Next was audio I/O as well as timecode, tach, and direction signals needed to implement tape machine control. We did this by running multi-pair balanced audio cable from the Audiofile computer bins to our existing audio central patch facility so that we could access virtually any of our control rooms by just changing patches. The timecode ins and outs and tach/dir signals also went to central patch. Back in the core, a patching system was created to enable connecting any control surface in any room to the appropriate computer bin needed for the session.

ARCHIVE AND RESTORE PROCESS

When this was completed, we went on to installation of the Archive and Restore system in the core room. The Audiofile currently uses a modified PCM701 system to archive the Cue Library data and sound information to video tape storage. The archive and restore process is in real time so it was necessary to do this more or less off-line to avoid delays when we were changing from session to session during the business day. We have two PCM processors and a dedicated control surface in the core room just to handle the archive and restore tasks.

In view of the large number of archive and restore tasks we have on a daily basis, the scheduling of Audiofile usage has to be carefully done. One really needs to know how long it will take to archive or restore a given tape for that session. Streeterville has developed a staff interface concept designed to make these tasks invisible to our clients. We maintain a separate schedule for archive and restore and units are available according to that time table.

Quite a lot of our sessions require assembling audio elements to picture so we decided to provide video tape machine remote control via Audiofile transport keys on the control surface. These commands are transmitted on the RS-422 port according to the Sony protocol so machines like the BVU-800 JVC-850 and BVH-2000 plug right in. Other machines such as the VO-5800 require an interface converter to work properly. The Streeterville R&D department then designed and built a single board computer to interface the machine control port on the Audiofile to an audio tape machine so that we could take advantage of the expanded track capability of locking an Audiofile to a 24-track audio machine, or for that matter, any audio tape with time code on a channel. This has proved to be very useful when working without picture elements or when more than eight simultaneous outputs are needed.

With this capability, it is possible to mix a master music element directly into Audiofile in a music session. A record announcer can take into the same Audiofile in a production session, add sound efx, and mix directly to a client's 1-inch video master in the layback room without an analog tape in the chain.

We also interfaced a Sony PCM-2500 RDAT deck to the Audiofile via the SDIF-2 port (1630 format) so that we could provide direct digital transfer to and from RDAT tapes for clients that are now using this format in their offices. It should be noted that this is only possible at 44.1 sample rate because the Audiofile is at 44.1. In the near future, the Audiofile system will be able to archive and restore to RDAT tapes as well as PCM-701 and an AES/EBU port will also be implemented.

SOFTWARE UPGRADE

In the area of software updates, we are currently involved with the system's upgrade to vers. 8.02. These changes involve new software installation as well as hardware modes. The enhancements include a greatly expanded Cut and Splice facility, timecode output from Audiofile that follows the real-time position of the events list, and enhanced editing capability among other things. AMS has also developed an ADR function which includes multiple take features and beep outputs.

In the future, we are looking forward to the installation of Audiofile networking hardware and software which will operate similarly to a local area network for PCs. This should allow us to interconnect the Audiofile systems to a large memory system which will act as a master file server and a temporary storage for our jobs in progress. This will also permit linking multiple computer bins for more audio channel capability as well as eight or more simultaneous audio record inputs. We are also looking forward to the release of AMS LOGIC1 which is a totally integrated digital mixing system for the Audiofile.

TROUBLE-FREE

As far as maintenance of Audiofile goes, the systems have been almost trouble-free in the hardware sense. The most notable would be a crashed Winchester disk drive which was replaced the next day from the AMS office in Seattle via Federal Express. The technical support from Seattle as well as the local area representative is as professional as can be found in the business. In the software sense, because of the number of systems we have on line, software problems become a routine topic of discussion both in-house as well as with the people at AMS. Part of the key to success at Streeterville is the fact that we can provide reliable service to our clients in this era of rapidly changing technical capability. As with any business or any technology, it's really the people that make it all work, after all, needs should dictate technology. Technology should not dictate needs.

WFMT—A TECHNOLOGICAL LEADER

GORDON S. CARTER

WFMT, the dream child of Bernard and Rita Jacobs, was founded in 1951. Their goal was to share the music they enjoyed with their listeners. Although WFMT is a commercial operation, advertisers were sometimes skeptical of a classical radio station broadcasting in the new medium of FM. Money was sometimes tight, and it often became necessary to "make do." High fidelity was becoming popular with serious music listeners, and the owners of WFMT wanted to make their station sound as true to life as possible.

In the effort to provide the best possible fidelity to its listeners, WFMT has always been an innovator and a pioneer. In 1961, before the approval of the current FM stereo broadcasting system, WFMT cooperated with Chicago's public television station, WTTW/Channel 11 in stereophonic broadcasting. The left channel was carried on the television station while the right channel was on WFMT. This prepared the listeners for stereo broadcasting which began on WFMT in 1962, when history was made by broadcasting the first live stereo concert series anywhere.

CONSOLE MODIFICATION

At the time broadcast consoles capable of stereo were available, but were not as good as the high-fidelity equipment available to consumers. To solve this problem, WFMT's engineers modified a monophonic RCA broadcast console for stereo. When finished, about the only resemblance it had to the original was the cabinet and front panel controls. The electronics had been completely rebuilt. High-fidelity turntables and phono cartridges were used rather than so-called broadcast turntables to get better performance. WFMT was using the same type of equipment on the air that many of their listeners had at home!

Magnetic tape has always been a major source of programming for WFMT, so to get better performance from their tape machines, WFMT obtained a Dolby A301 in 1969. This unit was the predecessor to the ubiquitous Dolby A361, found in most recording studios today.

In April, 1969, WFMT moved to new studios on Chicago's Michigan Avenue. Due to the age of most of the equipment then owned by WFMT, virtually all equipment for the new studios was to be purchased new. A search began for a stereo broadcast console that was truly high-fidelity and would meet our needs. The only consoles that met the desired audio specifications were recording studio consoles. They provided many features, but did not have some that were deemed essential. In addition, the cost of such these consoles was prohibitively high. As a result, we decided that we would build our own consoles. These consoles were designed and built by WFMT's Chief Engineer, Al Antlitz. These consoles had audio performance that was unexcelled in both the broadcast and recording world.

The 1970s were a time of discovery and invention in the audio world. Pioneers in the industry were discovering more and more factors that affected audio quality. The Mincom Division of 3M Company had produced a professional tape recorder using the patented "Iso-Loop" transport that addressed many of the problems of tape recording. These machines greatly reduced scrape flutter and had many other innovations such as dual-pitch screws for the azimuth adjustment to provide greater stability and fully modular electronics using the latest technology. Two of these machines were purchased in 1/4-inch versions for WFMT's main broadcast studio.

In the early 1970s the hot topic in the audio world was no longer stereo, but quadriphonic sound. A number of matrix systems were developed to provide quadriphonic sound on only two channels. WFMT experimented with this new technology, and in September, 1971, broadcast what is believed to be the first live quadriphonic (matrix) full-length opera production, Semiramide, from Chicago's Lyric Opera.

By the mid 1970s our changing needs necessitated some revisions in the studio equipment. The monitor and intercom facilities in the studios were found to be inadequate, so we designed a new monitor and intercom system.

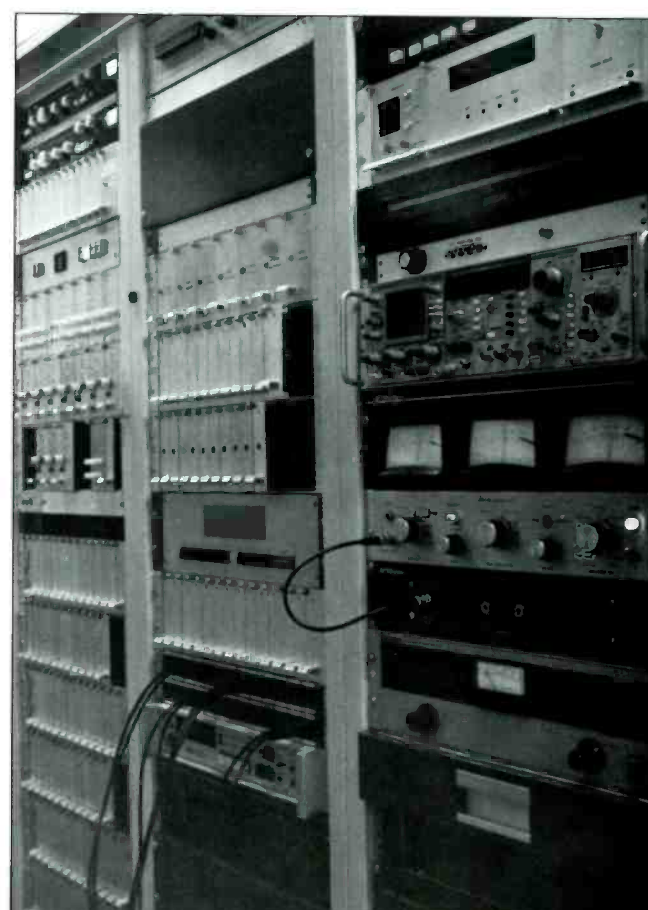


Figure 11. WFMT's central equipment room. The modular card cages are all WFMT-built equipment.

While this was a relatively small project, and did not make any changes in the station visible to the public, many new techniques such as CMOS logic control, audio integrated circuits, and cmos analog switches were tried at this time. This laid the groundwork for future projects.

PROCESSING PROBLEMS

Classical music poses some unique audio processing problems. Although FM has a limited dynamic range, even under the best of conditions, classical music has a wide dynamic range which is an integral part of the enjoyment of the music. Processing equipment that will deal with the full range of classical music without any undesirable artifacts and still deal with the 75 micro-second pre-emphasis requirement for FM is very difficult to find. After much research, we felt that we could do a better job than anything that was currently being built. Al Antlitz set out to design an audio processor that provided low distortion, accurate control of the audio, and was psycho-acoustically pleasing. The first incarnation of this device, called "The Antlitzer" by WFMT staff, was put on the air in the early 1970s. However, development of the device continued and the second generation, called the "Moduplex I" was put on the air in 1975. Research and development on this problem continues to this day.

When it became time to replace the aging exciter and stereo generator (the heart of the transmitter), we investigated all the equipment available at the time. After visiting the two leading manufacturers for demonstrations of their latest and best equipment, we found an exciter/stereo generator package we liked, but were not satisfied with the stated specifications. The demonstration showed that the equipment was capable of much better performance than was shown on the specification sheets. Negotiations with the manufacturer led WFMT to purchase their exciter, but with the demonstrated performance being guaranteed. This purchase allowed WFMT to broadcast what was probably the best signal at the time. Such performance is now routinely achieved by most leading manufacturers of such equipment.

As our operation expanded the necessity for more tape recorders became evident. The Mincom recorders were no longer available, so the search began for new machines. Two machines from Philips were tried, but they had some problems with acoustical noise in the control room that restricted their usefulness. By the next time recorders had to be added to the operation, Studer had opened their own facilities and distribution in the United States. A machine was tried for a month, and now, over 13 years later, is still performing as well as it did the day it was new. Since then, Studer recorders have become the standard for WFMT. Even though the initial cost is higher than other makes of recorders, the total cost of ownership has proven to be much less.

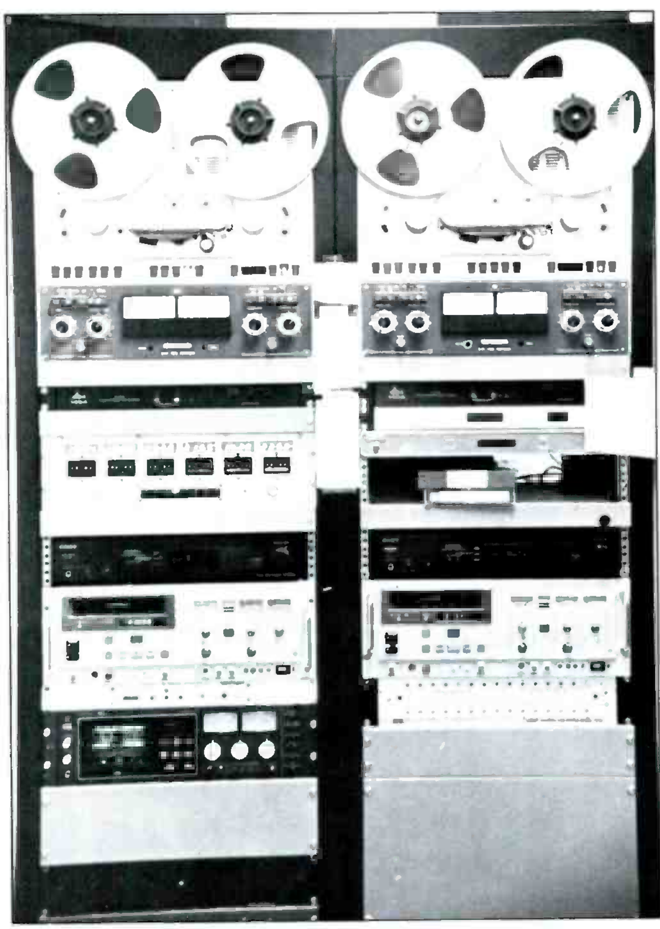


Figure 12. The uplink automation facility for Concert Music Consortium and Beethoven Satellite Network.

During the summer of 1978 WFMT's management determined that a tape duplication facility was needed for the growing Chicago Symphony Orchestra syndication, with more than 300 stations taking the weekly two-hour concerts. High-speed duplicators could not provide consistent stability while dealing with 10-1/2 inch reels without a great deal of time being devoted to quality control. WFMT's engineering department proposed that the tape duplication be done in real time. In order to provide quality control, a scanning monitor system was devised that allowed the operator to monitor the output of all tape copies as they are being made. Since the scanning system allows comparative monitoring of each output, anything unusual is easily noticed. This relieves the operator of the need to discern the absolute quality of the tapes. This system gained WFMT the reputation of having the best quality tapes of any classical music syndicator.

In 1978 WFMT participated in the first stereo-sound-only relay of a live performance by domestic satellite. Then, in June of 1979, we became America's first radio "superstation," being delivered by satellite to cable companies throughout the United States.

A MAJOR UPDATING

By 1980 we had outgrown our facilities on Michigan Avenue and began looking for new quarters. We moved to Chicago's Illinois Center in the fall of 1981, where we were finally able to realize a dream—a music-performance studio. This move also allowed for the first major updating of studio equipment since our last move in 1969. As available equipment was investigated, we decided that we should once again design and build our own studio equipment.

A total system plan was devised so that all equipment would be as interchangeable as possible. Modular microphone pre-amps were designed using Jensen transformers and 990 op-amps. The consoles were designed using VCA technology and CMOS switching to that no audio passes through any faders or mechanical contacts. Many other features not available on other consoles were built into the WFMT consoles, such as soft turn-on and turn-off of channels.

Since WFMT broadcasts more live remotes than most radio stations, we rely heavily on telephone lines. The proper interface for equalized lines was designed and incorporated into modular amplifiers. These amplifiers are used throughout the facility for interface and distribution applications as well.

All control and production rooms are fully self-contained, but may need to interface to remotes or other rooms. To accomplish this with a minimum of crosstalk, all lines to and from each room were run to a central equipment room. In this room is all equipment that is to be shared by the rooms, as well as the termination point for all telephone lines. Since patch bays can be confusing and unreliable if not used frequently, all signal routing is done by a routing switcher designed by WFMT's engineering staff. Switching is placed where it is most logical to reduce wiring and crosstalk, and the control is located at its most logical point. CMOS analog switches are used for all signal routing, and the status of the system can be read at the control point in English on an LCD display. Total system performance at WFMT is better than many radio stations are able to achieve with only a console.

Each room in the studio/control room complex is provided with a monitor and intercom system. As is the case with the routing switcher, the audio is switched in the central equipment room, along with the volume control and muting functions. This reduces the amount of audio wiring required as well as the crosstalk. The intercom system allows each room to talk to any other by pressing a single button. This system is interfaced to the muting and warning light system so that it is impossible to call a room that has an open mic. Also, signalling is provided so that the person being called knows what room is calling, without the caller having to identify his location.

Figure 13. The tape duplication facility.

All turntables in the facility are on vibration isolators to prevent footfalls from being heard on the air. The turntables are Technics SP-15s with Signet tone arms and Shure V15 Type V cartridges. These phono cartridges were selected to provide the best reproduction of records while still allowing for ease of maintenance.

SEARCH FOR BETTER WAYS

During the Summer Consumer Electronics Show in Chicago in June, 1982, Sony provided WFMT with a prototype compact disc player and some discs. We were the first radio station in the world to broadcast a compact disc. Within two years CDs were an integral part of WFMT's programming.

The search for better ways to record concerts and recitals never ends at WFMT. For years remotes were done using two Nagra IV-S portable reel-to-reel recorders. These were small and provided good fidelity, but were inconvenient to use. We did an experimental broadcast in the early days of digital using the Stockham/Soundstream system. While the sound was impressive, the equipment was bulky and expensive. When Sony came out with the PCM-F1 digital converter and the

Figure 14. Some of the many Studer tape recorders at WFMT.

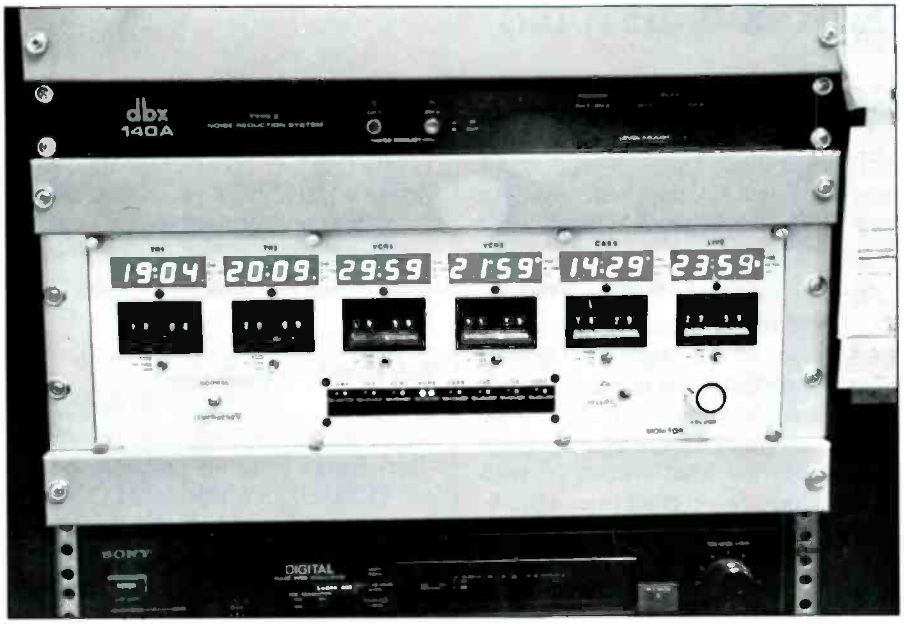


Figure 15. A close-up of the automation controller.

SL2000 portable VCR, it seemed that the solution for remote recordings had been found. The equipment was light, portable, and easy to use. Recording time was longer than reel-to-reel tape and the fidelity was better. Editing was a problem, but this was solved by transferring to Dolby A processed tape in the studio. In essence, a generation of tape was saved, since masters are not edited. The technique has become popular at WFMT, and now there are almost as many VCRs at WFMT as there are analog recorders.

A DIGITAL BROADCAST

The PCM-F1 converter and its successors have enabled us to do things other than simply make recordings. In 1983 WFMT broadcast a complete performance of Gotterdammerung live from the Bayreuth Music Festival in Germany. The audio was digitized in Europe and sent to the United States by satellite where it was converted back to audio and was distributed by National Public Radio to stations throughout the United States. The audio quality for this broadcast was better than most domestic broadcasts. Shortly after this, we began using the PCM technology to relay our main audio signal by video microwave from our studios to the transmitter in Chicago's John Hancock Center. This technique provides better signal-to-noise than is available by conventional microwave STLs or telephone lines.

In 1986 WFMT launched the Beethoven Satellite Network, a live format service available to radio stations by satellite. With studios located in downtown Chicago, the problem was to get the audio to the uplink, more than ten miles away without severe degradation. Once again PCM technology was utilized over a microwave link to relay the signal to the uplink.

EVERYTHING FROM CHICAGO

Within a few months after the start of the Beethoven Satellite Network, the Concert Music Consortium, of which the WFMT Fine Arts Network is a major participant, decided to do all their uplinking from Chicago. Concert Music Consortium distributes classical music programming to subscriber stations throughout the United States via satellite. The source material is a variety of taped concerts and programs, both on analog as well as PCM on video tape. A facility was set up to perform this function, but economics as well as practicality demanded an automated system. After some investigation, we found that there were no commercially available automation systems that could do the job as desired. Again, WFMT's engineering department was called on to design a system to do the job. The automation system that was built relies on human operators to load the tapes and set the automation system, when then starts programs automatically and on time.

TELCO USE

As was mentioned earlier, we rely heavily on leased telephone lines for remote broadcasts. Due to escalating costs over the last few years, we have developed a method of using low-cost circuits from the telephone company for the broadcasts. Special amplifiers were designed to provide proper sources and loads for the lines and do the necessary equalization. To get adequate noise performance, we use Dolby SR on the lines. This technique provides better quality than the telephone company could provide for a fraction of the cost.

TO THE FUTURE

WFMT was the first station to broadcast DAT in the United States, again with the assistance of Sony. WFMT is now using DAT for some remote recordings, dubbing to Dolby SR analog tape for editing, much as with PCM tapes. PCM on video tape is still used, and will probably still be around as long as equipment and repair parts are still available. It is unclear if DAT will become a large part of our operation or not. We have also been evaluating some of the new digital editing and storage systems, but have been unable to purchase anything to this point.

TO THE FUTURE

What the future holds is uncertain. Digital audio seems to be the wave of the future, but the form (DAT, recordable CD, or some other format) is still uncertain. However, regardless of the means to the end, the goal is clear—to provide the best possible quality within the technical and economic restrictions of the medium.

WFMT has an advantage with good equipment, but consistently high performance standards is a direct product of the people involved. Proper maintenance of the equipment is as essential as having the right equipment. While perfection will never be obtained, it is still the ultimate goal. The pursuit of that goal, not the equipment, is what sets WFMT apart.

Gordon S. Carter is the Studio Supervisor at WFMT in Chicago.

JERRY SOTO

Soto Sound Studio is housed in a 2-story brown building at 1215 W. Belmont in Chicago. To the right of the building is "The Theater Building" which rents out three different theaters to various Chicago theater companies. To the left is Murphy's "Gourmet" Hot Dog Stand. About five-hundred feet away is a 7-11 store open 24 hours—very convenient. Soto Sound is also three blocks away from one of the hottest intersections in Chicago—Belmont and Sheffield. The area clubs feature some of Chicago's best talent, ranging from blues to jazz, pop to R&B, rock to weirdness. Soto has been able to attract some of those acts as new clients since the studio moved from Evanston to Chicago on December 26, 1987.

The studio looked a bit rough at first—no control room window, and cables ran under the door. I bought some canvas dropcloths from a local paint store because they were extremely easy to put up. They covered the walls and ceiling well and got rid of the echo at the same time without turning the room into a vacuum. Some guys might say that it takes months and sometimes years to get a decent-sounding room. That might be true in some cases, but I haven't had any complaints, and I'm still improving and adding on as the days go by.

Ironically, the clients really like the dropcloth technique; it makes the room cozy and comfortable—sort of like being in a tent! Because cost was kept low, I was able to keep my overhead down and open for business soon after I moved in.

A LITTLE BACKGROUND

In 1968 I moved to Evanston, Illinois and spent the next nine years involved in various R&B groups playing James Brown, Isley Bros., Funkadelic, Stylistics, Temptations, etc. In the early 1970s, Experience II was a great band. We opened for Curtis Mayfield, The Spinners, Alice Coltrane, and also got signed with Capitol Records. I also played with Eddie Clearwater's blues band for two years which got me a lot of my blues clients. I've probably recorded everybody that isn't signed with Alligator Records—Buddy Guy, Eddie Shaw, Phil Guy, Robert Covington, A.C. Reed, Lefty Dizz, Hip Link Chain and many others. I also co-produced a rap record with Chicago Bears running back Walter Payton and William "Refrigerator" Perry.

I got my first 4-track (Teac 2340) in 1975. At that time the lead singer in our group had some originals that he wanted the band to learn. He and I went to my house and I arranged the music around his lyrics. I have a pretty good ear for hearing what chords go with what melody. So on track one of the 4-track I recorded the voice, track two was the guitar, track three was the Fender Rhodes, and track four was my Univox drum machine, the kind that wasn't programmable (that was unheard of). We mixed it on a cassette through a "Panamix," a mixer that I built from "Popular Electronics." It was 6x2-volume and pan, and that was it.

We took the mixed tape back to the band and they had no excuse for not learning the songs. Soon after, we were in a real studio—Gary Loizo's Pumpkin Studio. It was in his garage at the time.

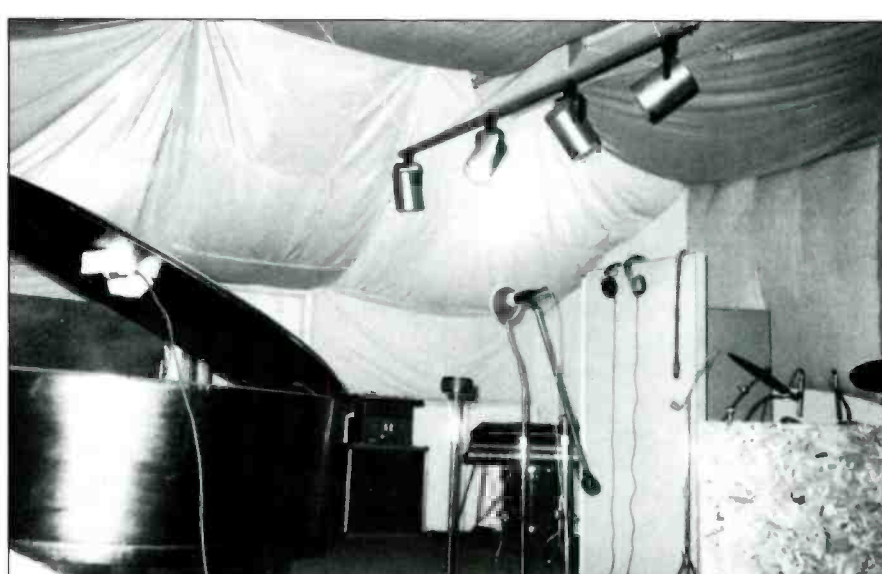
On April 15, 1977, Soto Sound Studio opened its doors at 931 Sherman Ave. in Evanston, Illinois. I advertised in the classifieds of the Reader (a local liberal paper)—seven tracks at $15.00 an hour. I managed this by recording the guitar, keyboard, bass and drums on four separate tracks, mixing them down to another reel-to-reel, and taking that reel off and putting it back on the 4-track. Then I would record the lead vocal, background vocals, and lead guitar on three other tracks. On the final mix, I was able to pan the background vocals and lead guitar to create some kind of stereo effect. A year and a half later I got eight tracks.

In 1984 I needed 16 tracks to keep up with the competition. I asked my friend Terry Burke (now my partner) to lend me $5500.00 for a Tascam 90-16 1-inch 16-track deck that a studio downtown was selling. It's an older machine that takes a licking and keeps on ticking.

NARAS AND EARS

I joined the National Academy of Recording Arts and Sciences (NARAS—the "Grammy" people) because I have a firm commitment in raising Chicago's profile in the recording industry on a national level. In 1982 I had a friend who was already on the membership committee. He stepped down and put me in his place. When I joined, NARAS was not very active, so I asked why not and pointed out that if there was any organization that could do something, NARAS was it. At that time I was regarded to some as an "upstart," but things have gotten a lot better. I'm not saying that I was the only one with this attitude, by any means, but I can say that I did make a contribution. I am now on the Board of Gover-

Figure 16. Soto Studio. It's a studio, not a tent.



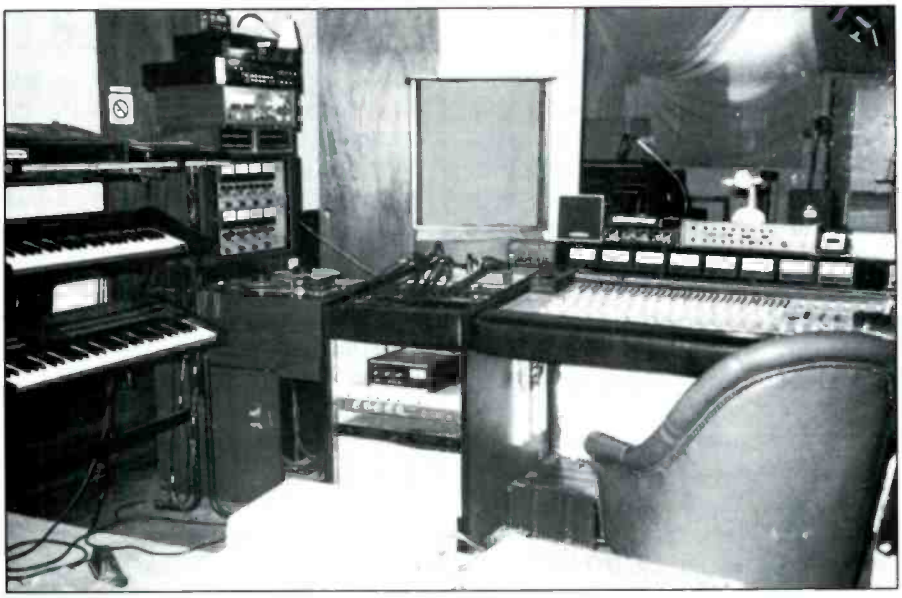


Figure 17. Soto Sound's control room is quite traditional.

nors and Co-Chairman of the membership and studio outreach committees.

In the past three years, the Chicago chapter of NARAS has grown in leaps and bounds, and has given new life to the Chicago music/recording scene with seminars, outreach committees and very active networking. One committee that has become very successful is the Engineering and Recording society or E.A.R.S. EARS was formed when Mike Rasfeld of Acme Studio, Marty Feldman of Paragon, and Mike Freeman of Hedden West bumped into each other at a New York AES convention. They noticed that the only time they really spoke with each other was when they were out of town at a convention and how great it would be to form a group to help each other out with any problems or ideas in the business.

Today EARS meets at the legendary Keenan O'Malley's, a bar on Chicago's north side, on the last Tuesday of every month to discuss issues facing the Chicago Recording Industry and to foster a cooperative atmosphere to benefit all. Since its conception, EARS has had tape, mic, and reverb shootouts and visits from many major tape, console, outboard gear, mastering and musical instrument suppliers. They think it's great because now they can give their presentations at one place and they know where and when to find us.

EARS also holds an annual roast where we pay homage and rib the colleague who gets the most votes. Last year it was Marty from Paragon. This year it was for the immortal father of the Chicago Recording Industry, Mr. Malcolm Chisholm. Among the roasters were Marty Feldman of Paragon, Murray Allen of Universal Studios, Gene Barge, the noted saxophonist and arranger. There were also phone messages from veteran studio musician Phil Upchurch, and Bruce Swedien. Engineer Randy Kling came up from Nashville. Hosting the roast was Nate Herman, formerly of Second City and writer of four years with Saturday Night Live. Nate did a brilliant job, especially when it came to the slides. It was very much like the weekend update with Dennis Miller. Proceeds go to the center on deafness in Des Plaines, Illinois.

We've also held a round table forum about the "tricks of the trade," a discussion of easy-to-implement little known secrets that brought more "bang for the buck" and we voted for the best one. There were some pretty good entries, but for some reason everyone thought mine was the best. I told them to take a 1-inch metal hub and the metal insert from a tape box and glue the insert on top of the hub, and presto—the perfect studio ashtray! To get really fancy, silkscreen the logo of the studio on the side. My prize was a $10.00 Vom meter.

EARS has helped me tremendously. Just from networking with others in my field, I have gotten great deals on tape and equipment. I was able to get a great deal from Seagrape Recording on a Neotek Series I console. Just by keeping in touch with peers can benefit a career.

I'm also on the public relations committee of EARS. One of my formulas for achieving success is "The more people you know, the less money you have to spend." I'm very glad EARS and NARAS are working well and at this rate I see Chicago well on its way to becoming one of the country's music capitals once again.

Figure 18. The author at the console.

MUSIC TO LYRICS

The unique thing about Soto Sound Studio is that I arrange music to lyrics for people who can't find musicians, for $175.00 per song. This is not where the client sends their song through the mail. They are there for the duration of the session to let me know exactly how they want their song to sound, and their voice is recorded on their song. All the client has to do is sing their song on a cassette at home, while snapping their fingers or tapping their feet so that I can hear the tempo and time signature. They bring the cassette to the studio for a 1-hour pre-production meeting at no charge. It is then that I hear what they have, and they can see the studio and hear what I've done for other songwriters. A recording of the style of music by their favorite artist is also helpful to zero in on the style they want. We then set the recording date, usually a week later. The client doesn't necessarily have to leave the cassette with me, because I do all of the work on the day of the session in a period of five hours. The parts that I play are always in my mind ready to go onto tape.

A hypothetical situation would be: I program the drum machine with a simple beat, with snare, kick and hi-hat. After checking the song structure, we then record a scratch vocal, drums and keyboard. I put in the bass, either on guitar or keyboard, then lead and rhythm guitars. After that I add strings or synthesizer depending on the song. Finally we put on the lead vocal and mix. This is done on the 8-track machine and the price is based on five hours 8-track time at $30.00 per hour plus tape, an 8-track multi-track master, a reel-to-reel and a cassette. By the way, all rights belong to the client and I don't charge extra for playing instruments myself or producing. The benefit here is two-fold. It keeps their overhead down and my chops up. Asking for a percentage in addition to my fee is only a deterrent to the client. As far as I see it, when the client has a hit, hopefully the next project will be done with me. Then we'll strike up a collaboration deal. One client, Michael Griffen, has been signed with AKA Records, a local dance music label.

I've been doing this for about thirteen years, eleven years with the studio, and it has kept my head above water since noone else in the Chicagoland area offers this service. I also get a lot of referrals from other studios.

My equipment is very modest. On one occasion a studio owner that I've known for years, from a studio with over forty gold and platinum albums, came to the new Belmont location after an EARS meeting and was very impressed. He went crazy over the video camera I have hanging over my 16-track counter. I don't have a remote for the Tascam 90-16, so I came up with the camera idea. I also rewired a remote that was really meant for a Nakamichi cassette deck.

OVERSEAS ARRANGEMENT

Soto Sound has a unique arrangement with JSP Records in London, England. Specializing in blues recordings, JSP and I have had the privilege in recording such blues notables as Buddy Guy and his brother Phil, Brewer Phillips, Louisiana Red, Jimmy "Fast Fingers" Dawkins, Carey Bell and many others. The arrangement we have is that JSP contacts the artist here in Chicago and then contacts me at the studio. I then set up a date with the artist and do the session. Blues doesn't take that long; it's very spontaneous with a lot of first takes. It takes about 6-8 hours to record an album. After the tracks are laid, I then do a rough mix at 15 in./sec. and make some cassette copies for the band. The next day, 1-inch and 1/4-inch masters are sent to JSP air mail. They receive the tapes within five to seven days. The album is then remixed in England. Sometimes they use the rough mix such as on the Phil Guy album "Bad Luck Boy," JSP 1061.

Being a musician in the last session with Carey Bell, I was able to sit in and overdub organ on four tunes. Gladys Cooks, Carey's manager, noticed all of the keyboards around the studio and asked if I could play on some songs, so I did. I felt something like Billy Preston sitting in with the Beatles. John Stedman at JSP liked it so much that he said he would leave it in. I do this with my regular clients and hope to do more of this on future projects with JSP.

Another benefit of recording with an overseas label is that people in France have seen the studio's name on the albums and every year a group from Versailles in France come to the Chicago Blues Festival. While they are in town, they make it a point to stop by the studio either to watch a blues session or record a few tunes themselves. It's flattering, and I really enjoy it.

As far as goals are concerned, we plan to get a new piano around the first of the coming year and eventually a new 16-track deck. Another goal is to buy 1215 W. Belmont within two years and rehab the interior.

SIDEBAR: E.A.R.S.

TOM HABAN

If a group of A+R men from America were polled and asked the question, "Which local music scene has the most impact internationally?", the answer would be unanimous! Chicago, with its vibrant dance music scene which created first House Music and then Acid House, is the leader in style in both England and the European mainland. The spirit of cooperation that exists here between studio owners, engineers and clients has played a large part in that success. The formation of E.A.R.S. (Engineering and Recording Society) is both a cause and an effect of that cooperation.

Formed three years ago as a way to address issues facing the studio industry, this group has evolved into a combination of tech seminar and frat party. When formerly competitive studio owners become drinking buddies it's, "Katie, bar the door and deadbeat clients beware."

Among the topics covered by the group are: microphone shoot-outs, reverb units under $400.00, how to raise your rates, proper test tones, and a host of other subjects. There have also been numerous manufacturer demos including: AMS Audiofile, Neve, Studer, Focusrite, Synclavier, Ampex and Scotch tape, and many others. Manufacturer reps say that they can accomplish more in one visit to an E.A.R.S. meeting than they can in a week of cold calls.

Cooperation between studios has also been overwhelming. Used equipment sales are facilitated, blank tape is lent freely, and information is exchanged on client's preferences and idiosyncrasies.

Cooperation between all parties to a record project improves the situation for all concerned and in fact, it's hard to imagine what it was like before E.A.R.S. Anyone interested in starting a chapter of E.A.R.S. in their city should call Tom Haban at Seagrape Recording, (312) 784-0773. E.A.R.S. is an outreach committee of the Chicago Chapter of NARAS and we appreciate the strong support and encouragement they have given to us.

SHURE BROTHERS, INC.

LARRY ZIDE

One morning recently, we sat in our office awaiting a telephone call that would come from the Shure Brothers Inc. plant in Evanston, Illinois. We had set up the telephone conference call to find out just where this Chicago-suburb based company is and where it is going. The Shure group, led by James Kogen, President, was seated in a conference room with their automatic microphone system already in place.

James Kogen: I want to begin by telling you a little about what Shure Brothers does in general, and then we'll let you start asking the questions.

In general, our business is making electroacoustical products, and doing so in depth in terms of technical development and advance manufacturing capability. We see that as our primary effort. It's not a secondary effort or a side issue, as it is with so many of our competitors. The different areas that we get into that are pertinent to this morning's discussion are represented by a number of people here. We make microphones for broadcast, live performance, home recording and a variety of other things. Alan Shirley and Sandy Schroeder are here to discuss that subject. We make circuitry for broadcast, live performance, and public address, among other things. We also have teleconferencing equipment, and Michael Pettersen is here to discuss that subject.

db: I assume we are talking on that teleconferencing equipment right now.

JK: That's right. We are using our ST-3000 in my conference room. We make home theater sound equipment which is used for home video viewing. Bob Schulein is here to discuss that. The reason we've asked Bob in—normally that is thought of as being a high-fidelity product—is that we've started working on an encoder for recording applications and I think Bob can address that.

db: One of the things that we wanted to address briefly is the history of Shure. While we think Shure has been making microphones almost from day one, it has and probably still is very active in the hi-fi end of things and was known for many years as a phono-cartridge manufacturer, and for all we know, still is...

JK: Definitely.

db: We don't know that we want to totally ignore that aspect of the business, but obviously in a magazine such as **db**, it's not our main thrust. Who would like to talk a little about the history of the company and where it actually came from.

JK: Mr. Shure started the company in 1925, when he was a very young man out of the University of Chicago. He's still active in the business, comes to work on a regular basis, and is deeply involved. The company went into the microphone business in about 1931, and prospered and grew during the Depression, became a major vendor of microphones to the armed forces during WWII, and of course has continued to grow and add to its line since then. We went into the phono-cartridge business in 1936, and were an OEM supplier until the 1950s. During the mid-50s we started into the high-fidelity phono-cartridge business, and of course that grew significantly when stereo came in and high-fidelity became popular.

We still manufacture a great many phono-cartridges, and we expect to continue in that business as long as there's a business around. We make a lot of phono-cartridges for broadcast and other professional applications and of course for the hi-fi market. Our microphone area—we've always approached that strictly from an engineering standpoint. That is, we have very strong technical support. Our first engineer in microphones was Ben Bauer, who later became one of the most famous engineers in audio. Ben did a lot of development including the invention of the first uni-directional single element microphone. Following that, we've always had a very strong engineering core for the company, and a manufacturing facility that's been built up to support that. Our main offices are in Evanston, Illinois, and we have factories in Wheeling, Illinois, Aguaprieta and Juarez, Mexico, and supporting facilities in El Paso, Texas and Douglas, Arizona.

db: That covers that part of the question. One of the things that we are curious about is why the company is known as Shure Brothers. We believe that's still its legal title, isn't it?

JK: Yes, for a very short time in the early 30s, or late 20s, Mr. Shure's brother was in business with him. He just never changed the title.

db: What happened to the other Shure?

JK: He went off and started a company of his own, I think in St. Louis, Missouri. That company prospered. It has nothing to do with our company, but that company is still in existence although Mr. Shure's brother passed away quite a number of years ago.

db: It's interesting that suddenly Shure has gotten a history thanks to some films (notably, Good Morning,

The Participants At Shure

Jim Kogen, President

Sandy Schroeder, Director, Microphone Products

Michael Pettersen, Director, Mixer Products

Alan Shirley, Product Line Manager, Wired Microphones

Chris Lyons, Market Specialist (Professional Products)

Bob Schulein, General Manager, HTS Division

and Lee Habich, Director of Market Development—present but not speaking for interview. He was also extremely helpful to db in the accurate translation from the taped interview to the printed page.

If you wish to write to Shure the address is:

Shure Brothers, Inc.
222 Hartrey Avenue
Evanston, Illinois 60202-3696

Vietnam) which featured early Shure microphones in them. That famous Shure Unidyne, is, of course, not the beginning of Shure's history in the microphone business. You've alluded to the fact that it goes back to the early 30s. In the early 30s, there was radio primarily, and motion pictures. Were Shure microphones being used in both applications at that time?

JK: By all means. A lot of those old microphones that looked like they had a ring with some springs and a circular disk hung in the middle of the ring—as seen in old pictures; we made a lot of microphones like that. We've got a book of photographs showing every President since Franklin Roosevelt using our microphones. Some of those early ones were condenser mics, back in the middle 30s.

db: If we go back to the very early history of the microphone, after carbon, wasn't the condenser almost preceding the dynamic microphone?

JK: It was at least concurrent with it; they didn't have transistors in those days so the amplifier was a great big box.

db: Let's move on to the new things happening with surround sound at this point because **db** got the fax on the Grammy Awards being done with it—is this something competing with or complementing what Dolby is doing in the film industry?

Bob Schulein: It certainly is complementing. The driving force story behind what we are doing is to take this 4-2-4 matrix concept, which is 4-channel on a 2-channel transmission scheme and take it beyond the feature film business into other markets that are now available because they have turned into stereo, or because they are stereo with picture. It is, in our view, a better way to use a 2-channel transmission system. You get more excitement out of it than conventional two-speaker stereo.

db: Can you talk about some of the uses it is now being put to on the pro end of things?

BS: We're at what you might call a beta test phase; we have a limited amount of equipment available for use and the first national use was this Grammy broadcast. Prior to that, we had done some experiments with sports, namely baseball in the Chicago area. We had done a PBS production last summer, which was an outdoor event, called the Great Circus Parade. Present plans are in the area of broad-

Figure 19. The Evanston, Illinois corporate headquarters and manufacturing facility.

cast—a live entertainment element. I can't give you too many details because they haven't been all firmed up. Sports—that I can tell you that we are going to be starting baseball with WGN, which is a superstation in Chicago, or local Cubs games starting in April. Other markets that we see having potential are music only, music video, all types of commercial industrial productions in which an audio-video presentation is the end result.

We have trademarked a word, Stereosurround, which is one word that we are making available to anybody that really wants to use it. The idea is to try to create in people's minds an easier-to-understand term for something beyond two-channel stereo. Stereosurround seems to be a term that most people can inherently understand.

db: There are a number of companies now out there making consumer versions of surround decoders. If Shure is now going to be encoding certain programs, what degrees of compatibility can we talk about that might exist?

BS: This is something that we've been very concerned about. Here is the way we are looking at it: producing in what we call the Stereosurround format is a production involving the most sophisticated type of decoding. If it's a successful production under this basis, then all of the known decoding processes we believe are quite compatible. It's sort of a downward compatibility with mono being at one end and the various logic-type encoders at the top end. Then in between there are non-logic decoders, two-speaker stereo, and then finally mono. It's highly compatible with existing decoding technology.

db: If someone doesn't have that equipment, would it still be possible to get the normal stereo transmission with its own compatibility capabilities?

BS: That's correct.

db: Can we now move on to other sound reinforcement products that are currently in the catalog?

Sandy Schroeder: We've just completed the market introduction of a very significant new dynamic microphone. There are actually two versions—they're called the Beta 57 and the Beta 58. This is the culmination of a program that has been underway here for about five years to produce a performance-oriented dynamic microphone which would make a significant advance in performance over existing mics in high-volume situations.

db: Why does this industry need a performance microphone as opposed to using the traditional cardioids and omnis that have been available up until now?

SS: When I say performance, I'm talking primarily in the sense of live music performance or other sound reinforcement applications that are related to some sort of artist performance.

db: I think what I'm asking is what is different about microphone usage for that application that a standard microphone doesn't fit?

SS: That's a fair question. Generally, sound reinforcement has become a lot more demanding. The quality of sound reinforcement loudspeakers has gotten much better in the last decade. The levels of sound that have to be dealt with on stage have gotten much higher, which means feedback situations are a lot more critical. Namely, I would say the primary factors which have been affected here, and which we have strived to improve, are the fidelity of the microphone and its ability to generate a considerable amount of gain before a feedback situation occurs. Of course, this is directly related to the uniformity and controllability of the pattern of the microphone.

db: Are there other sound reinforcement products that we can talk about at this point?

SS: I think that shortly we'll have things to talk about that we can't talk about yet. One of the areas that we are taking a very serious look at is wireless. We see the long-term impact of wireless as being very significant in our business, and we also see a lot of areas where big improvements can be made. We see ways to make a lot of those improvements, and we are indeed looking at every aspect of this business.

db: The supposed biggest problem with wireless is that when they are needed most they usually don't work.

SS: The equipment is getting a lot better. There have been some significant technology advances in the last few years, not only with respect to reliability but flexibility also, such as giving the user a choice of frequencies never available before. Of course, the airwaves are getting much more crowded as more wireless equipment comes into use, not just wireless microphones but other types of RF-related devices. It does get trickier, and many of the large-scale productions that are being done today for television and also live entertainment are using more and more wireless microphones together than was ever done before.

db: That's true.

Figure 20. An array of Shure products: (upper left) SM83 Lavalier Condenser Microphone and preamp; (center) FP31 Compact Portable Audio Mixer; mics (left to right) SM81 Professional Instrument Condenser, SM87 Vocal Condenser, SM58 Vocal Dynamic, and 545 Multi-purpose Dynamic.

JK: I'd like to interject something—if there's any one thing that has really built our company and our reputation, it's our reliability. The reliability of our products—it's something that's been a major priority with us for a long, long time. We feel the same way about wireless microphones. We don't want to attach our name to anything that doesn't support our reputation for reliability. I think as the years go on, it will be seen that when we do bring out our wireless they will work when they are needed to work.

db: The other side of that area is that there will always be a situation where if a wireless system is made that will work in a 200 foot range, somebody is going to try it at 225 feet, and if it doesn't work, the company gets the blame.

SS: I suppose that could happen. Someone could take a high-impedance microphone and put 250 feet of wire on it, too.

db: There's some of that going on now, and it's only something to stay aware of that any new technology must be fully understood and used within its parameters if it's to work correctly.

SS: Absolutely. We have calls every day relating to this type of problem, and in many cases by walking through the problem with the customer we can determine it's not the wireless equipment at all but how it's being set up, how it's being used, or perhaps interference from interfacing equipment. We find that quite often RF interference is experienced not from the radio microphone but from cabling, or other pieces of equipment that are in the chain.

db: What can we tell our readers about the most frequent problems that come across your desks?

SS: You mean with respect to wireless?

db: No, with all kinds of microphones.

SS: There are many problems. I don't know where we'd even start—obviously equipment compatibility is a big one.

There are still people trying to use a high-impedance microphones into a low-impedance input. Just selecting the right microphone for the job—we have people still trying to use an omnidirectional microphone in a live environment looking to get a reasonable amount of gain out of a PA.

We know from a recent survey of people who have bought our microphones that many users still don't know the difference between a dynamic and a condenser, and between an omni and a cardioid. There's a lot of basic educational problems to be dealt with.

We've tried to address this problem with a number of publications that we've offered recently, some oriented toward sound reinforcement.

Chris Lyons: One of the things that Sandy is referring to is a little booklet called *The Guide to Better Audio* that we recently published, about a month ago. I wrote it, with tremendous help from everyone else here at the company. The goal of the booklet was to provide a basic starter tutorial for people just getting into audio or who need a little refresher in some of the basics such as different kinds of microphones, why to choose certain models and not others, how to interface them with other equipment, what kind of cables to use, and basically decoding the terms and what they all mean. We've offered this primarily so far to the audio/visual industry, working with corporate TV and that type of market. We've been getting calls from educators at colleges, from radio and TV stations—there seems to be a tremendous interest for information out there.

Something else that I wanted to throw into the fray—you mentioned common problems that people encounter. As someone who answers numerous technical questions on our products every day, I can tell you that the biggest problem people have with microphones and other equipment is that they don't read the instructions. If people would just read the manuals that come with our equipment and other equipment, they would get rid of probably 75 percent of the problems that they have, but the fact is they don't look at the information they have, and that's the cause of a lot of problems right there.

db: That's something that exists in every industry, especially the technology industries. I hear this from computer manufacturers, software manufacturers, and from other audio manufacturers, so that's a problem that we don't know how to address anymore.

CL: Neither do we. If you find a way, let us know.

db: One of the things we see at the magazine is an increased need to do more tutorial-type work, because this is what our readers actually want—to learn more and to find out how they use these products. You can't break a rule unless you know the rule. What other publications does Shure have that address these kinds of problems?

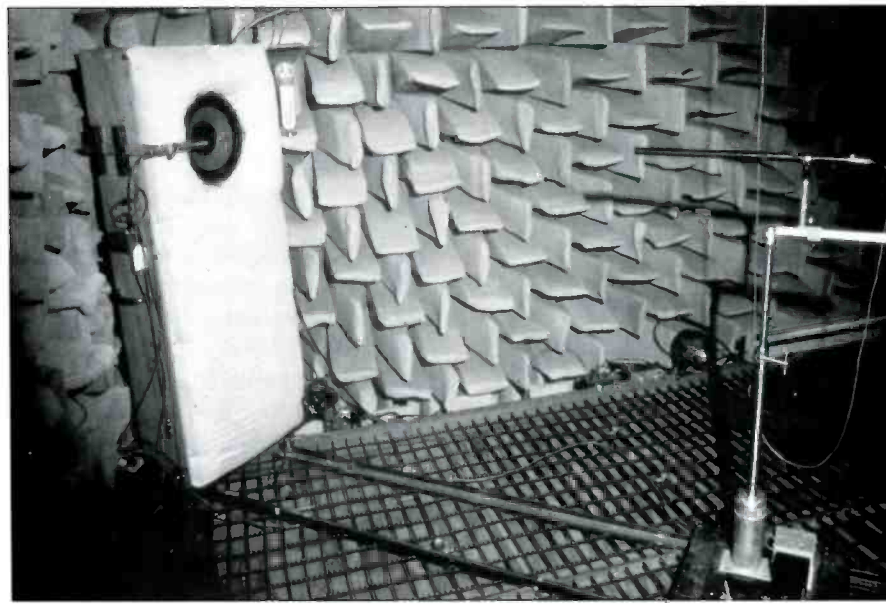


Figure 21. An important tool in the Shure microphone lab is this anechoic chamber. It is also used for loudspeaker and psycho-acoustic research testing.

CL: Well, we have a guide to microphone placement—there's a little brochure called *Mic Techniques for Music* which deals with mic'ing musical instruments and also deals with the subject of stereo mic placement for stereo pickup. We also have three product selection guides that are designed for different markets such as studio recording, broadcast, sound reinforcement, that deal with choosing from our lines of products the correct mics for a given application at various levels of quality and price. Some of these might be handy for your readers to know about.

db: Are some of these with price and some without?

CL: They are all free of charge.

db: So we can tell our readers to write to Shure for the particular application.

CL: The one microphone book on placement, *Mic Techniques for Music*, is about twenty pages long and it's generic. It doesn't talk about any specific model numbers, it just talks about different instruments: where to place microphones and what results are expected. It's a basic guide—it's certainly not the end-all, but it's a good starting point.

db: What I asked about earlier is what other sound reinforcement products is Shure making at this point or contemplating making besides microphones?

MP: Are we moving away from sound reinforcement for musical applications now?

db: Yes.

MP: Okay, because that leads us into a whole other area. We can take it in several different directions. Let's start with broadcast, for instance. Besides making microphones for broadcast, lavalier microphones, hand-held interview microphones, shotgun microphones, and the whole gamut for that, we are also well-known in the broadcast field for mixers. This is interesting because we are not well-known for mixers for musical performance. Musicians know us for the SM 58 and say, you make mixers, too? The mixers that we make are not really designed for that application. They are specifically designed for broadcast.

The M67, which was our first entry in that, was brought out in about 1965. This was, as far as I know, the first battery-operated 4-channel mixer that could actually be picked up and carried around. Some of the first applications this was used for was for on-site work at Cape Kennedy. That got us into this area, and interestingly enough, to talk to a broadcaster about small mixers.

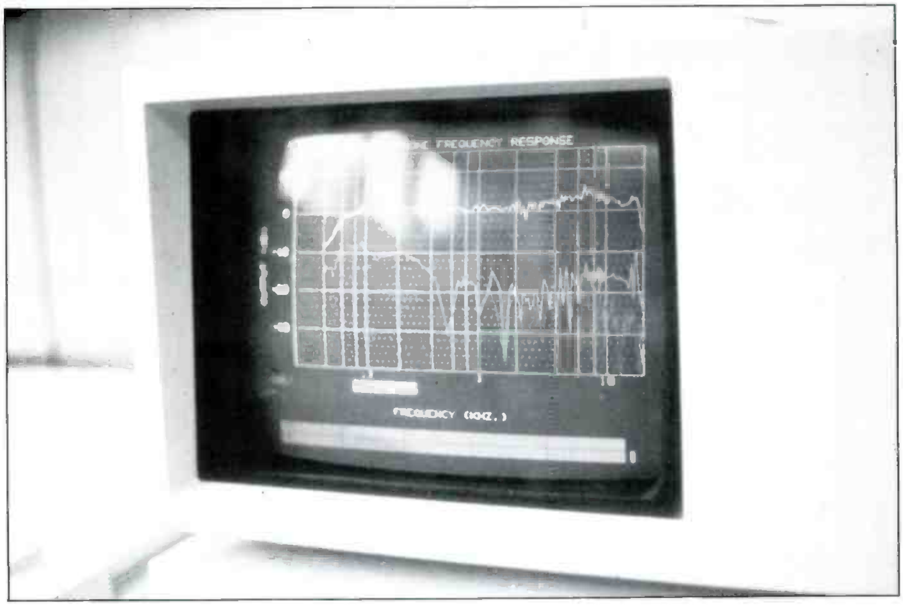


Figure 22. New microphone designs are computer evaluated. The terminal displays an on- and off-axis frequency response of a new design.

db: Well, that doesn't hurt the image any.

MP: It certainly doesn't, but the M67 was actually in the line for over twenty years. It was only recently discontinued.

We now have a M267 which is a 4-channel mixer. It's hard for your readers, unless they are in the broadcast industry, to realize how ubiquitous this product is. You go into a station like WBBM here, which is our news/radio station, and they'll have thirty or forty of them sitting around. It's amazing. They use them for everything—driving headphones, driving phone lines, they need them for an additional four microphones in the studio. It's a building block; I think that's the easiest way to describe it.

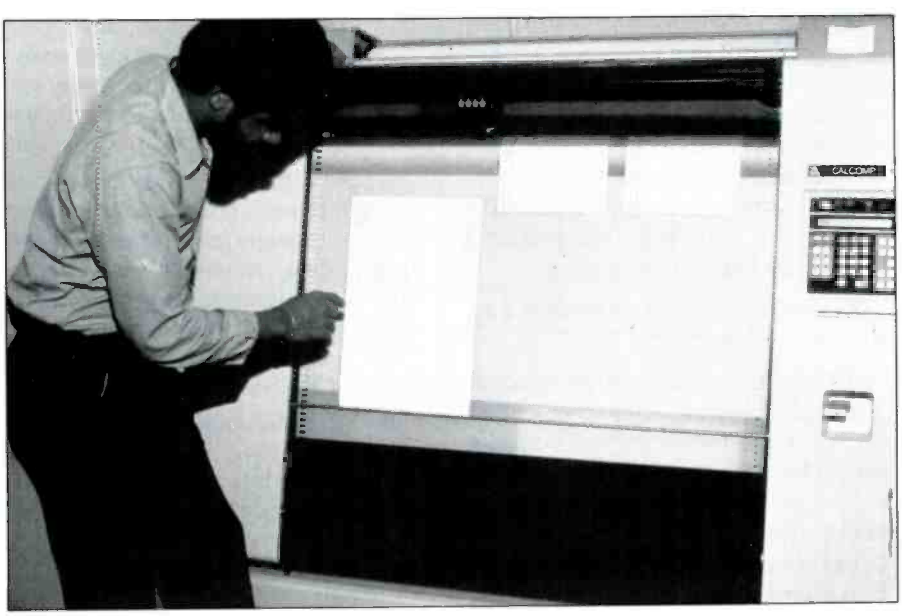
db: But why couldn't they be used, equally logically, in the middle-sized to larger recording studio market?

MP: They probably could, but they are a little bit basic. It's just a four-input mixer, with a mic and line switch, and a volume control and a roll-off switcher, no equalization. It's pretty basic. There's no reason they couldn't be used as a sub-mixer; I have seen them used that way.

I should say that we have another line that we brought out and introduced the first product in 1983. We refer to this as our FP series, Field Production series. Briefly, that has six different devices in it. The one thing that's key is that they can all be run off 9-volt batteries. They are designed to be used in the field. Some of them, which I'll describe briefly again, can be run off a.c. as well. They're all designed specifically for broadcast applications. Starting with the FP11 which is a mic to line amplifier: a mic can be brought in and amplified to line level. If telephone lines are dragged with it—a typical application of that would be doing a golfing remote and they've got microphones out on every tee. Imagine the cost of trying to run seven or eight miles of mic cable...what they do to make their situation cost-effective and not have to worry about destroying it (people with cleats for example walking on the mic cable), is run a pair of twisted telephone cables. To drive these telephone cables is to take the mics and plug them into a FP11, and the FP11 drives the telephone cable which goes back to the main truck. That's an example of how that product would be used. We have an FP12, which is a sister product to the FP11. It's a battery-operated headphone amp. What's interesting about this device is it can monitor a mic or a line level signal but not interrupt that signal. In other words, it can be put in line with the microphone level signal.

A typical example of that would be using a parabolic mic along the sidelines at a football game, and the sound engineer wants to hear what is being aimed at, but not to have that signal end at the headphone amp. That needs to

Figure 23. In another part of the lab a CAD/CAM printer draws the "blueprint" of a new mic design.



go back to the truck. So this allows the user to stick it in line and monitor things.

Now, that product recently has become pretty popular with musical performance people for when they're setting up these large sound reinforcement systems and they want to check out real quick if a line is working or a mic is working. They can carry one of these on their belt with their headphones and check out anything they need very quickly.

db: With the introduction of portable R-DATs going out into the field for all kinds of applications doesn't having small mixers to go with them make a lot of sense?

MP: That's correct. That brings me very nicely, thanks for the segue, to the FP31 and the FP32 which are to my knowledge the smallest professional mixers in the world. We've never seen any smaller, at least not ones that can be operated. The 31 is a 3-in, 1 output mixer; the 32 is a 3-in, 2 output (stereo) mixer. It certainly could operate with R-DAT. It was originally designed to operate with Betacams and video production like that, but it has also found its way into radio applications because somebody can now pack a mic and a mixer in an overnight bag and go out and do a remote. They've done remotes from all over the world with these products.

To fill out the FP line, there is also a battery-operated 1-in, 6-out distribution amp, which is designed for press bridges or any type of application where one needs to take a signal and split it. We also have our FP42, which is a 4-in, 2-out stereo mixer, more like a rack-mounted mixer, something that would be put on a table.

There's also our FP51, a 4-in, 1-out mixer with a compressor in it. The whole FP line in the last five or six years has pretty much established itself—no brag, just fact if one goes into radio and television stations, that's about all that will be seen.

There is an interesting statistic that I heard that might get the point across to your readers. A year or so ago there was a study that I read that when professional users are asked to name a manufacturer of small, portable mixers, 85 percent of them named Shure Brothers first. What's really interesting, however, is that when asked to name a second company, they can't think of anyone. Basically, we are the standard in small mixers for professional use.

Figure 24. The ST3000 Portable Audio Conferencing System.

db: We ought to now move to a kind of final question, so we go back to Jim Kogen's domain if we ask about anything that's coming down the pike in any of these product lines that we can talk about.

JK: That's something obviously that we don't talk about. I can only say that we're doing an awful lot of engineering work, a lot of development work at this time in all the areas we're involved in—microphones, mixers, and teleconferencing (which we haven't discussed). So you can look for many new products from Shure in the coming years.

db: Have we left anything out that we should be asking?

MP: One other thing about sound reinforcement—you may remember about three or four years ago I wrote an article for the magazine called *Broadcast Applications for Voice-Activated Mixers*. In general, I think the automatic mixing market is expanding. To go into a typical application such as city council—in fact, New York City Hall has a fairly large example of our AMF system in there.

As a follow-up to that article, broadcast applications are really expanding for automatic mixing. The last application that was very large, and a lot of people didn't even realize they were listening to our system, was the Iran-Contra hearings which was completely done through our automatic system. It's grown enough in interest that I'm presenting a paper at NAB on this very subject and it's essentially being used now for radio talk shows, any type of situation where there is a large number of microphones. They're sometimes even used on news sets when they've got five or six people talking.

People have become sensitive enough to what comb filtering sounds like and the interference between multiple open microphones—I think you'll see automatic mixing, voice-activation, let's call it that, growing in the '90s. I think the acceptance of this paper at NAB kind of shows that.

db: We didn't talk much about your teleconferencing system, but at the same time, while we have been sitting in our office, you're all sitting around a table which has your system on it. It can safely be said that it's been working splendidly, balances among everyone were perfect. We want to thank everyone for allowing us to do this. db

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The Birth of the German Magnetophon Tape Recorder 1928-1945
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Editel is one company that has decided to do something about changing the image of sound mixing for video.
A Review From

COREY DAVIDSON

A Technically-Minded Musician—Charlie Elgart

One way to tell whether or not there is technical prowess behind an musician's output is to simply listen.

New debut-album artists have the focused creativity and hands-on skills in a recording studio, much less the confidence and blessing of the record company to compose, arrange and produce their own projects entirely by themselves. With his debut album on RCA/Novus Records, Charlie Elgart has successfully combined the most high-tech synthesized sounds with sampled environmental sounds...a difficult thing to do while maintaining a powerful rhythmic and fusionesque/funk-like quality.

Charlie says, "When I was 9 or 10 years old, my brother was a big influence. He was six years older and was a musician/keyboard player. He had 8-track (cartridge) tapes of Sly And The Family Stone, Blood Sweat And Tears, Chicago, Jimmy Smith, J. Geils, Jazz Crusaders...a vast selection of modern music. I was turned on to Blood Sweat And Tears, particularly *Spinning Wheel*. It was the walking bass lines and horn solos that really excited me."

"My instrument was the piano and at the early point in my development, my brother was turning me on to blues. 12-bar blues progressions and boogie woogie bass lines were my introduction to improvisation. Then one summer, when I was 15, my brother had gone away. And while he was away, I worked ever so diligently on my boogie woogie bass lines for about seven or eight weeks. When my brother came back, my father told my brother, 'Hey, you gotta hear your little brother play!' So I started riffing with the right hand and pumping with the left."

"There was always music in the family. My grandfather was a walking encyclopedia of opera and orchestral music."

Where did your experience with synthesizers begin?

"I started with a Multi-Moog analog, monophonic unit. A repair man told me that the ribbon controller would never work accurately so he said that nose grease had to be used. So he took his left index finger and wiped it in the crease between his nose and his face and proceeded to improve his ribbon technique. This worked quite well...especially for a teenager like me."

What was the first polyphonic synth complement?

"The Oberheim OBXa. It was in tune and had plenty of memory that allowed me the luxury of fast programming that was very similar to the early Moog-type structures of synthesis. I found myself drifting away from the piano and more towards electronics. I bought an 8-track machine (Otari 50-50 Mark III) and worked on creating sounds and honing the continuity of my pieces. Peter Gabriel and Brian Eno became strong influences on my sounds."

You have incorporated environmental sounds into your special brand of music. Could you elaborate on some of your sources for these environmental sounds?

"There is a series of sounds that are available on records and CDs. Of course, the CDs are preferred for reproduction. There are all sorts of sounds that one can choose from. The trick is blending sounds in a believable fashion. The psychoacoustic power of certain sounds is a useful tool. The right choices of sounds can evoke strong and subtle responses."

Is stereo a part of the psychoacoustics?

"Of course. However, I like to use stereo in a live context. The live shows that we do are in stereo. The stereo aspect of it is not so much for panning things. It's for the lush spread. The chorusing and the stereo reverb take on bigger dimensions when stereo is employed. Many of our current gigs have been in small club-type venues. Utilizing stereo in these small spaces helps thicken the texture of the instruments. It's very effective. I have a dedicated PCM 42 just for the sax. I also run a PCM 70 for the sax and change the ambient reverb program settings during the performance. There is almost always a slap-delay setting on the sax and the sax chorusing comes from an SPX-90. The reverb is the last in the sax's chain of processing so that the modulation of the chorus is reverberated giving the sax an unusually lush sound. Mack (Mack Goldsbury-saxophonist) loves playing in the echo."

Are the musicians hearing all the processing that is going on?

"Sometimes we run the reinforcement speakers behind us. The sax mics are pointed down and two mics are used...one on the neck and one on the bell. There's no feedback problems because of the angle of the mics. I put EV monitors in front of the band and set up very closely. We are all actually playing in the stereo image. It's almost like playing with headphones on. For Karl (drummer Karl Latham) we have two small earphones that go in your ear, and he has a stereo feed. For me, there are two small reference-type monitors on top of my keyboard rack, facing me. Everyone's experiencing the sounds that are being delivered to the audience. It's like being in the studio."

In listening to your album, it is apparent that a number of things are sequenced and time generated. Does this happen at your live performances?

"No. The things that are sequenced on the record are; the bass line to Float. Most of Sojourn was sequenced only because the budget didn't allow for the

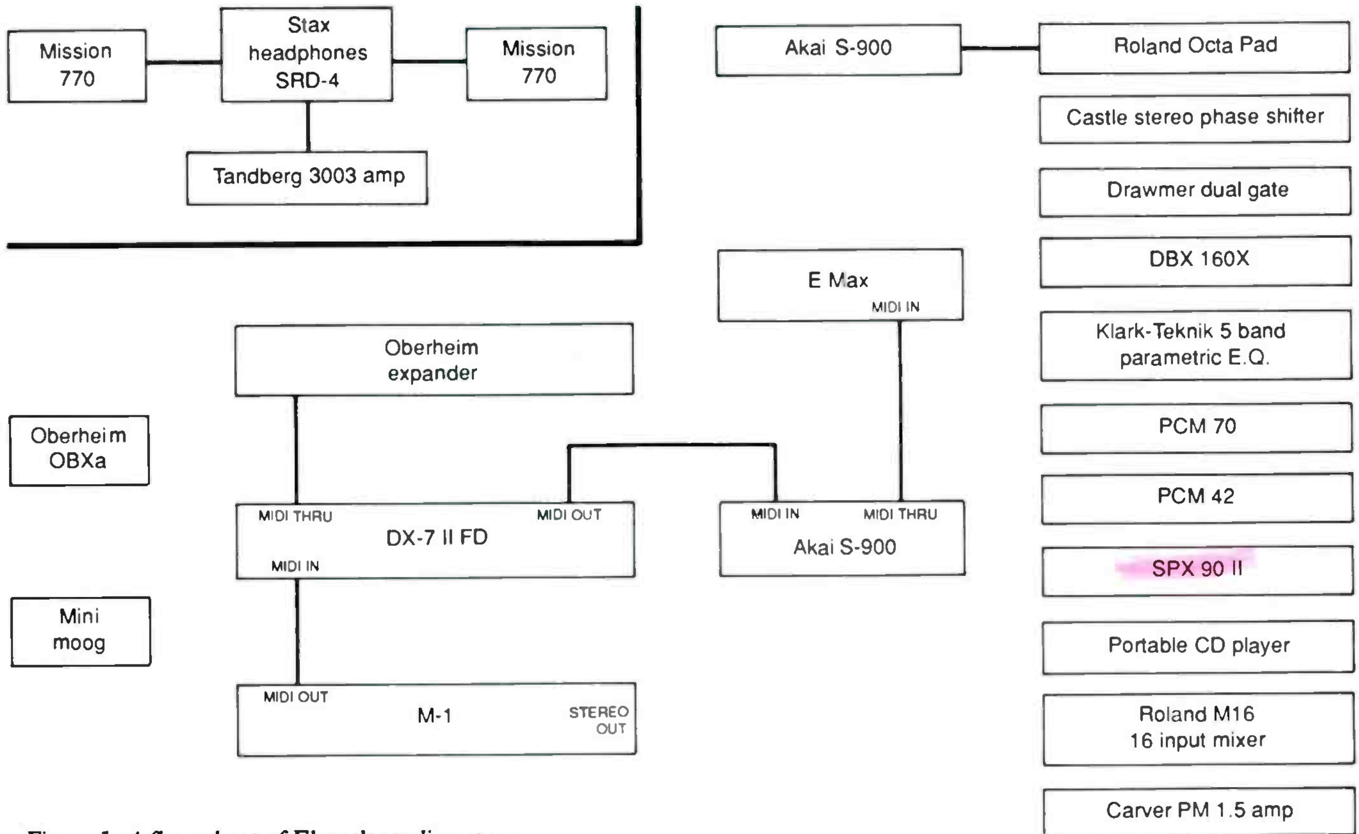


Figure 1. A flow chart of Elgart's studio set-up.

repeated takes of the live rhythm sections. Hopefully the next album will allow for a totally played album. There was one part in This Thing We Share that was sequenced. Live, we don't sequence at all, however, I have sampled sections from the album for the sake of live reproduction. For example, there is a passage from a piece...a flourish, that I sampled in stereo. Because the S900 is only mono, I ran it through the PCM-70, took the two outs and sampled them separately, then assigned them to one note but to two outputs. I pan them left and right. There are some other rhythmic things where I have sampled the whole phrase. Otherwise, I'm trying not to be a slave to the machines. At some point when I have a tech crew or a monstrous tapeless system like the Fairlight, then I'll do it."

Judging by your music, it seems to me that you are inches away from fulfilling a highly technical, possibly tapeless show.

"That's an interesting thought because at this point in my career, there is no equation for the quality of your music to whose supporting you or whose backing you with funds. That's all business. I'd love a Fairlight."

Is it the tapeless manipulation of an entire piece, or is it the vast sound library aspects that attract you?

"Both. Not so much the sound library because I use very few sampled sounds. Most of the sounds that I come up with are realized on the instruments that I now own. I enjoy the art of synthesis and combining the various formats of synthesis to arrive at composite sounds. That's where something like the Fairlight would come in because I would be able to document the composite sounds that I come up with and have the ability to recall them and place them in context. Once we get hooked up with an agency that will put us on the road opening for bigger shows, we'll have the funds to realize some of these great ideas."

How far did you take your early 8-track recordings and did any of that material end up on the album?

"The 8-track was really the growth stage. I acquired all my multi-track skills on that 8-track. I got into tape editing and splicing. I once did a project that was a documentary for an art museum. The curator was speaking and every five to ten seconds he would pause, or clear his throat, and sometimes he just wouldn't know what to say so there would be dead air for moments at a time. So we razor-bladed all of those bad spots. That editing effort took about 15 hours. Those tape-cutting skills were later used in my music. I experimented with editing-in sections and effects. Of course the advent of sampling made those kinds of things much easier and less messy."

Where did the use of effects enter into your musical life?

"I got into a lot of tape delay. I had an AKAI 4000DB open-reel deck and used it for feedback loops, stereo panning and had the tape actually play parts while I played along with it. I would use these early kinds of delays to build a musical piece. I'd play the Mini-Moog into the tape delay system, get that part going, the echo would take over and I could still play on top of it all establishing the premise for a musical idea. Then I worked with combining those things with reverb. I have a Castle analog stereo phase shifter (designed by Ben Cahill) that I used back then and it was an integral part of my little orchestra of effects. I spent hours and hours...years creating that space with those primitive effects. A piano player/song writer named Mark McMillan was one of my early mentors for the use of live effects. When I had seen him years ago, he was doing his show in stereo using two Fender Twin Reverb amps and a Mutron chorus box. So he'd go from his Fender Rhodes piano and organ into the Mutron and into the Twin Reverbs. At the time I had never heard anything like it. So I tried a similar set-up with a Boss stereo chorus. That was way before I had the 8-track. I had a Teac Model 2a mixer and got a lot out of it. I hooked up this little stereo rig with the mixer and the effects and went to heaven."

The Fender Rhodes Piano, as a single instrument, probably generated more effects experimentation than any other keyboard in its day. Even Fender tried marketing things that were stereo in order to image the instrument.

"The Dyno-My-Piano and the Satellite amplifier system were available mainly for the Rhodes. I had a Satellite system that I used live on either side of the stage...it was very effective."

What instruments stick out in your mind as ones that you are particularly fond of?

"The Oberheim X-Pander is really my main axe. Few know about it. I just bought a Korg M1 and I'm using it a lot. An instrument like the M1 is a whole new breed of synthesizer and as a result will give rise to a whole new breed of synthesists. It's not going to be the same."

How do you feel about that?

"I'm not particularly crazy about the direction of process that an instrument like the M1 proposes. The best way for me to express my feeling about this subject is to relate an experience that I had some years ago: At one time I owned an ARP 2600 that I picked up for $200. The thing was beaten up, but it worked. The 2600 turned me on to the process of synthesis. Later when I saw the X-Pander, I was able to relate to a process by which I could create my own sounds. I used the CV and GATE inputs on the X-Pander in conjunction with the DSX sequencer. Although the X-Pander is MIDI, the DSX isn't. So the CV and GATE inputs on the X-Pander had to be used. It was nice working with a system that actually taught me things as I worked. It (DSX) also enabled me to sequence my Mini-Moog. I would sequence a very fast musical figure, put echo on it, and run it through the Mini-Moog. So in the middle of the mix, I would lean over and hit the key on the Mini-Moog, gating the sound that was already running through it. My point is that many of the newer instruments are more sound source oriented and less process/learning oriented. The results from this new era and the impact on new musicians have not really surfaced as of yet."

As your own producer, how do you envision yourself producing other artists' projects?

"I was just called to do a project. It's dance music. Now I'm still battling inside whether or not I'm going to do this. I think I am just going to do it. It's an exercise. However, this person that I'm doing it with wants to be a co-producer. They want to do it quick and get it done, use a MIDI programmer to pull up 30 snare drum sounds, and pick and choose. Now I think a lot of people call themselves producers based on the fact that they want the most current/happening sound. So they want the orchestra hit...they want that killing cracking snare drum...they know what kick drum they want only because they hear it out there in the world already. That's not being a producer. Being a producer is really getting it to tape...in tune...making it tight."

"It is so easy now to make a tape either at home or in a studio and there are certain aspects of the real artistic side of it that they are circumventing. For instance, take drum machines. Drum machines come with samples that are gated with reverbs on them...they're hot...they sound great. Engineers no longer have to come up with those sounds. That's good and bad. For one it's quick, in and out of the studio. I believe that great art takes time. Making the best product that you can is, by far, more important than making something as quickly as possible that has the elements of a quick sale of an item that has a transient level of acceptance. Much of dance music is the latter. The material doesn't entertain us for very long, and that kind of product has the unfortunate ability to promote more products like it. Fewer and fewer people are willing to experiment. Many simply see dollar signs in a quickie production. The sad fact is that this is the trend in music today."

Fortunately, technology has also given rise to a new level of productivity in the more progressive and experimental areas of music and art. Hasn't this been true for you?

"It is the technology that has enabled me to take the studio context and attempt to duplicate it in a live context. As the gear and my understanding progresses, I am more and more capable of realizing some of my wildest ideas without detracting from the live performance elements of spontaneity." db

EQUIPMENT LIST

REINFORCEMENT:
Tandberg 3003 power amp
Mission 770 monitors
Carver 1.5 power amp
Roland M16 mixer
Stax headphone control center

INSTRUMENTS:
Oberheim OBXa
Oberheim X-Pander
Mini-Moog
Korg M1
Yamaha DX-7 IIFD
AKAI S-900 (2)
EMU EMAX
Roland Octa Pad

SIGNAL PROCESSORS:
Castle stereo phase shifter
Drawmer dual gates
dbx 160X
Klark-Teknik 5-band parametric EQ
Lexicon PCM 70
Lexicon PCM 42
Yamaha SPX 90II

Ad Ventures

KEEPING YOUR VOX MAGNA

• With all the electronic gizmos, digital circuits, effects boxes, MIDI, SMPTE, and other devices used in recording studios, some people in the recording business neglect the essential constituent in radio commercial production. I'm referring to the primary organic transducer, the human voice. If you're going to do a lot of this kind of work, you'll use your voice for a variety of demanding applications, whether you're announcing, singing, or portraying characters.

In my career as a producer, voice-over talent, and full-time radio announcer, and in my collaboration with seminar trainers, I've picked up a some useful folk wisdom and I've conferred with leading speech pathologists. Here I've combined scientific and medical facts with everyday practical ideas to offer the following tips and procedures on the care and maintenance of the human voice apparatus.

ANATOMY AND PHYSIOLOGY 101

Take a look at *Figure 1A*. This is a simplified sketch of the anatomy of the human throat, including the larynx, or "voice box," the trachea, or "windpipe," and the esophagus.

The larynx contains two small flaps of thin membrane called the vocal folds or vocal cords. Normally these folds rest along the sides of the larynx (*Figure 1B*). When speaking, they are stretched across the airway and air passing between them cause them to buzz at a particular audio frequency (*Figure 1C*). The resulting sound is conducted throughout the throat, chest cavity, and sinuses, modified by the placement and movement of the lips, tongue, and associated muscles, and then sent out through the air. The sound waves emitted are what others hear as your voice.

Since the vocal cords are so small (only an inch or so in length) they are somewhat delicate and easily affected by abuse. Prolonged periods of talking, shouting, and singing can irritate the vocal cords, and they may become sore, inflamed, and strained. Here are some proven techniques for reducing unnecessary stress on this sensitive part of your body:

DRINK WATER

This is not to moisten your larynx: liquids don't coat your vocal cords. Referring to Figure 1, you can see that the epiglottis acts as a natural valve between the main throat area and the trachea. The epiglottis is open only during respiration. It closes each time you swallow. Otherwise, food and liquids would enter the trachea, pass between the vocal cords, and get into the lungs, causing severe distress. Even the slightest amount of foreign matter that gets past the epiglottis causes choking, coughing, and gagging. Remember the last time you "swallowed down the wrong pipe?"

Consequently, nothing you drink is able to "coat" your larynx; rather, liquids indirectly provide moisture via the internal blood supply to the vocal organs. This is very important. One of the biggest causes of problems with your voice is dehydration, that is, loss of water. Drinking plenty of water hydrates the tissues that make up the larynx.

Don't just gulp a pint of water once an hour; sip regularly at every free moment to maintain a consistent, adequate flow through your bloodstream. The only drawback is that you'll require a few more trips to the bathroom as the day wears on. Hot tea, honey, throat sprays, cough drops, and the like may help soothe soreness in the upper throat and the mouth, but remember that they do not directly bathe the larynx itself.

DON'T smoke, drink alcohol or caffeine, or take antihistamines. These substances act as vasoconstrictors, which means they stimulate the blood

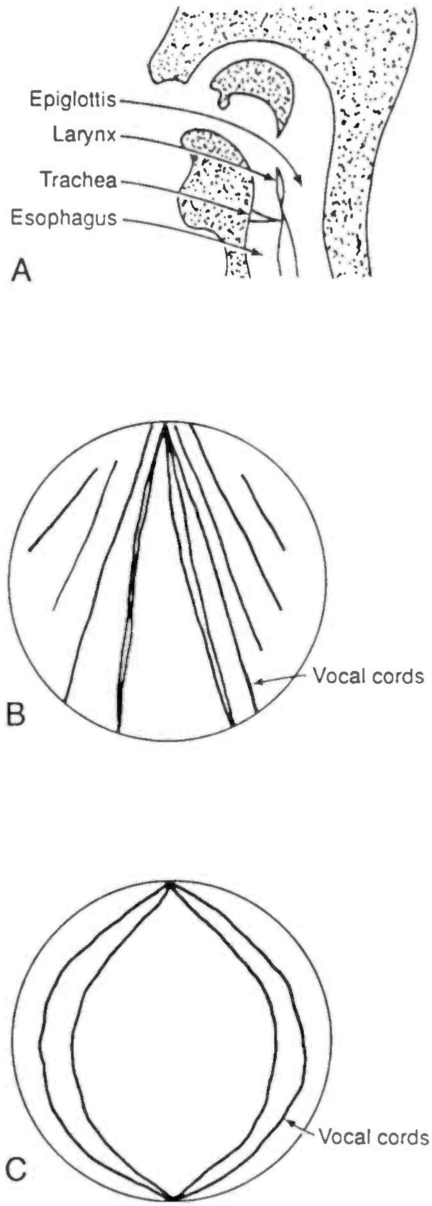


Figure 1. At (A) a side view of the upper throat, while (B) is a side view of a relaxed trachea, and (C) shows it stretched.

vessels to close off some of the blood supply. This actually creates a drying effect in the mucous membranes.

CLEAR YOUR THROAT CAREFULLY

An explosive cough makes the vocal cords slam together abruptly, and can raise havoc with their delicate structure. If you must clear your throat, hum first, then gradually allow the humming to build smoothly to a rumble (say, "mmmMMMHHMMM", rather than an abrupt "AHEM!"). This gently tightens the vocal cords and avoids the shock of suddenly becoming taut and slapping shut against each other.

RELAX

Slacken your throat, shoulders, chest, neck, and jaw muscles as much as possible. If you tense up, you put more pressure on the organs that produce and modulate your voice. Speaking with a tightened chest, for example, decreases the effectiveness of the naturally resonant chest cavity, causing a greater percentage of sound to be projected via your sinuses. This gives your voice a thin, tinny, and strained timbre, and limits the richness of sound. Posture contributes to this as well. Sit upright to avoid compressing your diaphragm and thorax.

SPEAK SOFT AND LOW

Go down to a slightly lower, quieter register and let the gain of your electronic equipment pick up the slack. Dropping your pitch to a comfortable zone, say about a half-octave lower than usual, lets your vocal cords work in a more relaxed state. Keeping your voice volume down a bit saves your energy and allows you to breathe in a more natural cadence. If the mic isn't picking you up as well, move a bit closer.

EXERCISE YOUR VOICE

As with any system, your voice needs proper and regular exercise. Build your comfortable range by singing in the shower, the car, whenever you can. Try to hit a few high notes, but don't strain for those extremes. Get as low as you can, too. (Can you imitate Lurch, the butler on the old *Addams Family* TV series?) A hint: When diving for those low notes, don't go for volume, just a mild rumble will get you used to letting those vocal cords flap loosely.

Never overdo voice exercises. Unless you're working with a professional voice trainer, just try this stuff for a couple of minutes a day to maintain flexibility, not to artificially extend your range a few octaves. If you want to sing *Rigoletto*, get a personal coach.

PACE YOURSELF

Get into a comfortable rhythm with an even rate of speech, rest, speech, rest, etc. Don't go on nonstop for twenty minutes and then take a ten-minute break. If possible, go one minute on, 15 seconds off, one minute on, 15 seconds off, etc. This gives your body a chance to cycle through the working/resting periods more completely. A sporadic approach is tiring, and puts greater demands on your muscles.

BREATHE DEEPLY

Short, shallow, tight breaths limit your ability to take in necessary oxygen and to exhaust excess carbon dioxide. Let your respiration operate at equilibrium. Too little oxygen causes sluggishness, fatigue, and loss of efficiency. Too little carbon dioxide creates an imbalance as well, disrupting the body's pH balance, and producing dizziness, anxiety, and muscle tension. This is commonly taught in meditation, progressive relaxation, and stress reduction therapy. Why do you think hypnotists always have their subjects breathe deeply? Your respiratory system knows what works best. Let it regulate itself naturally and you'll find that it consistently adjusts for peak performance.

CONCENTRATE

Pay attention to content, not technique. You should practice control of your breathing and speaking organs sufficiently to make a habit of feeling relaxed and refreshed, but when actually recording you should focus on the material you are to deliver, be it an advertising script, a set of lyrics, or complex instructions. Do your breathing and speaking exercises outside the studio, so that your unconscious mind can take care of that stuff, leaving your conscious side free to work on the task at hand.

TALKBACK MIC

Received some excellent letters from Gene Josephsen in Chicago, Illinois; Richard LaVoir of Ultra Live, Inc. in Lantana, Florida; Dennis McAtee of KKOW in Pittsburgh, Kansas; John McPherson of AVR Network/Magic 97 in Kentville, Nova Scotia...Hello to Jeff Hedquist of Hedquist Productions in Fairfield, Iowa; and to Dave Chase of The Community of Jesus in Orleans, Massachusetts...Special thanks to: Steve Kiely of Vernon, Connecticut; Mark Lewis of CareerTrack in Boulder, Colorado; Rob Evert of Norwich, Connecticut; and Kevin White, the Mansfield, Connecticut area's finest composer and Allstate agent....Greetings to the members of the Middlesex Amateur Radio Society (MARS)...Speaking of Amateur Radio, if you have a PC with a modem, get online with Dave Corley's BBS at (203) 871-3791. You can also leave me messages via The Willi-Board at (203) 456-1933.

PLUGOLA

I'm thrilled with the volume of orders for my new audio tape program, *How To Produce Great Radio Commercials*. It's a four-cassette compendium of advice, techniques, ideas, and tricks of the trade for putting together creative and profitable radio advertisements. To get your copy mail a check or money order for $99.95 (in U.S. funds) to Porkpie Productions, P.O. Box 176, Colchester, CT 06415. I'll provide free shipping inside the USA. db

BRUCE BARTLETT

Recording Techniques

RECORDING TECHNIQUES: ON-LOCATION MULTI-TRACK RECORDING

• Can a professional, on-location recording be made with home recording equipment? You bet! Here is the story of one such recording, with examples of what to do—and what not to do.

Last Thanksgiving, a band named Retribution phoned and asked me to record them playing live in a club. It was to be a demo tape. The major pieces of available equipment were:

• Fostex Model 80 8-track recorder (a small portable unit with Dolby C)

• Soundcraft 2-channel mixing console

• Revox A77 2-track tape recorder

I drew a block diagram of the recording system (*Figure 1*) and used it to generate an equipment list (*Figure 2*). Note that the list included small-but-necessary details such as adapters, razor blades, empty tape reels, etc.

The direct-connection cables were unusual (*Figure 3*). Each one took the signal from the guitar amps, synthesizer, and drum machine, and sent them at a reduced level to the mixer mic inputs. One end of each cable plugged into the external-speaker jack of a guitar amplifier (or into a synth output); the other end plugged into a mixer mic input. It was a low-cost way to record electric instruments, and provided perfect isolation.

I wanted to make an 8-track recording, but had only a 2-channel board at the time. So I modified the board to have eight direct outputs, as follows:

I installed a row of eight RCA phono jacks in the console panel. Then I soldered one end of a 1-conductor shielded cable to the ground and wiper (bottom and center) terminals of each fader, and soldered the other end of each cable to a phono jack. Voila: 8 outputs from a 2-channel board. I plugged these direct outputs into the multi-track inputs. If your budget permits, however, it's much better to use a 4- or 8-channel board. (The 4-channel Fostex board I'm using now has 8 direct outputs as supplied by the factory.)

The signal from each fader was line level (–10 dBV), which matched the input requirements of the Fostex multi-track. Each fader controlled the recording level of the track it was connected to.

PRE-SESSION PLANNING

The first question to ask a band who wants to record is: "What is the instrumentation?" What are the instruments you'll be playing, and how many vocalists will there be? In this case, the list turned out as follows:

4 Vocals

1 Synth (mono output)

1 Drum machine (mono output)

1 Electric bass

1 Electric rhythm guitar

1 Set of percussion: congas and cowbell

1 Harmonica

1 Electric lead guitar (played through two amplifiers one at a time)

1 Acoustic guitar

How could all these instruments be fitted into 8 tracks? Fortunately, not all the instruments were played at the same time. The bass player occasionally played synthesizer instead. The rhythm guitar player sometimes switched to acoustic guitar. Also, the lead guitarist played only percussion or harmonica on some songs. As a result, the track assignments ended up like this:

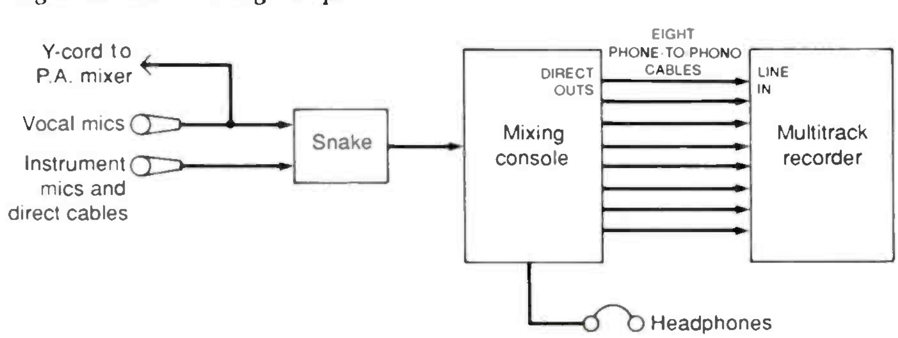
Track

1. Bass or synth
2. Electric rhythm guitar or acoustic guitar
3. Drum machine
4. Lead guitar (guitar amplifier 1 or 2)
5. Percussion or harmonica
6. Vocal
7. Vocal
8. Vocal (2 singing into one mic)

Normally you'd use channel-assign buttons to send two instruments to the same track. But the 2-channel mixing board I was using did not have this feature. Instead, I manually re-patched the input signals according to what was being played at the moment.

Before the gig, I checked the operation of each piece of equipment—a must. Then, on the equipment list, I checked off each piece of equipment as I packed it.

Figure 1. The recording set-up.



ON-LOCATION SETUP

Finally I arrived at the club and found a spot off-stage to set up. Using a heavy extension cord, I connected my power outlet strip to the same outlet that the band's mixer was plugged into. This helped to prevent hum due to ground loops between the P.A. mixer and recording mixer. Then I plugged all my equipment into my outlet strip.

EQUIPMENT LIST

2	CROWN GLM-100 OMNI CONDENSER MICS
1	DIRECT BOX
1	PHONE-TO-PHONE CABLE FOR DIRECT BOX
15	MIC CABLES
4	Y-CORDS (SPLITTERS)
1	SOUNDCRAFT MIXING CONSOLE
1	SONY MDR-V6 HEADPHONES
8	PHONO-TO-PHONO CABLES PLUS 2 SPARES
1	FOSTEX MODEL 80 8-TRACK RECORDER
8	REELS OF AMPEX 456 BLANK TAPE
2	EMPTY TAKE-UP REELS
1	CLEANING AND REPAIR KIT
1	ROLL DUCT TAPE
1	ROLL MASKING TAPE
1	FELT-TIP MARKER
1	POWER EXTENSION CORD
1	FLASHLIGHT

Figure 2., The list of equipment needed.

Next, connections were made between the mixer direct outputs and the multi-track recorder line inputs. I plugged the snake into the mixer and started connecting mics and direct cables.

The microphone list was as follows:

Electric bass: custom-made direct box plugged into the bass.

Synth, drum machine, electric guitars: direct cable.

Acoustic guitar, harmonica, percussion: Crown GLM-100.

Vocals: Electro-Voice N/D 457 (the band's)

Each vocal mic was plugged into a Y-cord. One leg of the Y-cord fed the band's P.A. mixer; the other leg fed my recording snake. In other words, each vocal mic was split to feed two mixers. Normally a mic splitter with transformer-isolated outputs is best, but the Y-cords worked fine in this small setup.

The Crown GLM-100s are miniature omnidirectional condenser units. I taped one to the harmonica's P.A. mic to pick up either harmonica or percussion, and taped another to the acoustic guitar's P.A. mic.

Finally, I plugged headphones into the mixer, cleaned the tape heads, and threaded on some blank tape.

SIGNAL CHECK

Time to see whether everything was working. Using the monitor knobs in the mixing board, I listened to each input by itself. One of the vocal mics sounded weak in the bass and low in level. I suspected that it was a high-impedance mic, and my mixing board's low-impedance input was loading it down. I substituted a low-impedance mic which worked fine.

The signal levels coming from the drum machine and synth were so low that I had to push their faders all the way up. This is bad practice because it results in audible noise. So I asked the musicians to turn up the synth and drum machine to maximum. This is also bad practice! Musicians should not have to change their settings to accommodate your recording (unless the settings are way off).

These signals were low level because the direct-connection cable use was designed for speaker-level signals, not line-level signals. After making this recording, I constructed some new cables to handle line-level signals. I couldn't plug a guitar cord directly from the synth output to my line input because this resulted in hum.

RECORDING-LEVEL SETTING

During the sound check, I asked each musician to play alone so that I could set recording levels. To do this, I set each fader to design center (about ¾ up), and turned down the trim pot for each input just until the LED overload indicator stopped flashing. Some inputs needed no trimming.

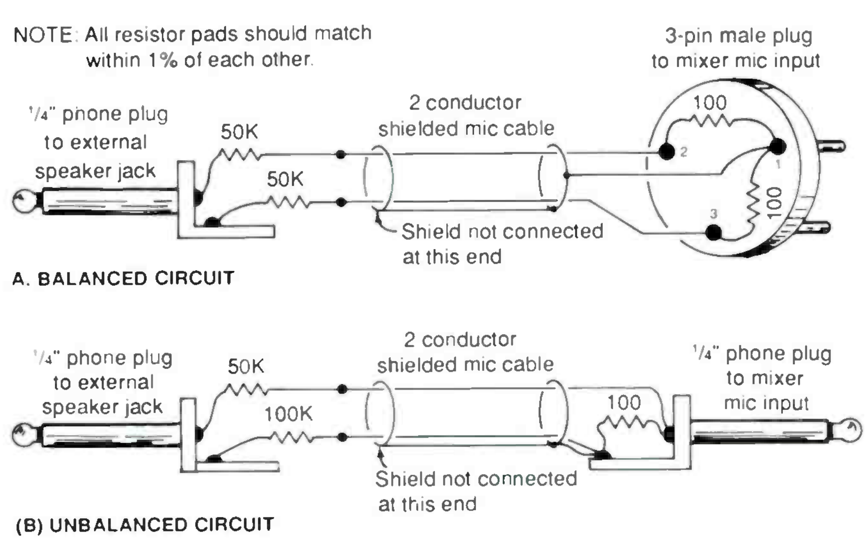
This also resulted in 0 dB recording levels on the multi-track machine.

Time for the gig to begin. I started recording and kept careful watch of the recording levels. Most of the levels needed re-touching, but they were in the ballpark thanks to the sound check.

When an alternate instrument was to be played, I switched input connectors and quickly reset recording levels. A better way, if you have channel-assign switches, is to plug each instrument's cable into a separate input and set the recording level for each input. Then assign the appropriate instrument to its track as required.

While the recording was in progress, I set up a rough monitor mix over headphones to approximate the finished mix, and to listen for any problems.

Figure 3. A method for direct connection to an external speaker jack (courtesy of Steve Julstrum).



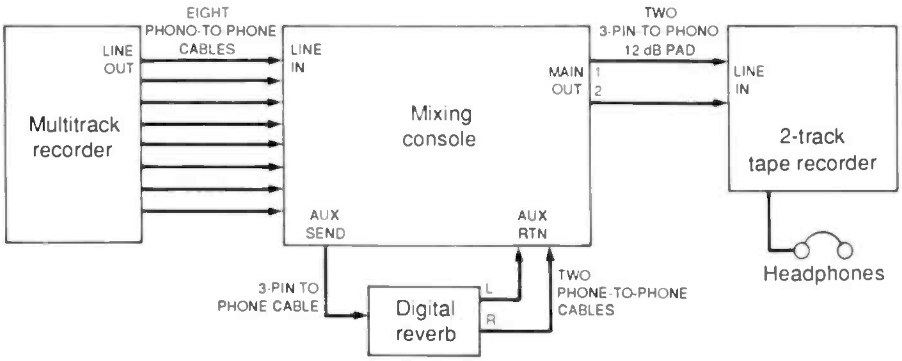


Figure 4. The mixdown set-up.

Also, for each song, I kept notes of what instrument was on what track.

Since the Fostex Model 80 accepts only 7½-inch reels and runs only at 15 in./sec., the recording time was about 22-½ minutes per reel. The band knew about this limitation and paused while I switched reels. A few times, however, they forgot to pause, so I missed the beginning of some songs and had to fade them up during mixdown. This is not good practice. A bigger budget would allow a backup machine connected in parallel with the first. The second machine is started just before the tape runs out on the first machine, and the two tapes are edited together back in the studio.

The next day at home, I set up the multi-track machine to play through the console line inputs (Figure 4). The console main stereo outputs fed a Revox A77 2-track tape recorder. (Ideally, for mastering you'd use a digital audio adapter into a VCR, an R-DAT recorder, or a VCR with VHS Hi-Fi.) In-line with the cable between the console and the Revox was a 12 dB pad (Figure 6). This reduced the console's high output level (called "+4 dBm") to a lower level (called "–10 dBV") suitable for the Revox input.

To add ambience to the dry tracks, an Alesis Microverb was connected to the console's aux send and return jacks.

I monitored the mixdown with a pair of Thiel CS3.5 loudspeakers. These are audiophile speakers for home hi-fi use.

I threaded the first reel of tape onto the multi-track. Referring to my track sheet, I made a designation strip—a long strip of paper placed below the faders, indicating which instrument each fader controlled. Unrecorded tracks were muted.

All console controls were set to "off," "flat," or "zero" so as to have no effect. You have to start from ground zero in building a mix.

The first step in a mixdown is to listen to each track alone and clean it up: erase and filter out unwanted noises. The lead-guitar track had guitar-cord crackles which occurred before the guitarist started to play; I erased them. The rhythm-guitar track was hissy to due a noisy guitar amplifier; I rolled off the high frequencies to reduce the hiss.

Next, I set the tape counter to "000" three seconds before the song started, and enabled the return-to-zero function.

Time to roll tape. I hit "play" and brought up the drum machine track, panned to center. The synthesized hi-hat cymbals were a little dull, so I boosted +4 dB at 12 kHz. A 6 dB boost at 80 Hz added punch to the kick drum. Next I brought up the bass guitar (also panned to center) and balanced it against the drums. The bass lacked definition, so I boosted +12 dB at 3 kHz and cut –3 dB at 250 Hz.

Then the guitars came in left and right. They were recorded with a direct connection to the external speaker jack. This jack bypassed the guitar-amp loudspeaker with its high-frequency rolloff, so the direct-recorded signal was too bright or edgy. I rolled off 12 kHz by –9 dB to restore a more natural tonal balance. The guitars also needed a low-frequency cut of 3 dB at 250 Hz and 80 Hz.

The lead vocal was panned to center and balanced with the instruments. It needed no EQ except for +3 dB at 12 kHz.

Finally, I added the harmony vocals panned half-left and half-right, and blended them with the lead vocal so that everyone could be heard equally, yet no one was louder than the lead vocalist. The harmony vocals required a little low-end rolloff to compensate for the proximity effect of the microphones, as well as a slight high-end roll-off to reduce sibilance (over-emphasis of "s" and "sh" sounds).

With the balance well underway, I added a little reverb to everything but the bass and kick drum. The "small room" setting on the Alesis Microverb seemed to suit the music best because it was recorded in a small club.

The kick drum was on the same track as the rest of the drums. I wanted to add reverb to all the drums except the kick drum. To do this, I rolled off the low frequencies in the reverb-return signal. Since the kick drum is mainly a low-frequency signal, not much reverb was audible on the kick drum.

I practiced the mix several times by hitting the zero return button on the Fostex, playing the tape, and making console adjustments. I made a cue sheet that noted fader changes occurring at various tape-counter times, and practiced these changes. The recording levels were set to peak at +3 VU maximum. EQ and reverb were touched up.

Finally I was satisfied with the mix. It sounded solid; every instrument and vocal was clearly audible; reverb enhanced the recording but was not excessive. The tonal balance was similar to that of good commercial recordings played through my headphones. Distortion was inaudible. The Fostex multi-track contributed very little noise; more was added by the Revox 2-track—but the noise level was still acceptable. By far the greatest source of noise was the rhythm-guitar amplifier.

Now I was ready to record the mix. I started the Revox A77 in record mode, then hit play on the Fostex multi-track. Following the cue sheet, I made fader and EQ changes as needed. The mix-down for the first song was complete.

Finally, I made a Dolby C cassette copy of the master tape to play for the band. I was with the band when they played the tape, which is fortunate because their cassette deck and loudspeakers were inferior and needed a lot of tonal adjustment to reproduce the tonal balance I had heard over headphones. If possible, always carry your own high-quality equipment with you for playback—don't rely on the speakers the band has available.

The band was delighted with the sound but unhappy with their performance. Consequently, they couldn't use the tape, but they learned from their mistakes—and called again for another session. They found out which songs worked the best, and recorded only those songs for their final demo tape.

After several weeks of listening to the demo, one of the band members wanted to completely re-do one song. In my home studio, he did a new bass line, recorded over the harmonica track with a synthesizer, and added new backup vocals over the old ones. The new production totally revamped the song.

As we've seen, some home recording equipment can be used to make a professional-quality demo, as long as care is taken at every step. db

Sound Reinforcement in Central America and the Caribbean, Part II

ED LEARNED

Wednesday, November 25, marked our departure for El Salvador. Our arrival time of 7:30 pm proved to be a concern, however. The country has been torn for years by a nasty guerrilla war, internal political violence, and terrorism. Travel from the airport to San Salvador, about ½-hour away, could be dangerous, especially at night: there have been cases of armed ambush. We were greeted by the heaviest security I'd ever seen during my five years of USIA-sponsored touring. Our group was ushered into a waiting armored van as soon as we left the plane, and driven to a VIP waiting area. PAO (Public Affairs Officer) Jake Gillespie and CAS (Cultural Affairs Specialist) Beatriz de Cortez welcomed everyone with drinks, snacks, and a short briefing designed to calm our nerves. They were especially happy to see me again, and the feeling was mutual; I've always had a great time with them.

I supervised collecting baggage and equipment, and our convoy was soon on the way into town. Convoy? We were accompanied by two jeeps with mounted machine guns, and each of our armored vans contained an armed guard with an Uzi. It was pretty impressive; they even blocked intersections during red lights so we never had to stop. A security guard was even assigned to our floor at the hotel for added safety.

A morning press conference for Don and the group was the major responsibility on Thursday. While this took place, Bea Cortez and I held our own meeting in the hotel lobby with my sound people. Bea had tried to hire the same company that had handled Wayne, but they were booked. In El Salvador, the November-December season is a time for school proms, graduation parties, and general holiday revelry, so it is extremely difficult to find sound systems due to the increased demand for their services. Two different contractors would handle our three concerts. I outlined our needs, and handed out stage layout diagrams and input lists while Bea arranged for load-in times.

With business out of the way, we adjourned to Jake's house, where we enjoyed a sumptuous Thanksgiving dinner with the Gillespie family. Their warm hospitality made for an enjoyable holiday respite, although the armed guards outside their home were a sobering reminder about where we were.

THE CONCERTS BEGIN

Friday's work began at noon, when Bea picked me up at the hotel and drove me over to the Cine Presidente, site of our first Salvadoran concert. This theater/movie house was a huge space, seating 1400; its plaster interior and cavernous dimensions contributed to a reverb time of 2-½ seconds. Everyone was justifiably concerned about the acoustics, and I made a mental note to keep PA volume as low as I could so as not to overly excite the room. The PA system was a collection of Peavey and Altec components (*Figure 1*). Two Peavey CS-800, one Peavey CS-400, and a Yamaha P-2200 amplifier provided power for the PA and the Yamaha monitor cabinets. A Peavey bass head with separate cabinet was procured for Bob. The a.c. was available from US-type receptacles on either side stage wall. The grounds on these receptacles were not functional, and I measured voltage at 122, with occasional drops of no more than 5 volts.

My house electronics package was comprised of a Peavey MD-16 mixing console, DOD R-430 dual 15-band graphic EQ, and a Digitech RDS 1900 digital delay. Once I fired up the PA, I found some work was required to smooth out system response. The low frequency cabinets were run full-range, while the horns were passively crossed over (via internal crossovers) at 800 Hz. The range from 700 Hz-1.5 kHz in particular was *really* hot, and I had to turn down amp sides feeding the low end to even the relative balance between lows and highs. Our concert was an early one, scheduled for 6 pm so a movie could still be shown afterwards. The audience of 800 was more reserved than the wild crowds of Nicaragua, no doubt due to the large diplomatic rep-

Figure 1. House left PA stack, Cine Presidente, San Salvador, El Salvador.

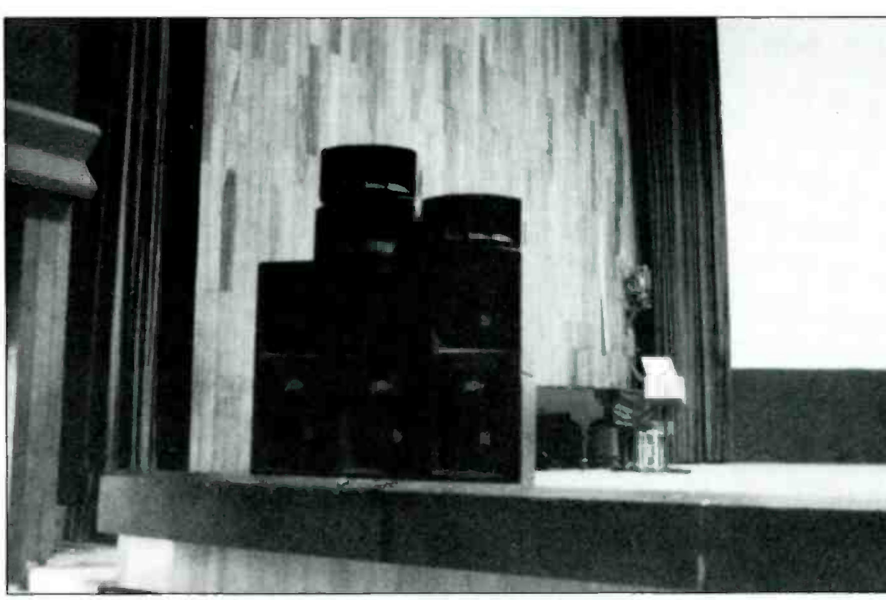


Figure 2. House left PA stack, Jose Simeon Canâs, Central American University, San Salvador.

resentation. IO Joe McManus had me tape the show using an excellent Revox reel-to-reel deck; we enjoyed listening to this tape at a post-concert reception in our honor at the Gillespie's.

Saturday's concert was scheduled for 5pm; Bea picked me up at 10:30 am, and we enjoyed a pleasant drive to the Universidad Centroamericana. Our venue, the Jose Simeon Canâs, was a small auditorium seating 500. It was, unfortunately, another acoustical disaster area: the brick walls, the concrete floor, and the metal roof created a reverb time of 2-⅓ seconds. I checked power, located on the stage right wall: US-type receptacles provided 117 volts, but again the ground lugs were not functional. Sound was due in at 11:00 am, so we sat down to wait, and wait and wait! After Bea made a few angry phone calls our equipment finally showed up at 3 pm, and to make matters worse they did not have the proper complement of mic stands. Apparently, the company we'd hired had over-booked itself, and was forced to sub-contract at the last minute. The school managed to locate a few extra stands for me, but there would be a few mics taped to stands this day.

The PA provided was substantial: it was based around 2 15-inch woofers, 2 12-inch woofers, 2 mid horns, and 4 tweeters per side (*Figure 2*). This system was tri-amped via a Rane AC-22 crossover; a Crown MT-1200, 2 Peavey CS-800, and a Peavey CS-400 provided PA power. Monitors were 2 Yamaha wedges and 2 custom wedges, powered by a Peavey M-2600 amplifier. Bob's amp-du-jour was an Ampeg head with a Sunn bottom. My mixer was a Peavey MK-IV 16x8 monitor console, an Ibanez GE-3100 ⅓-octave graphic (house), MXR dual ⅔-octave graphic (monitors), and an Aria AD-05 analog delay completed the house electronics package. All crossover sides and amp channels were full up; my first task was to back the high and mid levels down to balance out overall system response. A minimum of EQ was used to attenuate the remaining problem areas. Despite the long reverb time, the band felt much more comfortable here than at the larger Cine Presidente.

The predominantly student audience completely packed the place, even sitting in the aisles, which helped dry up the room enough so I could mix more aggressively. They also fired up the band; no one could accuse this crowd of being reserved (*Figure 3*)! Ed had the audience clapping in time during his drum solo, and some improvised dancing by Don brought the house down! Many students stayed on afterwards for autographs and a chat with the group; they were very excited by what they had heard, and several said they'd bring friends to the concert tomorrow at the Metrocenter.

OUTDOOR VENUE

Who says that Tiffany is the only person touring shopping malls? Sunday's concert was an outdoor performance at the largest mall in San Salvador. We planned to use a temporary stage located in one of the mall's large courtyards. Our 4 pm concert allowed for daylight visibility, yet used shadows from nearby buildings to protect the uncovered stage from the blazing sun. My first concern here was the stage: not only was it too small, but it was treacherously unstable. Bea and I managed to locate the mall's director, who soon had a team of carpenters at work extending and reinforcing the stage (*Figure 4*). While the finished stage was not the sturdiest I'd ever seen, it was safe enough for our group to perform on.

The a.c. power came from a light pole about 20 feet from the stage. My PA was the same system I'd used at the Cine Presidente, with 4 bass bins and 2 horns per side this time. The "mix point" was located about 60 feet from the stage, extreme house right, next to a tree for shade (*Figure 5*). I started playing program music about an hour before the concert; this attracted quite a crowd. By show time, we had about 2000 people crowding in the courtyard, covering every square foot of grass and sidewalk; they ran the gamut from holiday shoppers to student fans. They loved the group; Don's composition, *Art Deco*, a tribute to Billie Holiday, was a special favorite (*Figure 6*). Its infectious swing had heads nodding and toes tapping everywhere I looked. Don and Carlos playfully threw some Christmas song quotations into their solos in response to the holiday shopping mood.

MIAMI TO JAMAICA

We left San Salvador Monday, November 30, to return to the U.S. Our tour, however, was far from over; we were scheduled to connect with an Air Jamaica flight in Miami after a three-hour layover. I spent some of this time following our baggage transfer while

Toney arranged a clean office for Ed's scheduled fluid change. We needn't have rushed: our departure was repeatedly delayed.

After a seemingly endless eight-hour wait, we finally departed for Jamaica at 2 am Tuesday, arriving at Kingston's Manley Airport as daylight broke. There were quite a few USIS staffers there to greet us despite the early hour, as their boss CAO L.W. Koengeter, was also on our flight. Despite a chaotic baggage claim area, we soon had our stuff in hand, cleared customs without a search, and drove into Kingston to the Wyndham hotel. We were pleased to discover that Ed's medical stockpile had arrived and was already waiting in his room. Tuesday had been designated as a rest day, and after our epic travel experience on Monday, we needed it!

Wednesday morning involved a full slate of radio and television interviews with the group. I elected to stay at the Wyndham to coordinate plans for our first concert, which would be held in the hotel's ballroom. I was able to dictate stage size and placement, an important consideration since I preferred to "play" to the long dimension of this rectangular room and enjoy a narrower coverage angle. The ballroom seated 500 in row after row of folding chairs. Reverb time was about 1-½ seconds, and I noticed no overt colorations. Electrical outlets were located on the rear and side stage walls: US-type receptacles provided grounded 117 volt a.c. The PA equipment came in around 4 pm; it was provided by the Fabulous Five, a Kingston-area band.

One of the PA guys recognized me; he'd played keyboards for Rita Marley at a show where I'd done sound. I guess my world really *is* small. This PA was based on Carvin single and double 15-inch cabinets for bass, a custom box with 4 12-inch woofers for mids, and 2 Carvin radial horns per side (*Figure 7*). The system was 3-way electronic, using an Ashley XR-77/12 stereo crossover. Two Peavey CS-1200, 2 Peavey CS-800, and 1 Carvin DCA-800 rounded out the amplifier complement for the PA and 4 Carvin monitor cabinets. Mix electronics included a Carvin MX-1602 console, Yamaha SPX-90 and Roland SDE-1000 digital effects, and a Furman LC-2 compressor. The Carvin console had internal graphic equalizers that were used for PA and monitor EQ. The band had scattered during the afternoon, so I got them to come down individually for sound checks.

Figure 3. The band performs at the Central American University, San Salvador. Don is playing on Carlos' flute mic.

EQUIPMENT SHARING

Dean Fraser + Friends, a jazz-oriented pop group, was our opening act for this concert; I arranged for Bob and their bassist to share the bass amp, saving changeover time. Bob came down for his sound check after they'd set up, and totally blew their minds with his tuba playing; a short but sweet jam session soon ensued. Our 8 pm concert was enjoyed by a capacity crowd that included the mayor of Kingston and other Jamaican government officials. I had to mix the show cold because of our fragmented sound check, but things quickly fell into place. After the concert, we all sat through 15 minutes of speeches before tear-down commenced. My last act was to arrange a system for Friday's concert. The venue was purportedly much smaller, so we agreed on a scaled-down system.

The group conducted a workshop at the School of Music on Thursday, which was well attended and well received. Friday's concert was a special matinee show at the University of the West Indies Creative Arts Center. I was over at the school at 10 am to prepare for our 1 pm concert. The hall seated 350, with the wood construction contributing to a pleasant room sound. The seats were steeply raked, and the hall was much wider than it was deep.

Figure 4. The view at Metro Center stage—my carpenters are busy extending and reinforcing it.

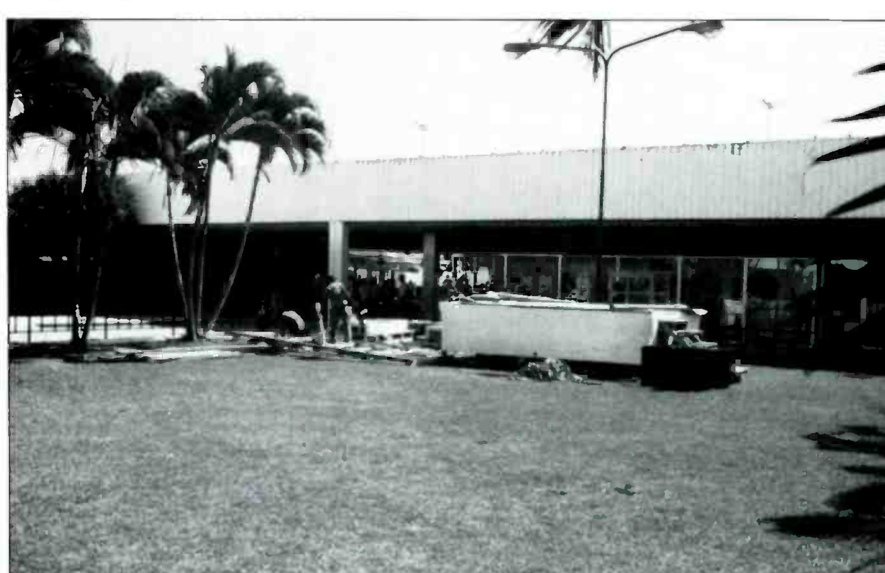


Figure 5. Mix point electronics, Metro Center. I had used the same rig at the Cine Presidente.

My PA system here was 2 Yamaha speakers, each with a 15-inch woofer and a radial horn/driver. Power for these was provided by a Yamaha 12-channel powered mixer with internal graphic EQ. These PA cabinets did not offer enough horizontal dispersion to cover the wide audience area, so placement became crucial.

The hall was so short that stage sound could easily carry the back row; I figured the center audience area would hear the group acoustically. I placed the speakers out on the extreme edges of the stage, panning them in slightly to cover the center/outside and outside audience areas that were furthest from the band's center stage position. Monitors also were problematical: there were 4 monitor cabinets, 2 Carvins and 2 custom boxes. Power was provided by a Crown DC-300A. However, one of the custom boxes had a blown woofer, the other a blown horn. With only two fully operational wedges, we had a problem; I demanded that other equipment be provided, but after a few phone calls it became obvious that this wouldn't happen.

Time was running short, so it was time to improvise. I elected to use the two Carvin wedges for Don and Carlos, as they needed the most volume. Ed could play with no monitor, and since Bob often asked for less highs I gave him the wedge with no horn. I opened this wedge and wired the woofer directly to the input jack, bypassing the internal crossover. This made the woofer full-range, and by boosting high frequencies on the graphic controlling this mix, I was able to get just enough high end for Bob to be happy.

We only had about half a house for the show, but they were more enthusiastic than the packed house at the Wyndham. After the concert, a question-and-answer session about the music was held; two compositions evolved out of this. Ed played an extended drum solo that represented an encyclopedia of African rhythms, and the entire quartet came up with an improvised reggae number, which had the crowd really going.

Saturday was another free day; Toney and I enjoyed it with a trip across the island to Ocho Rios, where we spent a leisurely day exploring the falls and relaxing at the beach.

Figure 6. A close-up of Don and Bob at Metrocenter Mall, San Salvador. Note the tuba mic in the bell and an RE-20 for the pocket trumpet.

BIRD'S EYE VIEW OF THE CARIBBEAN

Sunday, December 6, marked our departure for Trinidad on BWIA's "island hopper." We had a nice aerial tour of the Caribbean this way, but after our 8-hour flight we were thankful to finally disembark at Trinidad's Piarco Airport around 10 pm.

We were met by PAO Mark Glago and APAO (assistant PAO) Bob Dance. Mark knew Don from a previous tour in West Africa; he'd been instrumental in promoting the quartet for a Caribbean tour. After gathering the medical supplies and band equipment, we slid through customs and drove into Port of

Figure 7. The house-right PA stack. Wyndham Hotel ballroom, Kingston, Jamaica. PA amplifier racks are visible behind and next to the PA.

Spain. Our hotel was the Holiday Inn; out two Trinidadian concerts would be held in the hotel's ballroom. I made a trip to check it out with Bob Dance before I turned in; it was a rectangular room with a very high ceiling. There was a permanent proscenium stage at one end, playing to the long dimension of the room. Capacity was 600; the thickly carpeted floor helped damp room reverb to around 1-½ seconds. There were, however, some clearly discernible early reflections.

Monday, December 7, started with a series of media interviews with the group, including an appearance on the Community Dateline program of Trinidad and Tobago Television (TTT). The PA system was supposed to arrive at 2 pm; it didn't actually show up until 4. This system, provided by John Afoon Sounds, was a three-way design. Per side, the PA included 2 double 15-inch scoop-type bass bins, 1 double 15-inch Perkins midrange, a JBL horn/lens, and a 90 degree JBL radial. These components were fed from a Furman TX-4 stereo 3-way crossover; Crown MT-2400, DC-300A Series II, and Amcron MT-1000 amplifiers provided PA power. A Ramsa WR-8724 console, Ibanez 3101 graphic EQ (house), Teac GE-20 graphic EQ (monitors), and an Ibanez DM-1000 delay were provided for front-of-house electronics.

Engineer John Afoon was quite competent, and had the system sounding good from the start. My only quibble was with the monitor system, which was comprised of only 2 TOA SL-15 cabinets. These were powered by a Crown D-75 amplifier run in mono. I gave the two TOAs to Don and Carlos; Ed again went without one. My plan for Bob was to demand an additional small wedge from John; this was run off the bass amp as an extension cabinet, positioned on the floor pointing up at Bob. In essence, he would "mix" his own monitor. We had another local opening act: the Michael Boothman Kysofusion. This was a most interesting 5-piece band consisting of drums, bass, keyboard, tenor pan, and the bandleader on electric guitar. Their music was blend of pop/jazz with the calypso and soca rhythms of Trinidad. Bassist Wayne Bonaparte agreed to share his amp with Bob which was important because Michael's electrified group had lots of gear on stage; we needed to make the set change as quick and painless as possible.

Due to the late PA arrival, time became a little short. After getting two bands' worth of gear on the stage, Michael started to soundcheck. I diplomatically but firmly insisted that we would check first, and despite some grumbling, I got my way. Bob and I hung around afterwards to check out Michael's group; while they practiced, we talked a bit about effects. Bob explained an idea he had for delay on the tuba during his composition *Nonet*: during his solo to open the tune, he asked that I fade in a delay long enough so he could play along with himself, or "layer" a series of notes. I gave him a number of options as to time and recirculation, and we agreed to try it during this evening's show.

We enjoyed a receptive crowd of 400 at our first Trinidadian concert. Michael opened with a 50-minute set; we played 90 minutes after a short intermission. The echo effect on Bob's tuba solo worked out OK, but I thought we could go for a slightly longer delay time. Bob agreed, and gave me some helpful hints as to where I might fade the effect in and out. We also had a band meeting to discuss the overall show.

A MIXED SET

Michael Boothman's band was very electric with a contemporary sound; Don Cherry's quartet was acoustic, with a musical approach new to Trinidadian ears. We felt it would be better to have the electric music follow the acoustic music: a transition from soft to loud is always better than the other way around, especially when the soft music requires attentive listening. Michael had no problem with this, so we agreed to revise the schedule. We also planned a "third set," which would involve both bands playing together on a number. This necessitated a rehearsal before Tuesday's show. Both bands rendezvoused at the ballroom around 5pm, collaborating on Bob Stewart's *Surinam*. While this was underway, John Afoon and I reworked mic'ing and channel assignments to handle both bands at once.

Bob and I experimented with our new delay effect, and we worked out a musical cue that would allow me to get the effect out before the verse began. Tuesday's audience was slightly smaller, but they were treated to a wonderful concert. Don played a beautiful version of John Coltrane's *Naima* on solo piano; Carlos tore out a hard swinging alto sax solo on Thelonious Monk's *Bemsha Swing*. The delay on Bob worked like a charm: I chuckled to myself as I saw the startled audience react, wondering how he could play that many notes! The combination of groups was an absolute smash, as the infectious rhythm of *Surinam* had the audience up and clapping along; many in the back were dancing! Ed and pan player Sydney Joseph locked up in a drum solo that epitomized island swing. The hotel manager called it "the best concert we've ever had!" db

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Lab Report

LAB REPORT: TASCAM MSR-16 ½-inch 16-Track Recorder/Reproducer

GENERAL INFORMATION

• The MSR-16 is a high-performance 16-track, 16-channel tape recorder that uses ½-inch wide tape on 10-½ inch reels and operates at tape speeds of 15 in./sec. and 7-½ in./sec. The recorder features 8-bit microcomputer control for reliable tape operation. Each channel has its own 4-bit microcomputer to control record in/out circuitry, allowing gapless punch-in/out. The capstan motor is a phase locked loop servo type using a 9.6 kHz reference for precise tape speed and control by all major brands of synchronizers. Motion sensing logic provides smooth and fast transition from one tape transport mode to another. The MSR-16 has three different external control interfaces: a parallel port for connection of SMPTE/EBU based synchronizers, an RS-232C port for serial synchronizers and a remote control port for TASCAM's RC-416 Remote Control unit that was supplied to us for our tests as well. The MSR-16 automatically senses when it is under external synchronizer control, switching its servo system between the external source or its internal interface. Track 16 is equipped with a sync lock feature that allows high quality time code or FSK signals to be recorded and played back without interruption.

A digital fluorescent tape counter is driven by a tachometer and displays the distance that tape has moved from a zero reference point in minutes and seconds. The counter can also display speed variation in percentages of normal speed. Pitch control provides a maximum speed variation of ±15 percent, both in record and playback modes.

Rehearsal and Auto-In-Out features permit automated control of punch-in and punch-out times without external computer control. The rehearsal feature can program the MSR-16 to repeat a punch-in/out sequence as many times as you wish before actually executing it on tape, during recording. The distance between the erase and record heads is automatically compensated for, so as to provide gapless inserts. In addition to a conventional return-to-zero function, this recorder is equipped with a two-point locator that also provides a repeat playback over a desired segment of tape.

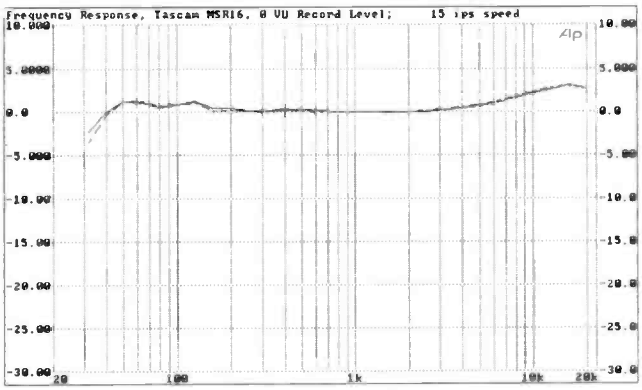


Figure 1A. Frequency response at 15 in./sec.

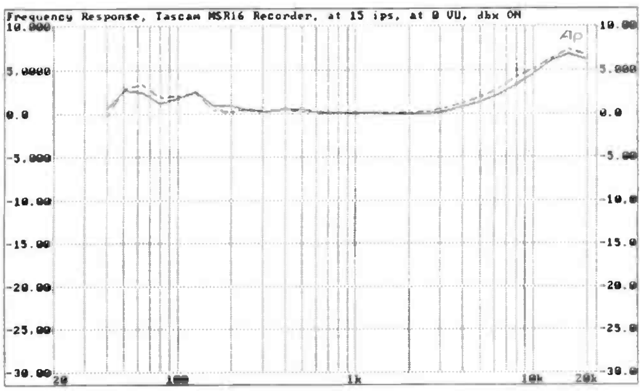


Figure 1B. Frequency response at 15 in./sec. with dbx on.

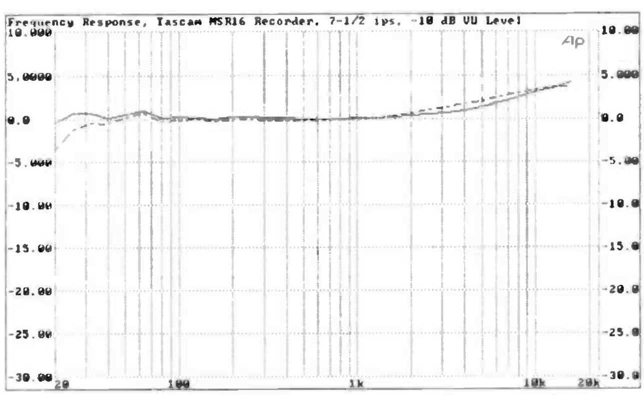
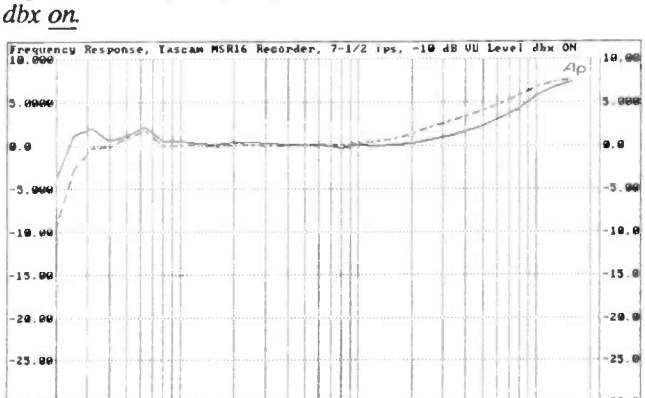


Figure 1C. Frequency response at 7-½ in./sec., –10 dB VU.

Figure 1D. Frequency response at 7-½ in./sec., –10 dB VU dbx on.



Other features of the MSR-16 include LOAD, which prevents tape from running off either end of the reel; a SPOOL mode for uniform tape pack; various EDIT and spot erase features; AUTO INPUT that facilitates communication between studio and control room through the tape returns, and more. dbx noise reduction is also incorporated.

CONTROL LAYOUT

Sixteen peak-level meters mounted above their associated track selector push buttons register signal levels fed to the MSR-16's output connectors over a range from –20 to +8 dB. The main power switch is at the lower left of the front panel, and next to it is the pitch control knob. Nearby is a selector for fixed, variable or external speed control and the speed selector switch as well as the display mode button that toggles the display between running time and deviation (in percent) from normal speed. A sync lock switch optimizes the recording of sync signals on track 16 while turning off dbx on that track if it had been enabled. Two dbx enabling buttons come next (one for tracks 1-8, the other for tracks 9-16.) An Auto Input button, if depressed, switches the output of the tracks in RECORD READY mode to input during rewind or fast forwarding, allowing the control room to hear then talent through the tape monitor for communication without having to change any settings on the mixer. When the next button, ALL INPUT, is pressed, all channels' outputs will carry signals derived from the input electronics, regardless of the transport mode. An "Insert" switch determines whether source or tape signals appear at the output of the tracks placed in the record-ready mode by Track Record buttons. The previously described "Edit" and "Spool" buttons come next, followed by the five familiar tape transport buttons ("Rewind," "Fast Forward," "Stop," "Play," and "Record.") Additional pushbuttons above these major transport buttons take care of such functions as memorizing two automatic location points on a tape, finding those points, and beginning tape play from those points. Rehearsal and other buttons involved in punch-in/punch out actions are also found here, as is the "Load" button that insures that your tape will never accidentally run off the reel during rewind or fast-forwarding. The dual function display is also found in this right-hand section of the panel.

The rear panel is equipped with 16 pairs of input and output jacks. Surprisingly, these are simple, unbalanced RCA type jacks, rather than XLR connectors. At the left of the rear panel are the parallel and serial connectors conforming to SMPTE/EBU and RS-232C standards for linking various synchronizer/controllers. A special connector is also provided for hooking up the TASCAM RC-416 remote control, which has a cable length of about 15 feet and duplicates all of the transport functions except pitch control, edit and spool modes. The RC-416 also provides the capability of programming punch-in rehearsal sequence, locating the MSR-16 transport at specific locations stored in the two memory registers, repeat playback of a selected portion of tape, and other functions. Indicator lamps on the remote control unit show tape speed and speed mode.

Getting back to the rear panel of the main unit, there is also a remote punch-in/out jack for connection of an optional footswitch and, of course, an a.c. IN connector for the separately supplied a.c. power cord.

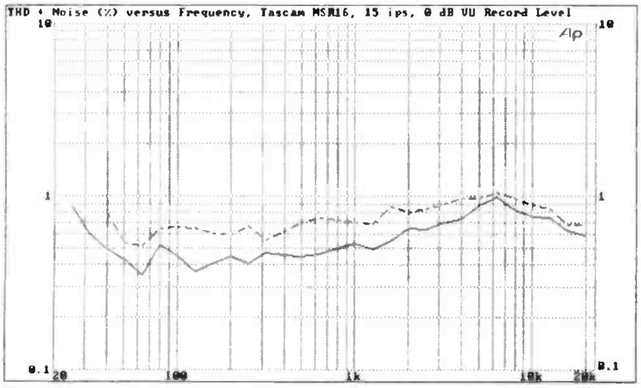


Figure 2A. Distortion versus frequency, 0 dB VU level, 15 in./sec.

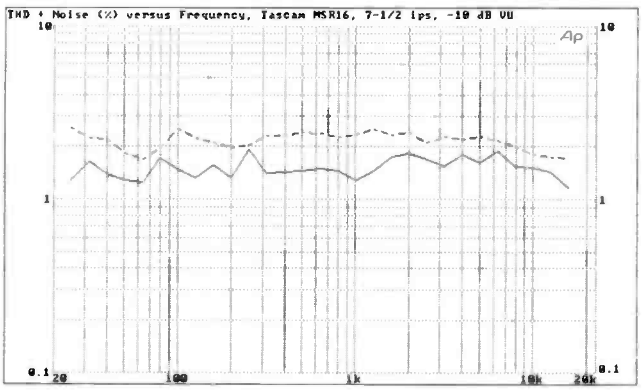


Figure 2C. Distortion versus frequency, –10 dB VU level, 7-½ in./sec.

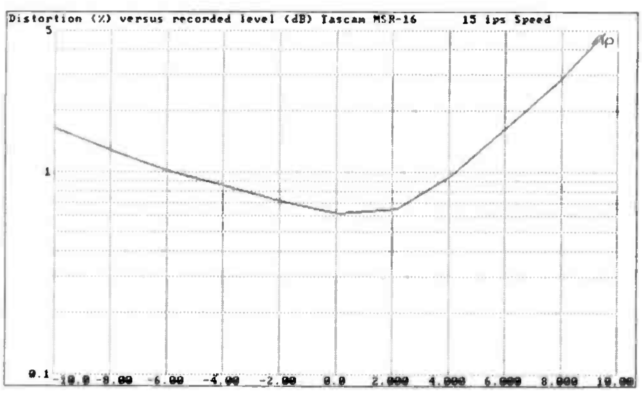
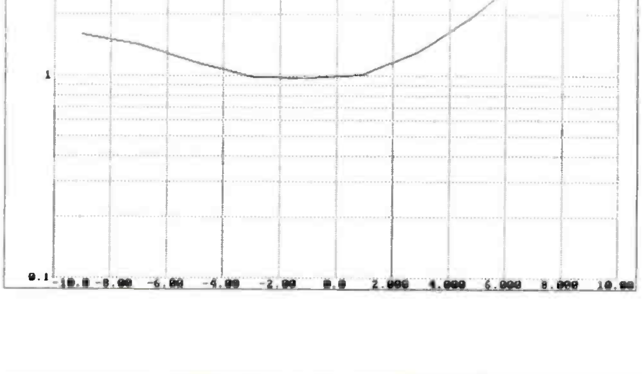


Figure 3A. Distortion versus recorded level, 15 in./sec.

Figure 3C. Distortion versus recorded level, 7-½ in./sec.



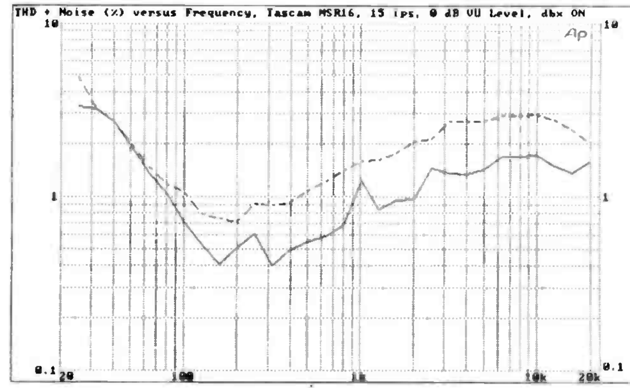


Figure 2B. Distortion versus frequency, 0 dB VU level, 15 in./sec., dbx on.

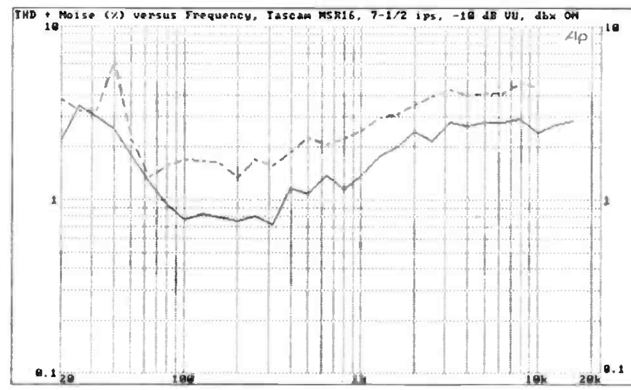


Figure 2D. Distortion versus frequency, –10 dB VU level, 7-½ in./sec., dbx on.

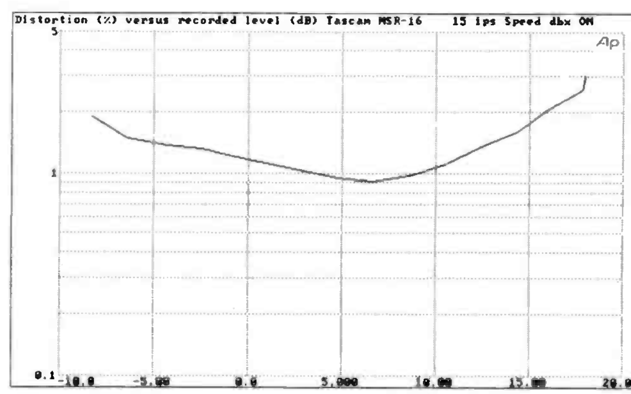
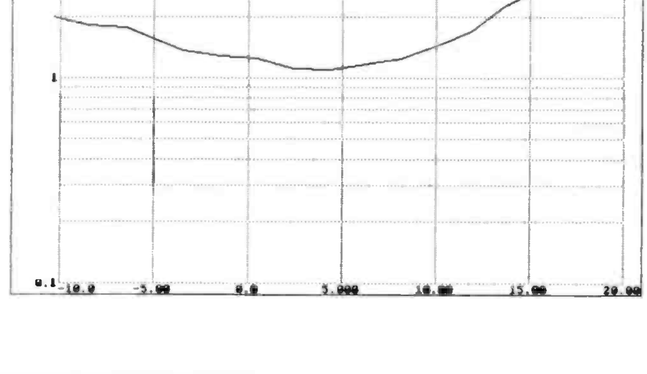


Figure 3B. Distortion versus recorded level, 15 in./sec., dbx on.

Figure 3D. Distortion versus recorded level, 7-½ in./sec., dbx on.



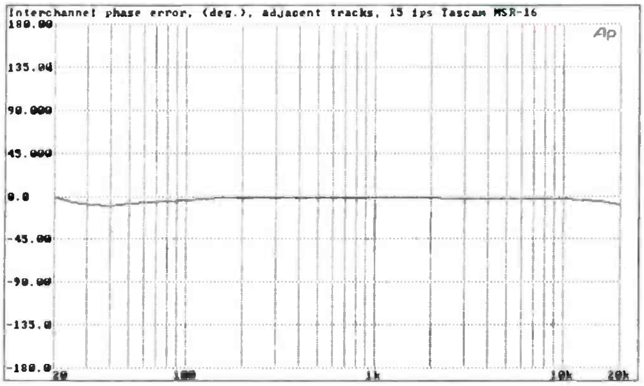


Figure 4A. Interchannel phase with recordings on adjacent tracks.

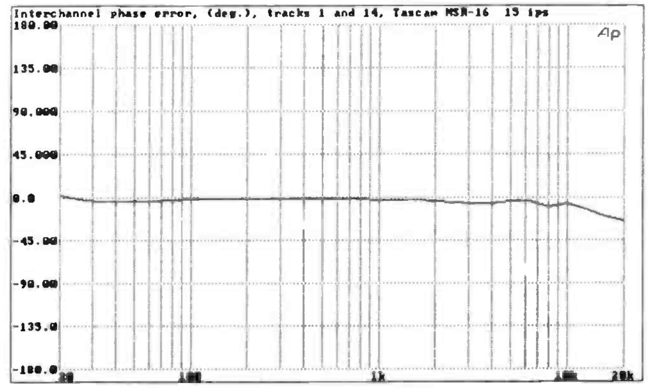


Figure 4B. Interchannel phase with recordings on tracks 1 and 14.

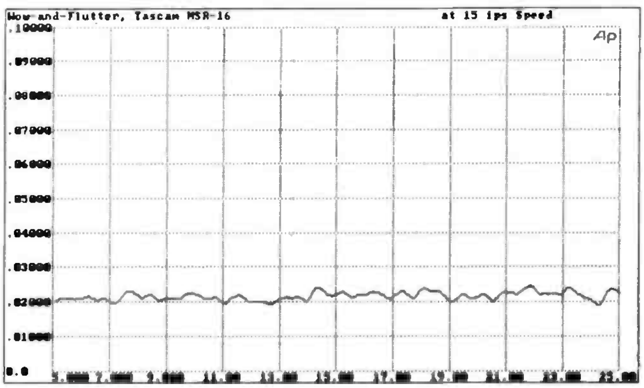


Figure 5A. Wow and flutter IEC weighted at 15 in./sec.

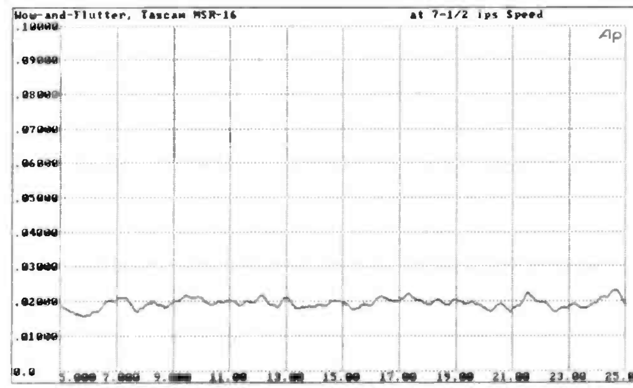


Figure 5B. Wow and flutter IEC weighted at 7-½ in./sec.

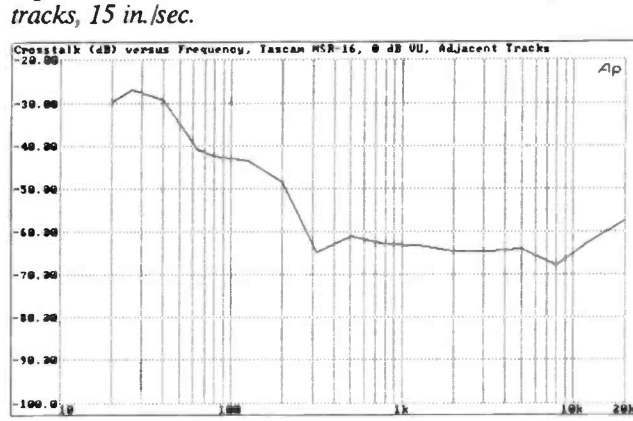
LABORATORY MEASUREMENTS

The MSR-16 has no input level controls and depends upon proper level being fed to it from the mixer with which it is used. In our tests, therefore, we had to adjust input levels of the generator section of our test equipment to meet the input level requirements of the instrument. Note, too, that this 16-track recorder/reproducer uses a single heads for recording and playback, so each test involved first making an appropriate recording and then playing it back while measuring and plotting the test parameters we were interested in. *Figures 1A, 1B, 1C* and *1D* are plots of the frequency response of this recorder/reproducer, using Scotch 250 tape. It should be understood that response will vary depending upon the tape used, and the rise in response at the treble end of the spectrum is due to the fact that we did not use the specific tapes recommended by TASCAM in their owner's manual for theses tests. Obviously, if one wanted to adjust bias and EQ, response could have been made flatter at the high end of the response curves. What is less dependent upon tape type are the slight "bumps" in response at the low end of the spectrum. We purposely repeated the response measurements using dbx (*Figures 1B and 1D*, for the two operating speeds) to emphasize the fact that any deviation in response will be exaggerated by dbx or, for that matter, by any companding system. The lesson to be learned here is to make certain your tape/machine combination yields as flat response as possible before using dbx on this or any other machine. Overall, response was at least as good as claimed by TASCAM, and specifics are listed in the table of *Vital Statistics* at the conclusion of this report.

Total harmonic distortion versus frequency for both operating speeds (and with or without dbx) is plotted in *Figures 2A, 2B, 2C* and *2D*. While THD at 1 kHz was well within specs without dbx, the addition of that companding system tended to increase THD at the bass and treble ends of the spectrum, as shown in *Figures 2B* and *2D*. While it would appear as though the THD exceeded specs at the slower 7-½ in./sec. speed, these curves are plots of THD plus noise, and not just THD alone.

Using a 1 kHz test signal, we plotted distortion versus recorded level for all four operating conditions (both speeds and with or without dbx). Results are shown in *Figures 3A, 3B, 3C* and *3D*. These plots not only show how much "head-

Figure 6A. Crosstalk versus frequency at 0 VU, adjacent tracks, 15 in./sec.



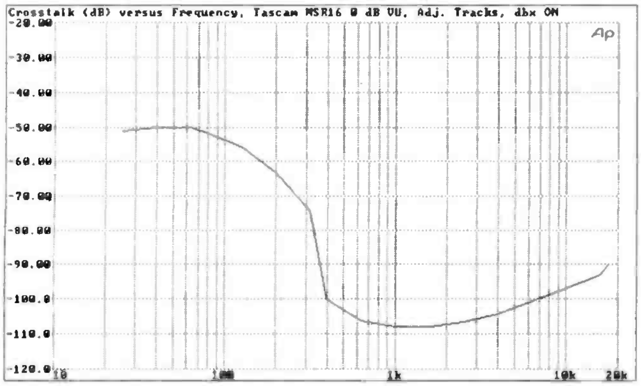


Figure 6B. Crosstalk versus frequency at 0 VU, adjacent tracks, 15 in./sec., dbx on.

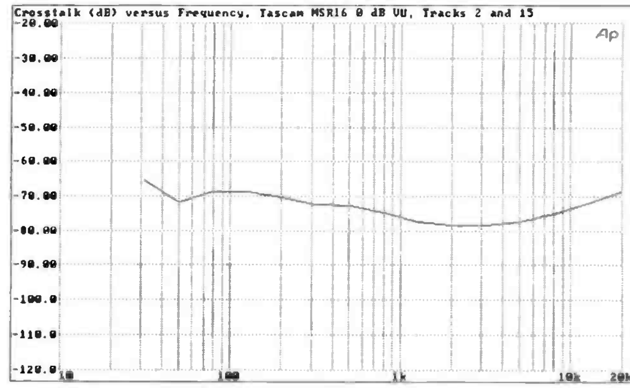


Figure 6C. Crosstalk versus frequency at 0 VU, from track 2 to 15, 15 in./sec.

room" exists in the system, but also help us to determine the 3 percent THD point, used later in determining the signal-to-noise ratios available for the various operating conditions. With dbx ON, the 3 percent THD point was reached at a level in excess of + 18 dB VU at the 15 in./sec. speed and at + 16.5 dB VU at the slower speed.

The latter's display LEDs make it literally unnecessary to be within sight of the master unit to know exactly what's going on with the tape transport...

With our particular tape samples, A-weighted signal to noise ratio measured 63 dB at the 15 in./sec. speed and 58 dB at the slower speed. With dbx turned on, S/N increased to an incredibly high 109 dB at 15 in./sec. and to 105 dB at 7-½ in./sec. We suspect that anyone will be willing to tolerate a bit more harmonic distortion (introduced with dbx) if those kinds of S/N ratios and dynamic range are made available by this companding system.

Since the same tape head is used for recording and reproducing, we didn't expect to find much phase error between channels. Nevertheless, we decided to measure this parameter for a pair of adjacent tracks as well as for a pair of tracks that are far removed from each other. As *Figure 4A* reveals, there was negligible phase error between adjacent tracks. When the same test was made using tracks 1 and 14, (*Figure 4B*) a very slight amount of phase error was noted at the extreme treble end of the spectrum. We would hardly consider a phase error of 25 degrees at 20 kHz to be worth worrying about.

Wow-and-flutter was plotted for a period of 30 seconds at both operating speeds. We used the IEC weighted method and were delighted at the low wow-and-flutter figures obtained: about 0.022 percent at the higher speed and even slightly less, 0.021 percent at the slower speed where we normally would have expected the wow-and-flutter to be somewhat higher. For all practical purposes, these levels of wow-and-flutter are inaudible.

At 15 in./sec., and without turning on dbx companding, crosstalk between adjacent channels measured –63 dB at 1 kHz. However, as shown in *Figure 6A*, at lower frequencies, crosstalk was considerably worse, decreasing to only about -40 dB at 60 Hz. Turning on the dbx system improved things considerably, as shown in *Figure 6B*. Since the crosstalk appearing in the unrecorded channel is at a low level, the action of dbx compresses it downward even further, so that at no frequency was the crosstalk any less than 50 dB and at 1 kHz it actually reached a level of –108 dB. The same sort of measurement was repeated for tracks that were far apart; tracks 2 and 15. Under these conditions, as we would have expected, crosstalk was better than "70 dB over most of the audio spectrum".

CONCLUSIONS

For the small studio that aspires to 16-track capabilities but doesn't have the small fortune needed for some of the more complex multi-track machines, this TASCAM unit offers a great deal of value. Of course, the fact that it is a two-head machine does pose certain problems insofar as studio production techniques are concerned, but TASCAM has compensated for that fact by making it extremely simple to review, edit, and punch in segments. In fact, in many ways, the computerized control of so many of the functions of this multi-track unit puts it ahead of some of the earlier machines that, while able to provide as many tracks, don't offer the sophisticated control provided by the MSR-16. TASCAM does make a 3-head version of this 16-track unit that is capable of real-time tape monitoring, but that unit, as you might expect, costs considerably more than this version. We operated the unit both by means of its front panel controls and using the RC-416. The latter's display LEDs make it literally unnecessary to be within sight of the master unit to know exactly what's going on with the tape transport and the current operating mode of the recorder/reproducer. As has nearly always been the case with TASCAM products, we found the combined owner's/service manual to be complete and well-written, arranged so that the inexperienced user can quickly gain familiarity with the operation of the unit. You'll probably want to check out the descriptions of the many useful optional accessories listed in the manual, too; everything from a MIDI-tape synchronizer to a console rack and even cleaning kits, bulk erasers and demagnetizers. TASCAM continues to serve the needs of the small studio in exemplary fashion.

VITAL STATISTICS

SPECIFICATION	MFR'S CLAIM	db MEASURED
	Mechanical	
Tape	½ inch	Confirmed
Track Format	16-track/16-channel	"
Reel Size	10-½ inch	Confirmed
Speed Accuracy	±0.2%	±0.13%
Pitch Control Range	±15%	Confirmed
Wow-and-Flutter (DIN Wt'd)		
15 in./sec.	0.06%	0.022%
7-½ in./sec.	0.08%	0.021%
Fast Wind Time (2400')	120 sec.	118 sec.
Spooling Time (2400')	400 sec.	380 sec.
	Electrical	
Nominal Input Level	–10 dBV	Confirmed
Maximum Input Level	+18 dBV	+20 dBV
Input Impedance	50K ohms	Confirmed
Nominal Output Level	–10 dBV	Confirmed
Maximum Output Level	+18 dBV	+20 dBV
Output Impedance	220 ohms	Confirmed
Bias/Erase Frequency	145 kHz	N/A
0 VU Record Level	250 nWb/m	Confirmed
Power Requirements 120V 60 Hz,	160W	Confirmed
	Performance	
Frequency Response		
15 in./sec. (±3 dB, @ 0VU)	40 Hz-20 kHz	32 Hz-20 kHz
7-½ in./sec. (±3 dB @-10VU)	30 Hz-16 kHz	20 Hz-16 kHz
THD @ 1 kHz, 0 VU	0.8%	See text
S/N ref. to 3% THD (A-wt'd)		
15 in./sec.	65 dB (108 dB*)	63 dB (109 dB*)
7-½ in./sec.	60 dB (105 dB*)	58 dB (105 dB*)
Crosstalk (Adj. Channels)	48 dB or more	50 dB or better
Erasure	70 dB or more	73 dB or more
General Specifications		
Dimensions (HxWxD, in.)	18-⅜ x 19 x 11-9/16	Confirmed
Price, MSR-16	$7499.00	
RC-416 Remote	$349.00	

* Measured with dbx companding on

Broadcast Audio

Video Tape Audio

• The professional video tape recorder has been with us since the 1950s. For much of the history of this wondrous device, a great deal of effort has been expended on the optimization of its video performance often to the neglect, if not the outright detriment, of its audio performance.

The venerable two-inch quad video tape recorder, which is still with us in spite of the television industry's mass migration to one-inch tape, was not a paragon of audio virtue, notwithstanding the fact that its origins may be traced to the audio recorder. The two-inch recorder's kinship to the audio recorder may be seen immediately in that its linear tape speed is 15 inches per second, a standard audio tape speed. With few exceptions, the two-inch machine has but one broadcast-quality audio channel which, while delivering acceptable performance, could not be accused of "pushing the envelope" of high fidelity.

AUDIO FOR VIDEO

The one-inch Type C video recorder that has become rather firmly entrenched as the standard professional video recording machine in the United States boasts three audio tracks, two of which are well-enough matched to be used as stereophonic audio channels. The overall quality of audio channels on one-inch video recorders suffers from the fact that they are subservient to the machines' video requirements. Everything from the mechanical construction of the tape transport to the alignment of the magnetic particles on the tape itself were designed to optimize video performance. It must be born in mind that at the time the one-inch format was created, audio, let alone stereo, was not the factor in television that it has since become. In view of all this, the audio performance realized from one-inch video recorders is actually better than might be expected, but there is little debate that the they are presently one of the weaker links in the television audio chain. Three-quarter inch video recording will not be considered in this discussion. Suffice it to say that the audiophile will not shed a tear for the imminent passing of this format.

Audio characteristics in need of improvement fall into two general categories: those that relate to audio fidelity per se, and those that relate to stereo, and thus mono compatibility. The first category includes distortion, noise, and headroom capability. The second includes interchannel frequency and phase response. Deficiencies in any and all of these parameters produce a buildup of errors from generation to generation of dubbing which are cumulative in their effect on the end product.

AUDIO SPECS

Table 1 lists some published audio performance specifications for a typical widely-used and highly regarded one-inch machine. The signal-to-noise ratio is shown to be 56 dB below the level producing three percent distortion. The three percent distortion point is typically about 10 dB above the standard one-inch audio reference level of 100 nanowebers per meter. Although actual testing will probably yield a somewhat better noise figure than this, an examination of these "guaranteed" specifications reveals that if audio is recorded at a nominal "zero" level, there is about 10 dB of headroom for peaks, and a noise floor 46 dB below the average recording level. The noise floor rises about 3 dB for each generational pass. Although 10 dB may be just enough headroom to avoid serious peak distortion from tape saturation, a minimum 15 dB of headroom would be preferable. Recall that in our audio metering discussions it was stated that program audio generally contains peaks 6 to 10 dB above the level indicated by a vu meter, and that for either a vu meter or a peak program meter, the highest peaks present in audio program material are around 15 dB above line-up level.

Present-day audio distribution and transmission equipment may be easily expected to have a dynamic range of at least 60 dB, and often 70 dB or more. With this realization, it becomes apparent that the television broadcaster's audio noise limitation lies in the video tape domain. Add to this a so-so distortion figure and wider than desirable frequency response variations, and we are faced with a medium that could stand some improvement.

IMPROVING PERFORMANCE

One way to effect some improvement in the audio performance of one-inch video tape is through the employment of a tape noise reduction system. Such a system will contribute improved headroom and noise floor characteristics, as well as some reduction in distortion products. But there are other things to worry about.

Audio characteristics that relate specifically to stereophonic performance are interchannel amplitude, frequency, and phase response. All these affect mono compatibility, because they affect the integrity of the sum signal produced by combining left and right channels.

The interchannel phase problem is in the final analysis one of differential time delays. Ideally, the azimuth of the gaps of a longitudinal audio head is exactly perpendicular to the linear motion of the tape. If the head gaps are actually tilted with respect to the tape's travel, one audio channel will precede the other in time, introducing an unwanted frequency-selective phase differential between correlated (identical) material present on both tracks. When the two tracks are summed to obtain mono, this phase differential will cause frequency-selective cancel-

	Table I Typical 1 inch Recorder	Table II M-II Longitudinal	Table III M-II FM
Frequency Response	50 Hz-15 kHz +1.5, -3.0 dB 200 Hz-7500 Hz ±1.0 dB	50 Hz-15 kHz +1.5, -3.0 dB 200 Hz-7500 Hz ±1.0 dB	20 Hz-20 kHz +1.0, -2.0 dB
S/N	56 dB below 3% distortion	55 dB below 3% distortion	
Dynamic Range			>72 dB
Distortion	<1.0%	<1% (1 KHz)	0.6% (1 kHz)
Wow/Flutter	<0.01% rms	<0.15% rms	0.005%
Crosstalk	-56 dB	-50 dB	-60 dB
Phase Differential	±15° at 15 kHz	±15° at 15 kHZ	0

lations of certain components of the sum signal. This problem often surfaces when tapes are interchanged between machines, and is cumulative from generation to generation. It is only relatively recently that some manufacturers of one-inch video recorders have begun to provide a means for azimuth adjustment of their audio head stacks. There are also devices on the market that either manually or automatically compensate electronically for the audio time base errors caused by azimuth misalignment.

In addition to such efforts to improve the performance of the longitudinal audio tracks of the one-inch video recording format, some completely new approaches to audio recording on video tape have been developed.

An example of the future direction of audio recording on video tape is seen in the M-II recorder. M-II is one of the two relatively new half-inch professional video recording formats. Along with component video recording, serious efforts have been to improve audio recording on M-II. *Table II* lists the published specifications of the two longitudinal audio tracks of this format. Although the longitudinal tape speed is slower and the tracks are narrower than those of one-inch tape, the use of metal particle tape and other improvements deliver basic performance from these tracks essentially equivalent to their one-inch counterparts. These tracks additionally incorporate Dolby C noise reduction, which materially improves their dynamic range, and gives them a significant performance edge over one-inch.

The M-II recording format also has two FM audio tracks. *Table III* lists their performance specifications. The FM audio tracks offer significant improvement over longitudinal audio recording in dynamic range, distortion, wow and flutter, and crosstalk, and they eliminate the potential for audio time base (phase) errors that is an integral component of longitudinal audio recording. On the down side, FM audio is recorded along with the video signal and may not be edited separately from it, restricting its usefulness in certain applications. In those situations where FM audio may be used, it offers a material performance advantage over longitudinal audio.

The M-II format and one of the one-inch recorders offer the option of two channels of PCM digital audio. On the PCM one-inch machine, the digital signal is time-compressed and recorded in place of the sync track, a track which is usually used for recording several vertical interval lines that are missed when the video head makes its transition from one helical track to the next and is actually off the tape. In the M-II recorder with the digital audio option, the tape wrap around the headwheel is extended an additional 20 degrees, and in this extra wrap a time-compressed PCM signal is recorded.

The M-II PCM audio option sacrifices one longitudinal audio track giving this machine five audio channels. The digital audio recorded is sampled at 48 kHz and with 16-bit linear quantization, so a very high level of quality may be expected. It should be pointed out that the digital recording option for M-II has been implemented for machines recording in the PAL format, but not for those operating in NTSC, and that very few machines incorporating digital audio in either the M-II or one-inch formats have been sold.

Digital audio on videotape offers the ultimate in audio performance, full editing flexibility, multi-generational dubbing without degradation, and the absence of wow and flutter or interchannel phase errors. All of the new digital video formats, D-1, D-2, and D3, incorporate digital audio, as might be expected.

Although there are some problems and compromises involved with recording stereo audio on video tape, significant improvements are being made in the traditional longitudinal audio tracks, and new, better methods of audio recording are producing even better audio from video tape. It may be anticipated that the future will bring nothing less than excellent audio, and video, from video tape sources. db

JOHN BARILLA

THE ELECTRONIC COTTAGE

HOT TIPS FOR THE HOME STUDIO

• 1. *Getting A Warmer Sound From Your Microphone.*

So you can't quite afford an upscale studio microphone just yet. Until you can buy the real thing, here's a way to get a warmer, more expensive sound from almost any decent microphone. All you need is a programmable signal processor (something in the league of Yamaha's SPX 90 will do just fine). More expensive mics tend to do nice things with the harmonic structure of the input signal—a most critical factor with the human voice. This is especially true of tube-driven condenser microphones. Whether the result is accuracy or flattery is a debatable issue, but it sure sounds good.

Some of this effect can be simulated quite nicely by radically adjusting the parameters of an Early Reflections type program on your signal processor and subtly mixing it in with the mic signal. To try this, select (if you can) a "natural" algorithm from the Early Reflections menu (something like a "room," rather than a "random" or "plate" type sound). Then adjust the parameters so that the "size" factor and "hardness " factor (reflectivity) are as small as possible. If there is any predelay, remove it. In other words, make it into a very small, acoustically dead room. Then, boost the level of the "first reflection" so that it totally dominates the program—something like 85 percent. Mixing this in with the mic signal while recording will give a new warmth to your microphone and add increased intelligibility to your vocal sound.

2. *Making A "Running Master."*

No matter what format you mix down to, it always pays to make a safety copy in order to preserve the quality of your original master tape. If you are making a cassette run—say forty or fifty copies of your studio demo for promotional purposes—you may blow some of that precious oxide right off your master before you're done. Highs seem to diminish first, leaving you with a mix that lacks the original luster it once had. So play it safe and make a secondary running master.

Making a running master has many other benefits besides preservation of your original. Just as a mastering engineer tweaks the sound for a record or CD, so can you tweak up your mix and print it to your running master. Then, when you make your tape run, you can just patch it in with as short a wire as possible and forget about it.

Here are some obvious and some not-so-obvious things to do when you are making your running master. If you study your mix on different speakers, you may find that a certain amount of equalization may be indicated. Limiting is also helpful in creating a "hot" master (maximum signal-to-noise). Using an aural exciter may also perk up your mix a bit. Beyond these obvious things, it is also possible to rehabilitate an ailing mix by adding a little extra reverb or echo. (You don't have to tell anybody what you had to do to get your tape to sound so good. Just do it and keep it to yourself.) Almost anything is worth trying at this stage if it actually improves the sound and does not add a significant amount of noise. It can be a bit of a trade-off at times. The balance point is, of course, up to you.

An acceptable medium to record a running master is a metal cassette with Dolby C or dbx. You will find the level of quietness with that combination to be quite acceptable. Just make sure you use a brand of tape that is compatible with your cassette deck. It does make a big difference. You can sometimes find this out from the manual that came with the unit or by calling the manufacturer. (If you know how to do an alignment on a cassette deck, then do it for the tape of your choice. Otherwise find out which tape the manufacturer used for its initial calibration and stick to it.) If you take a little time to make a good running master, it will save you lots of time later on when you need multiple copies. It will also preserve the integrity of your original master tape.

3. *Two For One: Stereoization of A Monaural Track.*

Ever run out of tracks? (No, not you, right?) How about tracking some difficult background vocals for several hours and wishing you had time to double it all. But alas time (or money) runs out.

Artificially synthesizing a stereo double can really do the trick, if it's done properly. Here's how it's done. Pan the "real" track radically to one side. Then split the signal in that channel (using a "mult" on your patchbay or a "Y" cord) so that a parallel signal can be fed to a delay line of some sort. Return the delayed signal—only the delay, not a mix—to a separate channel hard-panned in the opposite direction. Set the delay time to achieve the maximum stereo separation, but don't let it sound sloppy. Usually short notes benefit from shorter delay times (40-100 ms.), whereas longer flowing lines (string pads etc.) usually require longer delay times (100-250 ms. or so) in order to be perceived as stereo.

Changing the timbre slightly (on the synthesized track) with EQ will also help "convince" your ears that they are actually hearing a stereo image.

Got any brilliant ideas you would like to share with db readers? Tell us about them. While we can't guarantee that your idea will end up in print, if it does, we will acknowledge who sent it to us. db

THE SOUND ENGINEERING MAGAZINE

BUYER'S GUIDE

CONSOLES AND MIXERS

On the pages that follow, we present this issue's Buyer's Guide on consoles and mixers. The information supplied is furnished by the respective manufacturers. Further, if a manufacturer you seek is not listed, the chances are strong that as many times as we tried we could not get information from them.

ALLEN & HEATH

SR series is a low-cost mixer designed for sound reinforcement and 2 and 4 track recording. They are available in five configurations ranging from an 8x2x1 through a 24x4x2x1. Features include four-band EQ and 4 aux. sends per input channel, 2 additional FX return sections.

CMC SERIES has a 16-bus in line format designed for "home recording." Bus assignment and mute functions are under microprocessor control. With external software these functions can be controlled from SMPTE time code.The CMC 24 can accept up to 40 inputs, the CMC 32 can have 56.

Scepter 12-channel, 20-input rack mixer features Alps faders, direct outputs on all input channels. Balanced outs are provided on a left, right, and mono channels. Separate mono and stereo tape outputs are also provided.

Sigma consoles are 24-bus in-line consoles with input configurations up to 56 inputs. Each module can process 2 separate input signals, microprocessor-controlled MIDI muting is standard, EQ is 5-band, there is 8 knob aux. send into a maximum of 32 buses. Full patch-bay systems are also available. The console ranges from 16x16 to 56x24.

Saber 16-bus consoles are designed specifically for 8, 16 and 24 track recording and sound reinforcement. All versions include microprocessor controlled MIDI muting of all inputs. Input channels include 4-band EQ with switchable high- and low-frequency ranges and 2 bands of mid sweep plus high-pass filtering. Aux. sends include 6 buses, all recording versions have 16-channel tape monitoring. 24-channel tape monitoring as well as patch bays are available.

SRC Modular Series is an advanced version of the SR series with Alps Glass Smooth faders and fully modular design. Modules can be intermixed and include stereo input, fixed frequency EQ, sweepable EQ. A rack-mount power supply is standard.

SRM Monitor Mixers are available in 18 or 24 input configurations. This series is designed specifically for on-stage monitor application. All units include a built-in splitter and road case.

System 8 is a recording console but can also be configured for sound reinforcement. It is 8-bus for 8 and 16 track recording or MIDI recording. All versions have Alps faders, direct outs on all input channels and 16 channels of tape monitoring that can also be used as additional FX inputs when used for sound reinforcement. Configurations are 16 or 24 input. EX7 is an optional 8-input expander.

ALTEC LANSING CORPORATION

231A is a sound reinforcement mixing console with 12 mic/line inputs. Inputs may be mixed into 4 sub-mix channels, directly into the stereo mic and into the monitor, effects or auxiliary mix channels allowing a variety of simultaneous mixes to be created.
Price:
$5,276.00

1678C is a sound reinforcement automatic microphone mixer with 8 mic inputs (2 can be line inputs). Rack mount. Patented gain sharing principle holds total system gain constant automatically. Xfmr balanced inputs and output, priority, mute, TTL logic, linkable.
Price:
$3,376.00

1692B is a sound reinforcement mixer/preamplifier with 6 mic inputs (2 can be line inputs) and 2 main outputs. Rack mount. Switching is provided to assign any input to either output or both. Output VU meters, xfmr balanced inputs and outputs, tone controls, link.
Price:
$1,790.00

1699B specs are similar to 1692B except for 1 main output and weighs 12 lbs. Can be used to extend the 1692B.
Price:
$1,330.00

1689A is a sound reinforcement mixer/preamplifier with 2 mic/line inputs and 1 main output. Rack mount. Xfmr balanced inputs and main output, unbalanced line output for each input, linkable, tone controls. Single rack space.
Price:
$798.00

1700B is a sound reinforcement mixer/preamplifier with 6 configurable ports and 1 main output. Ports can be made into balanced inputs or balanced outputs. Rack mount. Built-in compressor/limiter, tone controls, linkable, automatic priority, mute, RVC.
Price:
$852.00

AMEK/TAC (TOTAL AUDIO CONCEPTS LTD.)

Bullet is a 10/4/2 modular free-standing or rack-mounting multi-purpose console with hard busing system, hybrid circuitry, 15 segment LED metering, 4-band EQ with swept mids, 6 aux. send buses with 4 stereo returns.
Price:
$4250.00

SR9000 Superconsole is for sound reinforcement and has 42 input modules (each with 2 mic and 1 line input), 16 subgroups, 8 VCA groups, 8 mute groups, 16x8 matrix, fully parametric 4-band EQ, swept HP filter, 16 aux. sends, in-place solo feature and 24-channel extender option.
Price:
$83,215.00

Angela 24-bus in-line recording console comes in three chassis sizes, 24 to 62 inputs. Features include: on-board or external jackfield, 4-band semi-parametric EQ plus HP and LP filters, 6 aux. sends, 6 aux. returns and has extremely flexible/comprehensive routing possibilities and automation options.
Price:
$40,471.00-$99,664.00 plus automation

Classic broadcast and general audio reproduction console is of a modular design and flexibility allows it to be configured to fit most applications. With 4-band EQ, swept HP and LP filters, 8 aux. sends, balanced busing, optional dynamics, optional multi-track monitor modules, this console offers automation options, including the GML moving fader system.
Price:
$75,000.00-$170,000.00 plus automation

G2520 studio/post-production 24 balanced-bus in-line console has 24 to 68 inputs for 24, 32 or 48-track operation. Features include: 4-band fully parametric EQ and swept HP and LP filters switchable separately into the channel or monitor path, master status switching, 8 aux. sends, 8 stereo aux. returns, 2 stereo cue sends, plasma metering, optional dynamics module. Automation options include GML moving faders.
Price:
$120,846.00-$439,261.00 (with GML)

Mozart multi-track recording console features 32 balanced buses, 32 to 80 inputs, in-line or all-input type modules (or a combination), integral switch grouping computer for user-configurable master status switching, 4-band semi-parametric EQ, HP and LP filters, 16 aux. sends, 12 stereo returns, 4 floating patchable stereo faders, and is available as standard with the AMEK Supertrue automation system. System is optionally available with GML moving fader automation.
Price:
84,971.00-$397,231.00

APC1000 assignable production console has up to 128 all-input type modules, 48-bus. Channel switching and routing is all under the control of a central assignment keyboard. Three automation computers work separately and simultaneously: Recall (knob positions), Reset (switch settings), and GML moving faders. Synchronous Rest recalls switch settings against time code. Optional dynamics modules also come under automation control.
Price:
$301,975.00-$587,110.00

AMS INDUSTRIES INC.

UA8000 is a music recording and mixdown console. It has a minimum of 32 channels, a maximum of 72 channels, filtering, 4-band parametric, compressor, limiter, and AMS/Calrec TASC automation.
Price:
ranges from $150,000 to $300,000

'M' Series Virtual Console System is an assignable computer controlled broadcast TV/video production console with up to 128 channels, 8 stereo groups, 4 stereo outputs, 24 recording groups and 16 auxiliary groups. Other features include filtering, 4-band parametric EQ, optional compressor/limiter/expander/gate per channel, and automation with total instant reset of all settings plus moving faders.
Price:
$100,000 and up.

Logic 1 Digital Audio Mixer for music broadcast production applications, integrates with AMS AudioFile Digital Workstation. 10 channels with full EQ and compression per channel. Features include full dynamic touch sensitive automation with instant reset and linear motorized faders.
Price:
depends on configuration.

Edit 1 Digital Mixer has 10 channels with 4-band EQ and assignable dynamics RAM memory can hold 99 console settings.
Price:
$40,000

ARIES AMERICA

Apollo recording console is a fully modular console available in 16, 24, and 32 input mainframes. An 8-channel expander unit is also available. All Apollo consoles have balanced mic and line inputs, switching on all inputs, inputs offer 8 aux. sends and 4-band high and low mid sweep EQ with selectable shelving points.
Price:
16x8x16—$11,950.00
24x8x16—$15,950.00
32x8x16—$19,950.00
8-channel expander—$5,450.00

Astrid series is a sound reinforcement console that is fully modular. This series is available in 24 and 32 input mainframes. An 8-channel expander unit is available. All Astrid consoles have balanced mic and line inputs, 8 aux. sends and 4-band high and low EQ with selectable shelving points and a sweepable mid-range.
Price:
24x8x2—$13,950.00
32x8x2—$17,950.00
8-channel expander—$5,450

16-track recording console features electronically balanced mic inputs with 20 dB pads available. Equalizers are +/- 14 dB switchable to 60 or 120 Hz, sweepable from 350 Hz to 7 kHz, and switchable from 6 kHz to 12 KHz. There are 8-group outputs and 16 monitor/FX returns-8 with 2-band EQ. There are 4 aux. sends per channel, switchable pre/post. Available with either 16 or 24 inputs.
Price:
16x8x16—$6,895.00
24x8x16—$8,595.00

Mix-Rack has the performance and sound characteristics of the 16-track console with the added flexibility of a rack mount. The basic unit comes with 10-inputs, 4-output buses and 8-monitor returns. It allows mixing of 18 channels on mixdown, 14 of them with EQ. A rack mount expander is available offering 14 additional inputs for a total of 32 channels on mixdown. There are 4 aux. sends, a 3-band EQ with sweepable mid-range and switchable high and low frequencies.

Price:
$3,450.00
Expander—$3,250.00 (14 channels)

AUDIO LOGIC

SC-610 is a rack mounted 7-input by 1-output mic/line mixer. It has 4-mic only inputs, 2-mic/line inputs with phantom power, a 1-aux. input with clipping indicators on each input, is one-rack space high, and has a master level control with bass and treble tone controls and a 6 LED headroom indicator.
Price:
$450.00

SC-611 is a rack mounted, 7-input expansion module for the SC-610. It has 4-mic only inputs, 2-mic/line inputs with phantom power, and 1-aux. input with clipping indicators on each input, is one-rack space high, and conveniently cascades into the SC-610 via mix bus inputs and outputs to add additional input capabilities to the SC-610.
Price:
$400.00

AUDIO-TECHNICA U.S., INC.

AT4462 is a stereo field production mixer. Its features include 4-input channels, 1 and 2 mono pannable, 3 and 4 true stereo on a dual concentric central, headphone with cue system, phantom power, slate tone and mic, limiter adjustable (mono or stereo) built in IFB system, 3-tone oscillator, low cut on all channels, switchable, level warning system in operator phones when approaching clipping. Has Cordura™ carrying bag.
Price:
$1,395.00

BIAMP SYSTEMS

MAXXAM is a line level mixer with 8 standard input channels and 8 stereo input channels. Other features are: 4 sends, 4 stereo returns, no channel EQ, patch insert points on each channel (stereo inserts on stereo input channels). Front panel switch interrupts main outputs to allow return bus number 4 to act as a 2-track tape return for tape monitor, or for playing recorded break music in a live set up.
Price:
$1,699.00

RACKMAX is a 16-channel input rack mount mixer. Features include 48-volt phantom power switchable on each channel, 100mm faders, complete solo system, 3-band EQ and 3 sends per input channel, LED ladder metering. Available is and optional Integral Digital Reverb with 16 programs, including digital readout and control.
Price:
$2,099.00
with IDR—$2,349.00

883RX is a compact mixer with 8 inputs. The output section is organized into main, monitor and two submasters. Features include 3-band EQ on each input, 10-segment switchable LED ladder metering, floating and balanced outputs, 4 transistor discrete front end, phantom power is standard, rotary faders. Optional IDR.
Price:
$1,099.00
with IDR—$1,349.00

Legend Series are modular, inline recording consoles available in models ranging from 12 to 32 inputs and from 8 to 24 submasters in two frame sizes. Features include 3-band semi-parametric EQ with true shelving filters on high and low filter, 12-segment high intensity LED meters on every channel, fully switchable, complete control room and studio communications, playback and tape monitoring systems, 48-volt phantom power switchable on each channel, 100mm faders with Penny & Giles 3220 conductive plastic faders optional, stereo monitor send and 4 effects/cue sends.
Price:
2016—$ 8,599.00
3216—$13,499.00
3224—$13,999.00

CARVIN CORPORATION —See our ad on Cover III

MX842 is a stereo powered mixer designed for sound reinforcement and multi-keyboard applications. It has a stereo 400 watt MOSFET amplifier, stereo effects loop, dual graphic EQs.
Price:
$749.00 sold directly only.

MX1222S is a stereo sound reinforcement/performance console with 12 input channels and 800 watts of internal MOSFET power. It also features 3-band EQ, 2 monitor sends and 1 effects send per channel and patchable graphic EQs.
Price:
$1499.00 sold directly only.

MX1644 is a recording/sound reinforcement console with 16-input channels and 4 sub groups. Its features include 4-band channel EQ, 4 aux. buses, 2 effects returns.
Price:
$1995.00 sold directly only.

MX2488 is a recording console with 24 inputs and 8 sub groups. Features include 3-band parametric EQ, 4 aux. buses, 2 effects.
Price:
$3995.00 sold directly only.

DOD ELECTRONICS CORPORATION

R 855 is a rack mounted, 4-input, stereo output mixer, one-rack space high, with pan pots, headphone output, master level control, an effects send/receive loop, and a clipping indicator.
Price:
$279.95
with XLR—$299.95

ELECTRO-VOICE

8108 is a rack mounted mono mixer with four outputs.

8208/8212/8216 are 2-bus 8, 12, or 16 channel sound reinforcement consoles. Features include equal headroom in all stages, gain-calibrated level controls for visual check of mixer input stages, transient performace not slew-rate or power bandwidth limited under any operating conditions from 20-20,000 Hz.

8408/8416/8424/8432 are 4-bus sound reinforcement consoles with 8, 16, 24, or 32 inputs. Features include rear-panel mic inputs including 48 V phantom power, channel patching and output jacks on each input panel, flourescent bargraph metering, talkback, panning, solo, and subgrouping.

BK832/BK1232/BK1632/BK2432 are stereo sound reinforcement consoles with 8,12,16, and 24 inputs, with 4 outputs. Features include channel insert send and return, input trim, high, mid, low equalization, panning, PFL assign, LED-bar level meters, subgroup patching send and return left and right phantom power, sub, main and monitor output master faders.

FOSTEX —See our ad on Cover II

260 Multitracker is a 4-track cassette mixer/recorder with 6 inputs. It has independent stereo bus, 2 mono buses, 3.75 in./sec. tape speed, Dolby C, parametric equalization, and true rolling punch-ins.
Price:
$1,195.00

160 Multitracker is a 4-track cassette mixer/recorder with 4 channel simultaneous recording and accessory patch points.
Price:
$840.00

X-30 is a 4-track cassette mixer with Dolby B and C noise reduction. The mixer section is 4x2 dedicated sub-mixer for overdubbing and bouncing tracks.
Price:
$499.00

460 is a multi-track cassette mixer capable of synchronization with video recorders. The mixing section contains 8 inputs, 4 bus outputs, dedicated stereo mixer for the 4-channel bus, selectable monitoring, switchable LED bar graph metering and accessible patch points for flexible system interface. The recorder section features true 2-speed transport (separate record/equalization circuits for 1.87 and 3.75 in./sec.), Dolby B and C noise reduction, 2-position autolocate, search to zero, auto repeat and SMPTE/EBU synchronization capability.
Price:
$2,495.00

FURMAN SOUND, INC.

Rackmount mixers MM-4A and MM-8A feature four inputs, mono (MM-4A) or stereo (MM-8A) outputs, pan pots on each MM-8A input, effects bus with send and return jacks, stereo aux. inputs with RCA jacks and level control, low-cut buttons on each input (-3 dB at 100 Hz), master fader, headphone amp with front panel jack and volume control. This is a compact utility mixer for sound reinforcement or recording use. "-B" models have balanced ins with both phone and XLR connectors and mic/line switches. "-BP" models are the same as "-B," plus 48-volt phantom powering on all inputs and phantom power switch.
Price:
$335.00 (MM-4A)
$375.00 (MM-4AB)
$405.00 (MM-4ABP)
$395.00 (MM-8A)
$435.00 (MM-8AB)
$465.00 (MM-8ABP)

GOTHAM AUDIO CORPORATION (AUDIO DEVELOPMENTS, LTD.) —See our ad on page 4

AD 160 is a mono engineering mixer with the following features: 3 mic inputs, 1 line input (all are balanced), 1 line output (balanced), limiter on line output, phantom/A-B powering on mic inputs, 1 kHz oscillator, talkback mic, VU or PPM meter, monitor output, battery or a.c.(with optional a.c. adapter).
Price:
$1,765.00

AD 260 stereo engineering mixer is the same as above with the following differences:
4 mic/line inputs with pan pots, 2 line outputs, 2 limiters on line output (linkable for stereo), 1 stereo aux. input.
Price:
$2,500.00

AD 145 Pico mixer has 4-8 mic/line inputs with pan pots (all are balanced), 2 line outputs (balanced), 3-band equalization on each input, phantom/A-B powering on mic inputs, 1 kHz oscillator, talk back mic, monitor output, cue input. Battery or a.c (with optional adapter)
Price:
$4,400.00 to $6,775.00

AD 062 multi-mixer has 4-16 mic/line input modules with pan pots (balanced), 2 line outputs, 3-band equalization on each input, phantom/A-B powering on mic inputs, auxiliary send on each input module. Options include stereo line input module to replace mic/line input module, dual auxiliary return module, communications module, stereo compressor/limiter module. Standard features include monitor level, selector controls, master auxiliary sends, A + B mixdown, battery or a.c (with optional a.c. adapter).
Price:
$6,000.00 to $20,000.00

HILL AUDIO INC.

Multimix is 16 channels in a rack and can be used as a 16x4x2x1 or 12x4x2x1 or 16x2x1 console. With 16 mic/line inputs, 4 subgroups, 3-band EQ with defeat, direct outputs on all channels, 2 auxiliary sends (expandable), 100mm faders, 48-volt phantom power, 4 RIAA equalized inputs, this console is designed for recording, broadcast or sound reinforcement.
Price:
$2399.00

Soundmix is a semi-modular console available as 24x4x2x1 or 16x4x2x1. Designed for recording, broadcast or sound reinforcement, features include: 16 or 24 mic/line inputs, 4 subgroups, 4 auxiliary sends, 4 auxiliary returns, 4-band EQ, 100mm faders (Alps or Noble), 48-volt phantom power, direct outputs on all channels and subgroups, insert points on all channels, subgroups and masters, PFL system, and 12 segment LED display on subgroups and masters.
Price:
$4499.00 (24-channel Noble faders)
$4999.00 (Alps faders)
$3499.00 (16-channel Noble faders)
$3899.00 (Alps faders)

Stagemix is a 12x6 rack-mount monitor console with a 12-channel zero splitter (built-in), 12 transformer balanced input channels with 3-band EQ, peak LEDs, PFL system and individual channel mutes. Other features include 6 transformer balanced output channels with 4-band EQ and EQ defeat, individual output mutes, 12-segment LED display, PFL and AFL system, TRS effects loop and the console is color coded for ease of operation.
Price:
$2999.00

Remix is a 24x8x16x2 console designed for 8 or 16-track recording or sound reinforcement as a 24x8 console with 8 effects returns. Features include 24 mic/line input channels with peak LEDs, 8 subgoups, 16-track monitoring, 6 auxiliary sends, tone oscillator, 2-track return, 4-band EQ with sweepable mids, 16 tape returns with EQ and aux. sends, direct outputs and insert points on all inputs, subgroups and masters.
Price:
$8999.00

Concept 200 and 400 Series are sound reinforcement and recording consoles. Available from 16x8x2 with an 8x8 matrix to 56x24x48x2, these consoles feature patch bays, Alps or P&G faders, 3-way speaker select, 2 types of EQ available (6-band with sweep filter or 4-band sweepable), 8 or 12 auxiliary sends using dual concentric or 4/6 sends switchable to 8/12, programmable mutes, A&B 2-track returns, d.c. controlled subgroups.
Price:
To specification

INDUSTRIAL RESEARCH PRODUCTS, INC.

DE-4013 automatic microphone mixer has modular construction in 2-channel increments. Its features include: up to 12 inputs in a single chassis, tandem connected chassis allows addition in 12 channel increments, 12V phantom power, level-matic AGC. Auxiliary output and chairman override are standard.
Price:
$3,395.00 (12 channels)

DE-4018 is the same as DE-4013 but includes DE-208 TEQ 9-band transversal equalizer module. 10 channels in single chassis before tandem connecting others.
Price:
$3,546.00 (10 channels)

DE-4014 is a mixer with 4-microphone input channels plus auxiliary line level input and auxiliary output. 12V phantom power is optional.
Price:
$1,467.00

DE-4016 is the same as the DE-4014 but with remote control via DE-207 remote control box. Remote functions are individual channel sensitivity and master gain control. DE-207 box includes ten feet of cable terminated with an amphenol 165-9 connector.
Price:
$1,860.00

DE-4024 is the same as the DE-4016 but includes both auxiliary gain control on both front panel and remote box. Auxiliary output is the mix of gated microphone inputs 2, 3, and 4 plus standard (ungated) microphone input 1 plus the auxiliary input.
Price:
$1,920.00

DE-4025 is the same as DE-4024 except the auxiliary output is a mix of standard ungated microphone inputs plus the auxiliary input.
Price:
$1,920.00

INDUSTRIAL STRENGTH INDUSTRIES

PM-1600 is a 16 channel stereo mixer with 2 built-in power amps (400 watts/285 watts), built-in 2/3-way crossover network, and built-in digital effects.
Price:
$2195.00 (factory direct)

INNOVATIVE ELECTRONIC DESIGNS, INC.

4000 automatic microphone mixing system is a modular, rack mount, automatic mixing system which allows custom configurations and programmable gain control. Each frame can hold up to 8 cards creating 32 inputs and offers optional computer control for room combining, and channel on/off.
Price:
On request

MITSUBISHI PRO AUDIO GROUP

SuperStar recording console has dual in-line I/O modules, 44 to 84 inputs, and 32/64 centrally assigned mixing buses. There are also 16 auxiliary send buses, 2 stereo outputs, modular bolt together aircraft frame, field expandable, optional meter overbridge for peripheral and in-line devices, with VU, or 60 segment LED bargraph meters. Selectable top panel plug-in equalizers, selectable top panel plug-in preamplifiers, selectable plug-in VCA. Audio, VCA, automated and intelligent digital faders are all standard as are Compumix PC automation, 20 Mbyte, moving fader automation, 30 Mbyte.
Price:
$139,000.00 to $354,000.00

Westar 8000 recording console has dual in-line I/O modules, 20 to 52 inputs, 24 mixing buses, 8 auxiliary send buses, 2 stereo outputs, a modular bolt together aircraft frame, field expandable, and VU, or 60 segment LED bargraph meters. Selectable top panel plug-in equalizers, selectable top panel preamplifiers, selectable plug-in VCA, Audio, VCA automated and intelligent digital faders are all standard, Compumix PC automation, 20 Mbyte moving fader automation, 30 Mbyte.
Price:
$70,000.00 to $225,000.00

Westar 8300 film re-recording console is available in 16 to 72 inputs. Features include: 8, 16, or 24 mixing buses, 10 auxiliary send buses, 3-channel pan bus, and a 2-channel pan bus film monitor system with 8x4 to 24x8 matrix select systems with dedicated monitor format buttons, recorder and bus/film pushbutton control panel(s) 8 to 24 tracks, reassign, transfer key, and mono and stereo composite modules, multi-track pre-dub input modules, a modular bolt together aircraft frame, field expandable, VU, or 60 segment LED bargraph meters, selectable top panel plug-in equalizers, and selectable plug-in VCA, Audio, VCA, automated and intelligent digital faders are standard as are Compumix IV film automation, 85 Mbyte, 1 to 4 sections, moving fader automation.
Price:
$89,000.00 to $600,000.00

NEVE

"V" series console is an automated "flying fader" multi-track recording console for the music, video post and film industries, available in 36, 48, 60 or 72 inputs, including the Formant Spectrum Equalizer, mic/dynamics unit and 8 mono/4 stereo auxiliaries. Additional benefits include a centrally positioned monitor path status indication to enable rapid console status checks and choice of metering options.

DTC (Digital Transfer Console) for compact disc mastering and is used for the preparation of compact disc master tapes, the unit also provides total digital mixing and processing capabilities. All console parameters can be instantly reset under SMPTE control, also permitting the user to select or mix either of 2 stereo digital inputs and 1 stereo analog input with manual or auto crossfade from AES/EBU or 1610/1630 inputs to compatible outputs.

8232 console for TV production, post-production and multi-track recording has 32 mic/line input channels with 24 mixing buses and optional stereo reverb returns. Each channel features the Formant Spectrum Equalizer, 4 mono auxiliary sends and 1 stereo cue send, and is "flying fader automated."

542 console comprise a range of small compact audio mixing consoles for broadcast and post-production applications. With 8, 12 or 16 input channels and 2 playback inputs, the console provides 2 main stereo program outputs with a mono facility, 2 auxiliary outputs and a comprehensive stereo/mono monitoring system.

Necam 96 is a computer-assisted moving fader automation system that is also available for fitting to non-Neve consoles, controlling up to 96 faders, and instinctive mixing, completely free grouping, unlimited mute groups, intelligent rollback. It keeps the pass as a virtual mix for review or update. Auto merge and merge-off-line capabilities.

Prism series is a range of rackmount units derived from the V series console comprising a 4U 19-inch rack with capacity for 10 modules that may be powered from an existing console or by a 2U power supply. The 2 modules are the Formant Spectrum Equalizer and the mic amp/dynamics unit comprises compressor/limiter/gate/expander.

PANASONIC/RAMSA

WR-S852 house sound reinforcement mixing console contains the following features: 52 mic/line inputs, all inputs and outputs are balanced, 8 groups, 8 aux. groups, L/R master, 8x10 matrix, 4-band sweepable EQ and high-pass on inputs, 300,000 operation, 100mm MRP faders, discrete design using hybrid ICs.
Price:
$33,000.00

WR-S840F stage monitor mixing console contains the following features: 52 mic/line inputs, all inputs and outputs are balanced, 18 discrete monitor mixes (18 outputs), 4-band sweepable EQ and high-pass on inputs, 300,000 operation, 100mm MRP faders, discrete design using hybrid ICs.
Price:
$35,000.00

WR-8428 production/post production mixing console is also suited for sound reinforcement. It contains the following features: 28 mic/line or stereo inputs, separate 24-input tape monitoring, 4 groups, 4 aux. groups, L/R and mono master, 4x8 matrix, 3-band sweepable EQ and high-pass on inputs, 300,000 operation, 100mm MRP faders, discrete design using hybrid ICs.
Price:
less than $20,000.00 depending on configuration.

WR-8616 production/post production mixing console is also suited for sound reinforcement. It contains the following features: 16 mic/line or stereo inputs, separate 16-input tape monitoring, 4 groups, 4 aux. groups, L/R and mono master, 3-band sweepable EQ and high-pass on inputs.
Price:
less than $11,000.00 depending on configuration.

WR-T820B 8/16-track recording mixing console contains the following features: 20 mic/line inputs, separate 20-input tape monitoring, up to 48 separately mixable inputs for mixdown, 8 groups, 4 aux. groups, L/R master, 3-band sweepable EQ and high-pass on inputs.
Price:
$8,500.00

WR-S216 sound reinforcement mixing console has the following features: 16 mic/line inputs, last two inputs accept mic or stereo source, "A/B" (L/R), effect, monitor, send and mono master outputs, 3-band EQ (mid-sweepable) on inputs.
Price:
WR-S216—$2,700.00
WR-S212—$2,200.00 (12-input)
WR-S208—$1,600.00 (8-input, rack mount)

WR-133 sound reinforcement mixing console has the following features: 8 mic/line inputs, L/R, effect, monitor, mono master outputs, 2-band EQ on inputs, rack mount option.
Price:
$1,200.00

WR-M10 sound reinforcement/music rack mixer has the following features: 4 mic/line inputs, 2 stereo inputs and each stereo input can select 4 different sources for a total of 8 stereo inputs (2 are RIAA phono), automatic music mute on page (defeatable), L/R, effect, mono master outputs, 2-band EQ on mic inputs, 2-band EQ on left and right master outputs.
Price:
$900.00

PEAVEY ELECTRONICS

Mark VIII sound reinforcement console is available in 24 and 36 channels. It also has the following features: 8 submasters, 8 aux. sends, 4-band sweepable EQ, 8 outputs with LED output level indication, 3-band EQ on aux. returns, stereo L/R outputs, totally modular design.
Price:
To be announced

Mark IV sound reinforcement console has the following features: 16 and 24 input channels, four submasters, mono (sum) output, transformer balanced inputs and outputs, 2-monitor sends, 1 effect send, PFL, 4-band active EQ, built-in heavy duty flight case.
Price:
$2,499.00 (16-channel)
$2,999.00 (24-channel)

Mark III sound reinforcement console has the following features: 16 and 24 input channels, 2 submasters, mono (sum) output, stereo, transformer balanced inputs and outputs, 2 monitor sends, 2 effects sends, PFL, 4-band active EQ, built-in heavy duty flight case.
Price:
$1,999.99 (16-channel)
$2,499.99 (24-channel)

MS 2421 sound reinforcement console has the following features: 24 channels, stereo and mono outputs, 3-band active EQ with sweepable mid, 2 monitor sends, 2 effects sends, four 9-band graphic EQ (patchable), electronic crossover, built-in delay, PFL, two submasters.
Price:
$2,699.99

MS-1621 has the same features as the MS 2421 but only 16 input channels.
Price:
$2,099.99

SRC-421 sound reinforcement console has the following features: 16-channels, 4 submasters, stereo, 3 aux. sends (one switchable); 3-band EQ with sweepable mid, LED output level indication, channel patching, PFL, 4 outputs.
Price:
$1,699.99

MD-11 sound reinforcement console is configured in 8, 12 and 16 channel inputs. Its features include 2 submasters, stereo, 1 monitor send, 1 effects send, 3-band EQ with sweepable mid, LED output indication, PFL, channel patching, dual outputs.
Price:
$749.99 (8-channel)
$949.99 (12-channel)
$1,1149.99 (16-channel)

701R rack mountable sound reinforcement mixer includes the following features: 4-band active EQ, 7 input channels, stereo L/R outputs, sum output, 1 effects send, 1 monitor send, LED output indication, channel patching, transformer balanced inputs and outputs.
Price:
$649.99

SECK (JBL PROFESSIONAL PRODUCTS)

SE1882 is a recording console, 18-channel mixer with track select, monitor section with 3 aux., 1 pre, 2 post fader aux., 3-band EQ with mid sweep, solo switch, stereo bargraph meter, headphone sockets.
Price:
$3,475.00

SE1282 is the same as the SE1882 above except 12-channel mixer.
Price:
$2,650.00

SE62 is a 6-channel live mixing console with balanced mic and line inputs, individual channel insert points, 3-band EQ with mid sweep, panning, solo control, stereo bargraph meter, 4 aux. sends, headphone level control, twin headphone sockets.
Price:
$1,095.00

SE122 is the 12-channel mixer version of the SE62.
Price:
$1,675.00

SE182 is an 18-channel live mixer with the same features of the SE122 above plus 48-V phantom power.
Price:
To be announced.

SE 242 is the 24-channel version of the SE182.
Price:
$3,275.00

SHURE BROTHERS, INC. —See our ad on Cover IV

M68A and M68-FCA have male XLR inputs and female XLR inputs respectively and are designed as a general purpose audio mixer. Features include four mic input channels (each switchable for high and low impedance, aux. level input, master volume control, and high and low impedance outputs).
Price:
$220.00

M268 is designed as a general purpose audio mixer with 4 transformers coupled low-Z mic inputs and 4 high Z 1/4-inch phone jack inputs, aux. input, switchable phantom power, mix bus, peak indicator and master volume control.
Price:
$300.00

FP31 is a compact field production audio mixer with the following features: 3 inputs, 2 outputs each switchable for mic or line level, VU meter with lamp, adjustable limiter with LED indicator, tone oscillator, built-in slate tone and mic tone and mic, headphone jacks, battery or a.c. power, phantom and A/B power.
Price:
$990.00

FP32 is a compact stereo field production audio mixer, a stereo version of the FP31. It includes all FP31 features plus center-detented stereo pan pot for each input channel and concentric clutched stereo master gain control.
Price:
$1,350.00

M267 is a field production audio mixer with the following features: 4 transformer-balanced inputs, switchable limiter, phantom power, VU meter, headphone jack with level control, LED peak indicator, tone oscillator, mic/line switches on each input and output, low-cut switches on each channel, battery or a.c. power.
Price:
$520.00

FP42 is a stereo version of the M267 and includes all the features of the M267 plus stereo pan pots for each channel and concentric clutched master volume control.
Price:
$990.00

FP51 is an audio mixer/compressor with the following features: 4 transformer-balanced inputs, 1 output, gated-memory compressor, phantom power, tone oscillator, pull pot cuing, 1/4-inch and mini headphone jacks with level control, low-cut filter switches, dual range VU meters, a.c. or battery power.
Price:
$940.00

Prologue 200M is a general purpose audio mixer with the following features: 4 XLR inputs, mic and aux. level outputs, on-off switch with LED indicator.
Price:
$117.00

SOLID STATE LOGIC

SL 4000 G series master studio system is a multi-track music recording and mixing system available with 24 to 72 input/output channel, full dynamics processing and G Series EQ. Also G Series Studio Computer and Total Recall TM.
Price:
On request

SL 5000 M series audio production system is a stereo broadcast, on-air, continuity and post-production system based on a modular cassette structure. Up to 96 channels, accepts G Series Studio Computer, Total Recall TM and Instant Reset TM.
Price:
On request

SL 5000 M series film post-production system is a modular, cassette based film console system, available in configurations for ADR/Foley, premixing, music scoring, final mix, video post-production, and multi-operator film dubbing. Up to 96 channels, optional moving fader automation, accepts G Series Studio Computer, Total Recall TM and Instant Reset TM.
Price:
On request

SL 6000 G series stereo video system is a stereo music, video and teleproduction system with 24 to 72 input/output channels, full dynamics processing and G Series EQ. Also G Series Studio Computer and Total Recall TM automation.
Price:
On request

01 Digital production centre is an integrated digital audio recording, mixing, and editing system, including digital dynamics and EQ. Includes edit suite, 8-channel mixes and hard disc store. Suitable for stereo program mastering, production and post-production.
Price:
On request

Screensound is a digital audio editing, mixing and recording suite for video and film post-production, with full VCR/VTR and film reproducer interfaces. Pen, tablet and monitor display user interface, 6-channels, optical disc and hard disc stores. Quantel Harry TM interface.
Price:
On request

SONY COMMUNICATIONS PRODUCTS COMPANY, PROFESSIONAL AUDIO DIVISION

MXP-3056 VF audio recording/remixing console is intended primarily for use in recording studios. It has 56-channels which allows for interfacing with the Sony PCM-3348 digital audio multi-track recorder. Each input/output module features modular equalizers and mic/line pre-amplifiers. The Audio Group Master (AGM) function allows for audio grouping on the ACN bus as well as conventional in-line operation.
Price:
up to $190,000.00 depending on configuration

MXP-3036 VF is designed with a vacuum fluorescent (VF) light meter that displays various selectable scales including VU, BBC Peak, Din Peak, Nordic Peak and a d.c. scale. This d.c. scale indicates fader position in the automated version of the MXP-3036 VF. The automated version includes Version 2.0 software and optional wild faders that permit a user to increase the number of effects in a mix.
Price:
up to $110,000.00 depending on configuration

MXP-3000 series also has the MXP-3036 and the MXP-3040 in addition to the two consoles mentioned above. Features common to all models include user configureable with a choice of 5 different equalizers, 4 different mic pre-amplifiers and a hard disc automation.
Price:
MXP-3036—up to $99,000.00 depending on configuration
MXP-3020—up to $40,000.00 depending on configuration

MXP-2000 series audio consoles includes the MXP-2016 (featuring a 20-module frame size), MXP-2026 (featuring a 30-module frame) and MXP-2-36 (up to 40-module frame size). These consoles are designed for broadcast and post-production applications. Features include 2 independent stereo outputs, 4 stereo internal monitor inputs and video editor, interface capability.
Price:
MXP-2016—up to $17,500.00 depending on configuration
MXP-2026—up to $25,000.00 depending on configuration
MXP-2036—up to $35,000.00 depending on configuration
MXP-29 is an 8-channel audio mixer. It can be controlled from the Sony BVE-900 editing control unit. Features include trim control for each balanced mic/line input, built-in 3-band EQ and VU meters with 15 segments of LEDs.
Price:
$3,849.00

MXP21 8-channel audio mixer is designed for audio-for-video applications. It features built-in 3-band EQ and low-cut filter, as well as 2-way operation (a.c. or external d.c. 12V).
Price:
$1,899.00

MX-P61VU is a 12-channel audio mixer. It is equipped with 12 mic/line inputs and 4 line outputs. Features include: built-in 1 kHz test tone for precise level setting, high-cut and low-cut filters for convenient bandwidth limiting, and a.c./d.c. operation.
Price:
$10,675.00

SOUNDCRAFT ELECTRONICS (JBL PROFESSIONAL PRODUCTS)

TS12 is a 24-track in-line recording console with each channel having electronically-balanced mic/line preamps, 4-band parametric EQ and 6 aux. sends with source selection. Each group module includes separate stereo effects return. Can be fitted with TS12 Automation System. VU or LED metering optional.
Price:
range from $33,100.00 to $51,100.00

6000 is 16- or 24-bus recording console with optional MIDI computer retrofit. Each input has 6 independent sends and 4-band EQ with 2 sweepable mids. Features include PFL and true solo-in-place, low crosstalk routing matrix, silent electronic muting. Consoles available in 16-56 inputs.
Price:
ranges from $16,300.00 to $40,300.00

8000 is a live console featuring 48-band parametric EQ, 8 aux. sends, direct routing, VCA subgrouping, LED input metering or three-way panning option for theater use. Consoles range from 16- to 40-channel inputs.
Price:
ranges from $18,785.00 to $41,500.00.

200B is a recording consoles with balanced lin/mic inputs, assignable input channels directly to the four subgroup buses and stereo mix bus. Four aux. sends selectable as pre-EQ/pre fader, or post fader. Has 48V phantom power, individual channel inserts, balanced left/right output separate with control room output.
Price:
ranges from $2,575 to $10,370.00

SOUNDTRACS

ERIC production console is a 24 bus split configuration digitally routed console with a high analog specification that is available in 32, 40 and 48 input sizes. It has the following features: software controlled features which include routing, muting, mic/line switching, PFL, in place solo, and record ready function (facilitating automated drop ins), all may be synced to time code in addition the 24 groups duplicate the line inputs providing up to 72 inputs for mixdown.
Price:
starts as $175.00.00

In line series contains a 32 bus console primarily suited to multi-channel recording with functional facilities for mixing. Dual line inputs are provided in addition to a mic on each channel which contains 8 aux. sends, 6 mono and 1 stereo. Master muting facilitates the pre-programming of 2 groups of channel mutes. Mix noise is better than -82 dB. Patchbay is standard.
Price:
$71,499.00—36 input (loaded)
$80,799.00—48 input

PC series MIDI mixer is an "in-line" mixer available in 16 and 24 input configurations with 16 or 8 bus outs. It includes MIDI muting on all inputs, monitors, masters, and auxiliaries, which may be programmed or written in real time or as up to 100 patches. Dual line inputs enable effects to be returned to the channel allowing up to 56 channels on remix. External function allows changing of MIDI patches on remote equipment from console.
Price:
$8,299.00 PC 16/16
$10,799.00 PC 24/16

Trackmix is a stand alone fader and mute automation system for up to 64 audio channels, using retrofittable dbx VCAs. Four fader modes are available, record, playback, trim and isolate, allowing each channel to be in any mode at any time. Mix information is held in RAM for optimum speed and may be automatically or manually saved on 3.5-inch floppy disc.
Price:
starts at $19,500.00

STUDER/REVOX

C279 mixer is for remote recording. It features 6 inputs, mono balanced mic, mono balanced line, stereo unbalanced line, 2 outputs plus direct outputs, 2-band EQ.
Price:
$2,799.00

961/962 is for remote recording and broadcast production. Features include up to 16 inputs, 4 master outputs, 2 aux. inputs, an optional editor interface is available, 3-band EQ on each input, compressor/limiter on outputs. A variety of metering options are available.
Price:
ranges from $13,000 to $21,000

963 is a recording console available with 16 to 56 inputs, up to 8 subgroups, 2-4 masters, 3-band EQ on each input, compressor/limiter on outputs. Alternate input modules, metering, monitor mixers, and machine remotes are available.
Price:
ranges from $44,000 to 90,000.

970 is a 9-24 input console specially configured for radio/TV on-air production. There are 2 stereo outputs and mono sum, 3-band EQ on inputs and compressor/limiters on outputs. Alternate metering options are available.
Price:
ranges from $22,000 to $40,000

900 series consoles can be configured for on-air TV, production use, or multi-track recording. Features included 12-60 inputs with 3- or 4-band EQ, mono or stere inputs, stereo master, GML or Audio Kinetics or Studer VCA automation is available, outputs have compressor/limiters.
Price:
$80,000 and up.

SUNN

MX4100 Series stereo sound reinforcement mixing console is available in 8, 12, and 16-channel versions with 3-band channel EQ, 2 aux./eff. sends, phantom power, gain and pan control.
Price:
MX4108—$749.99
MX4112—$949.99
MX4116—$1,119.99

MX4200 Series stereo sound reinforcement mixing console is available in 8, 12, and 16-channel versions with 3-band channel EQ, 3 aux./mon. sends with pre-post assignments, gain and pan controls dual in/out channel patch, cue, phantom power.
Price:
MX4208—$1,099.99
MX4212—$1,299.99
MX4216—$1,599.99

PX2100 Series stereo powered sound reinforcement mixing console is available in 8, 12, and 16-channel versions with 250 wattsx2 at 4 ohm, 3-band channel EQ, 3 aux./mon. sends, dual in/out channel patch, dual 10-band graphic EQ.
Price:
PX2108—$1,549.99
PX2112—$1,749.99
PX2116—$1,999.99

TASCAM, TEAC PROFESSIONAL DIVISION —See our ad on page 3

M-700 Recording/Production console contains the following features: 40-channel I/O console, 32 groups, quad (dual stereo) mix buses, 12 aux. send buses (2 stereo, 8 mono), 4-band EQ, high/low sweep, mid high, mid low parametric, variable high-pass filter, 3 group mutes, integral patchbay and producers desk, automation ready.
Price:
$69,999.00

M-600 Series modular recording/production console contains the following features: 16 group, stereo mic modular console that may be configured with any combination of mono or stereo input modules, and a choice of single or dial monitor modules. It has 4-band EQ, 8 aux. send buses, 2-dedicated effects returns. Minimum configuration is 16 mono inputs with 16 tape returns. Maximum configuration 32 stereo inputs with 32 tape returns. Third party automation available.
Price:
$9,999.00 to $27,000.00

MM-1 Keyboard mixers has 1 stereo line input, and 16 mono line inputs each with a trim control with 40 dB of gain. There is included 2-band EQ on each channel, small rack mount or table top operation, 4 effects sends, 4 stereo returns, direct outs on each channel, individual channel mute and solo, group mutes through 99 scene presets, MIDI control of individual and group muting.
Price:
$1,099.00

M-06ST Compact mixer has 6 stereo line input channels, 2-band EQ, 1 aux. send bus, 8 mic inputs, RIAA EQ on line inputs 1 and 2, 1 effect return.
Price:
$499.00

M-200 Series PA/recording consoles are compact 1 group stereo mix consoles with 8, 16, or 24-input channels, 3-band EQ, 2 aux. send buses, 3 effect returns, PFL and full monitoring. M-208 and M-216 have 8 tape returns. The M-224 has 16 tape returns. The M-208 can be rack mounted.
Price:
$1,199.00 to $2,599.00

M-3000 Series recording/PA consoles are a compact 4 group stereo mix consoles. There is a choice of 8, 12 or 20-input channels, 8 tape returns in the dedicated monitor section, separate mic/line trim controls, 3-band EQ, 4 aux. send buses, 2 effect returns, full solo system with AFL or PFL, phantom power.
Price:
$2,999.00 to $4,599.00

M-500 Series recording/PA consoles are 8 group stereo mix consoles with a choice of 12 or 20-input channels, 16 tape returns in the dedicated monitor section, 4 aux. send buses, 3-band sweep EQ, individual mic/line trim controls, separate solo and PFL controls, individual channel and aux. mute controls.
Price:
$4,499.00 to $6,999.00

YAMAHA MUSIC CORPORATION OF AMERICA, PROFESSIONAL AUDIO DIVISION

PM3000-24/32/40C Series is available in 24, 32 or 40 inputs and has the following features: 8 group buses, 8 aux. buses (each pre/off/post) and separate stereo bus, VCA assignable grouping with 8 submasters with automation interface, 8 bus muting master system with safety override. The XLR inputs are differentially balanced with 34 dB trim and 5 position pad for optimizing gain structure. There is also a 4-band parametric EQ with variable high-pass filters on each input plus 2-band EQ on the 4 stereo aux. returns, a 11x8 mix matrix, insert point selectable in/out on each input, extensive cue and solo system, a comprehensive talkback system with full intercom capability, VU metering, phantom power individually selectable on each mic input.
Price:
PM3000-24—$33,500.00
PM3000-32—$38,500.00
PM3000-40C—$44,500.00

PM2800M-32/40-C series is available with 32 or 40 inputs and the following features: 8 group buses, 4 aux. buses, signal assignment for the input channels via level control, 8 master mute groups with mute assign switches, 4 matrix mixes with level for all 8 channels, stereo L & R level and master. The XLR inputs are differentially balanced with 34 dB trim and 3-position pad for optimizing gain structure. There is also a 4-band parametric EQ with variable high-pass filters on each input, extensive cue and solo system, comprehensive talkback system with full Intercom capability, VU metering system, phantom power individually selectable on each mic input.
Price:
PM2800M-32—$31,500.00
PM2800-40C—$36,000.00

M916 has 16 inputs, each switch selectable to mic or line. In addition there is 3-band, 9 frequency EQ with post insert jacks, 11 mix buses—2 program, 2 foldback, 2 echo, 4 mix matrix and 1 cue. 5x4 mix matrix= x, cue switch on each input for preview/solo via headphones, headphone feed is stereo in PGM and mono in echo/foldback, 5 illuminated VU meters with peak overload, switchable to indicate stereo program, matrix foldback, echo, and cue levels. The main input/outputs are transformer-isolated XLRs, and there is switchable phantom power on each mic input.
Price:
$6,595.00

DMP7 digital mixing processor has all digital mixing and signal processing with analog inputs/outputs. Features include 3 onboard DSPs (Digital Signal Processors), digital 3-band parametric EQ on each channel, preset memories: 32 internal, 67 external via supplied RAM cartridge, motorized multi-function faders, digital stereo output compressor, MIDI control of preset changes and parameter manipulations, 4 bar-graph meters and LCD parameter read-out. A digital cascade input/output ties multiple units together.
Price:
$4,225.00

DMP7D digital mixing processor has all digital mixing and signal processing with digital inputs and outputs, 3 on-board DSPs, digital 3-bank parametric EQ on each channel, preset memories: 32 internal, 67 external via supplied RAM cartridge, motorized multi-function faders, digital stereo output compressor, MIDI control of preset changes and parameter manipulations, 4 bar-graph meters and LCD parameter read-out. A digital cascade input/output ties multiple units together.
Price:
$5,995.00

DMP11 digital mixing processor has all digital mixing and signal processing with analog inputs/outputs, 2 on-board DSPs, digital 3-band parametric EQ on each channel, preset memories: 32 internal, 67 external via supplied RAM cartridge, digital stereo output compressor, MIDI control of preset changes and parameter manipulations, 4 bar-graph meters and LCD parameter read-out. A digital cascade input/output ties multiple units together. This unit is a rack mount size.
Price:
$2,395.00

M406 sound mixer has among its features: 6 channels, with 3-band EQ and 6-position input level controls, high gain (84 dB) for full output, stereo program output with L & R master controls, echo/effects send bus with master send control, 2 effects inputs, each with level and pan control, dual illuminated VU meters with peak indicators, right VU meter and headphone output switchable to monitor program or echo output, rack mountable with front panel power switch.
Price:
$1,375.00

MC1204/1604/2404 series is available with 12, 16 or 24 inputs. It has the following features: 4 program mix buses, 2 effects buses, 2 foldback buses and a cue bus. Each input features a pad, gain control and peak LED for precise gain matching. There is also 4-band EQ with the two mid-bands featuring quasi-parametric control, foldback 1 and 2, and ECHO 1 and 2 strappable re/post EQ, channel ON (mute), is post cue for proper cue monitoring, complete talkback system, illuminated VU meters, each with peak LEDs, phantom power available for mic inputs.
Price:
MC1204—$2,695.00
MC1604—$3,295.00
MC2024—$4,395.00
MC2404M—$4,395.00 (stage monitor)

ADDRESSES

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Orange, CT 06477

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10500 W. Reno
Oklahoma City, OK 73128

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10815 Burbank Blvd.
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AMS Industries, Inc.
PO Box 31864
Seattle, WA 98103

Aries America
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Unit E
Torrance, CA 90502

Audio Logic
See DOD Electronics

Audio-Technica US, Inc.
1221 Commerce Dr.
Stow, OH 44224

Biamp Systems
14270 NW Science Park
Portland, OR 97229

Carvin Corporation
1155 Industrial Ave.
Escondido, CA 92025

DOD Electronics
5639 South Riley Ln.
Salt Lake City, UT 84107

Electro-Voice
600 Cecil St.
Buchanan, MI 49107

Fostex
15431 Blackburn Ave.
Norwalk, CA 90650

Furman Sound, Inc.
30 Rich St.
Greenbrae, CA 94904

Gotham Audio Corporation
1790 Broadway
New York, NY 10019-1412

Hill Audio Inc.
5002 N. Royal Atlanta Dr. #B
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Industrial Research Products
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Industrial Strength Industries
13042 Moore St.
Cerritos, CA 90701

Innovative Electronic Designs
9701 Taylorsville Rd.
Louisville, KY 40224

Mitsubishi Pro Audio Group
225 Parkside Dr.
San Fernando, CA 91340

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Berkshire Industrial Park
Bethel, CT 06801

Panasonic/Ramsa
6550 Katella Ave.
Cypress, CA 90630

Peavey Electronics
711 A St.
Meridian, MS 39301

Seck (JBL Professional
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see Soundcraft

Shure Brothers Inc.
222 Hartrey
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Solid State Logic
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Sony Professional Audio
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Teaneck, NJ 07666

Soundcraft Electronics
8500 Balboa Blvd.
Northridge, CA 91329

Soundtracs/Samson
Technologies
485-19 S. Broadway
Hicksville, NY 11801

Studer Revox
1425 Elm Hill Pike
Nashville, TN 37210

Sunn
6024 SW Jean Rd, Bldg. D
Lake Oswego, OR 97035

Tascam/Teac Corporation
7733 Telegraph Rd.
Montebello, CA 90640

Yamaha Corporation
6600 Orangethorpe Ave.
Buena Park, CA 90620

REVIEW

How can you make your recordings sound more like a record? How can you get your sound the way you want it?

With these provocative questions, so all important as they may be, three new video tapes have been released that go a long way to offering answers.

First Light Video Publishing (374 N. Ridgewood Pl., L.A., CA 90004) is the company producing the tapes. So far, there are three. The first released at the end of last year is "Shaping Your Sound With Microphones;" the other two newer ones are "Shaping Your Sound with Equalizers, Compressors, and Gates," and "Shaping Your Sound with Reverb and Delay."

Each tape is over 80 minutes long, and is packed with information. An eight-page printed guide about the use of the respective tapes also comes with each one. But information alone can be almost useless.

Fortunately, First Light Video has done their homework. They are first to realize that a video tape, like a book, can not be fully digested in one or only a few run-throughs. The tapes are therefore broken into logical sections, permitting the student to first get a total overview, and then go at it in detail, section by section.

Tom Lubin narrates the tapes and demonstrates most of the technology and practical techniques and use of many typical products. Lubin is a practicing recording engineer and producer. He also conducted a Fostex-sponsored lecture series that toured the U.S. recently.

More than his impressive credentials, Lubin also has the ability, so important in this application, to communicate both an enthusiasm for what he is doing, and a knowledgeable approach to the more technical things that also must come across. It is rare to find such a skill at both the theoretical and practical ends of recording, and with communication. Lubin is a delight to watch.

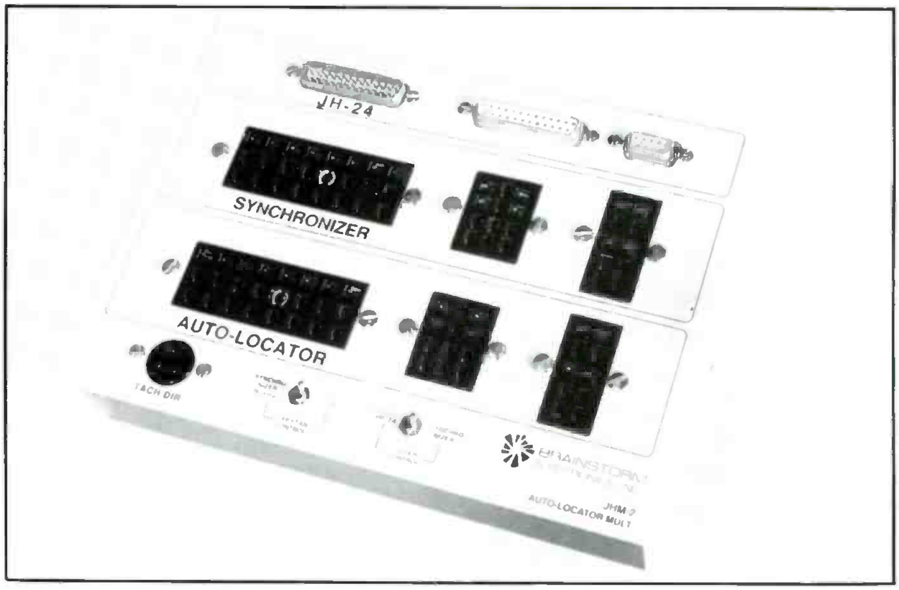
You really need a top-notch video tape deck to fully benefit from the sound on these tapes. Thy are recorded in "hi-fi stereo" and that was amply evident in the many product and musical demonstrations that abound on these tapes.

If anything at all was distracting, it was in a sense caused by the high sonic quality of the tapes. Many of Lubin's voice-overs have scene-by-scene sonic shifts of quality. Sometimes he is close mic'ed, the next surrounded by reverb.

No matter, these are excellent basic tutorials. And for the more advanced, they just might turn out to be excellent refresher courses as well.

Two versions of the tapes are available. A pro version for schools includes a 40-page workbook and a lifetime free tape replacement—for $119 per tape. An individual-user version with the 8-page guide is $59.95 per tape. L.Z.

New Products



INTERFACE

• The JHM-2 Auto Locator Mult is an interface for the MCI JH24 tape recorder that allows its auto-locator and a synchronizer to be used simultaneously. Housed in a metal chassis, it has two sets of "Remote, Auto Locator, and Capstan" connectors identical to those found on the rear panel of the JH24. It also features a lifter control switch and a capstan control switch and includes a set of JH24 interface cables.

Mfr.- Brainstorm Electronics, Inc.
Price- $495.00

Circle 60 on Reader Service Card

POWER AMPLIFIER

• The PPA600 is a two-channel power amplifier that delivers 225 watts of continuous output power per channel into 8 ohms, or 350 watts into 4 ohms, both channels driven. In the switchable mono bridge mode, the unit delivers 650 watts of continuous power into 8 ohms. It has heavy duty output terminals, toroidal power transformer and the input inrush current limiting circuit and relay-protected speaker connections. Additional safeguards are found in the independent rectifiers and capacitors for each channel. The unit is fully protected against electrical and thermal overload and short-circuited output terminals. If a d.c. voltage appears at the output terminals, the speakers will be disconnected. In the event that the heat sink temperature reaches 70 degrees Celsius, the input signal will be attenuated and, again, the speakers will be disconnected. When the temperature has dropped to 60 degrees Celsius, the speakers will be reconnected and the amplifier gain will gradually increase to the previous level. All of these actions are confirmed by the LED indicators located on the front panel. The unit features 16 MOSFET metal power transistors (eight per channel), with a total dissipation of 1600 watts to produce peak output currents in excess of 40 amps. This enables the power amplifier to drive low impedance and/or highly reactive speakers. The power devices are mounted on an efficient cooling tunnel running between the front and rear panels. For further cooling efficiency, it features a thermally controlled, variable speed fan.

Mfr.- Ortofon
Price- $2000.00

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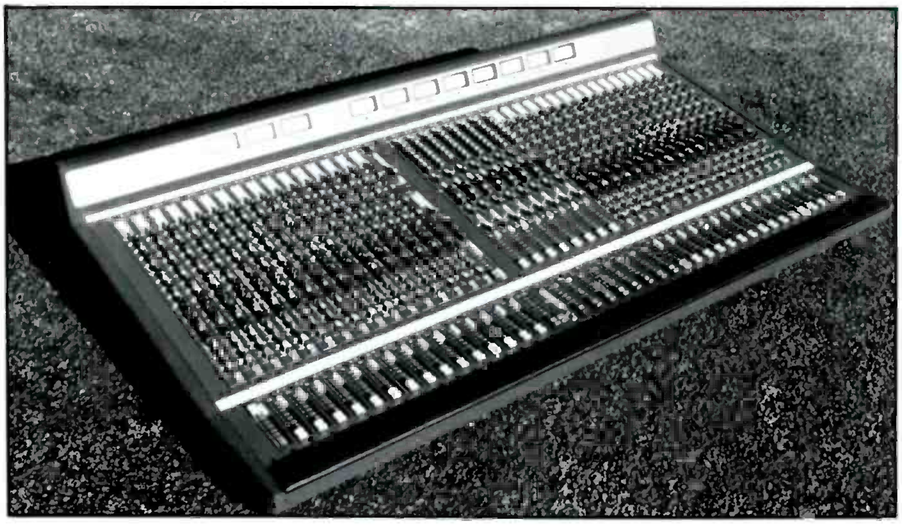
IN-LINE CONSOLE



• The SPA console is available in two frame sizes, allowing 32 or 48 channel versions. Access points to all audio, VCA and mute buses are available on rear panel connectors, enabling two consoles to be linked providing up to 96 inputs. The unit can be reconfigured via a single switch on each group to double as a monitor desk. The input channels feature a four band equalizer, with fully parametric mid and frequency sweepable high and low bands switchable between bell and shelving response. Eight auxiliary sends, all switchable pre/post fader, selectable pre/post equalizer are fitted. Each channel also has a ninth send, the local auxiliary. Channels may be routed to any of the eight audio subgroups and the stereo masters, and each can be controlled by one of the eight VCA groups and the eight programmable mute groups. Output metering is via eleven moving coil VU meters.

Mfr.- Soundtracs

Price- approx. $54,000.00 (32 channel), $60,000.00 (48 channel)

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TEST DISC

• The Studio Reference Disc (SRD) is a compact disc containing a large variety of tests for all audio and A/V applications. It will convert any CD player into a versatile test generator. The CD format enables the user to quickly program any order or repeating tones, sweeps, bursts etc. It features 70 minutes of audio test tones and references, and includes two sets of 17 different sine wave calibration tones, four bands of different sine wave sweeps, pink and white noise: steady state, bursts of different lengths, 1/3-octave and full bandwidth, CAVEAT header, two different control room tests, polarity check, left/right tests, musical pitch references, both digitally generated and acoustic piano, acoustic piano listening tests, and much more.

Mfr.- Prosonus

Price- $49.95

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AURAL EXCITER

• The Aural Exciter Type III features two modes of noise reduction and Aphex's "SPR" Spectral Phase Refractor, for improved bass clarity and openness. It is a single-ended processor that can be applied at any point in the audio chain, and requires no decoding. The unit's processing will recreate and restore missing harmonics, producing natural brightness, clarity and presence. It provides its audio enhancements without raising the noise floor of any nominally noise-free source. Two modes of noise reduction make this possible. Mode A functions as a linear sidechain expander with variable threshold. Mode B is an all-new noise reduction method which actually "erases" source audio noise while the unit simultaneously enhances the signal.

Mfr.- Aphex Systems Ltd.

Price- $995.00

Circle 64 on Reader Service Card

FLOOR MONITOR

• The FS-212 is a two-way, biamped concert floor monitor containing two DL12X woofers and one DH1A 2-inch compression driver mounted on an HP64 constant-directivity horn. The cabinet is constructed of solid, 14-ply Baltic birch covered with black Ozite Super TNT carpeting, and has a cloth-covered steel grille. The 60x40 degree constant-directivity pattern enables tight, consistent performer coverage while allowing for higher gain-before-feedback levels. The unit is capable of producing sound pressure levels in excess of 130 dB with full power input. The ITT-Cannon EP-4-14 connector is used for the power input and EP-4-13 for the loop-thru connection.

Mfr.- Electro-Voice, Inc.

Price- $1995.00

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STEREO MIXER

• The MX842 is a professional stereo powered mixer designed to give high performance in a wide variety of applications. This eight-channel mixer has a powerful stereo 400 watt MOSFET amplifier, stereo effects loop, and dual graphic EQs. The amplifier can be split to run both mains and monitors, and the heavy duty power supply allows loading of up to four 8 ohm speakers.

Mfr.- Carvin Corporation

Price- $749.00

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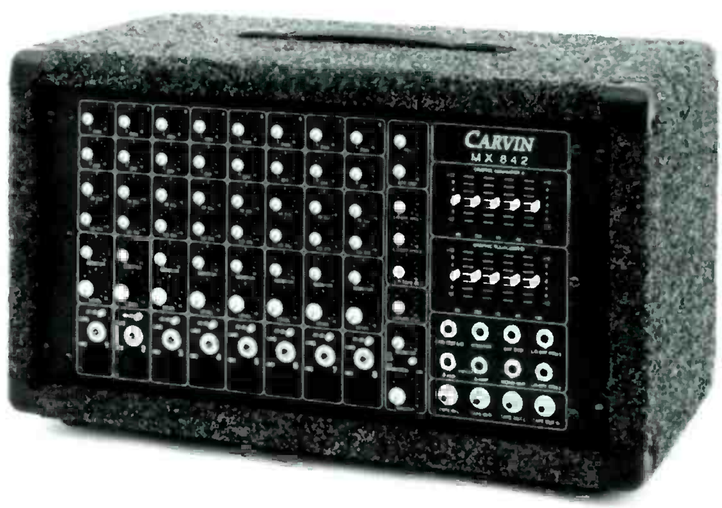

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People, Places... & Happenings

• **John Reitz** has been named recording mixer at **Buena Vista Sound**, announced by **Jacobus Rose**, director of post-production services for Buena Vista Studios. In his new position, Reitz will be responsible for re-recording and mixing work on motion pictures and filmed television productions for Disney as well as outside producers. Reitz has over 100 feature and television films to his credit.

• **Audio Animation** announces the expansion of its engineering staff. Recently hired are **Steven Kurant**, Senior Development Engineer, **Neil Martin**, Senior Development Engineer, **Mark Schmid**, Software Engineer, **Richard Guay**, Hardware Engineer, **Jeff Berry**, Laboratory Manager, and **Melissa Stallings**, technical support.

• **Willi Studer AG**, Switzerland, has named **Tore B. Nordahl** President of **Studer Revox America Inc.** The decision was made public at the occasion of the AES convention in Hamburg, Germany, by **Bruno Hochstrasser**, President of Studer Revox America. Nordahl has held the position of Vice President and General Manager at Studer Revox prior to his promotion. Hochstrasser has been elected Chairman to the board of directors of Studer Revox America Inc.

• **Telex Communications, Inc.**, has announced a reorganization of its sales and marketing staff. **Dan Dantzler** was promoted to vice president of sales for the company's professional audio, aviation and RF communications products as well as OEM sales. **Don Mereen** has been appointed executive director of marketing. This is a newly created position with responsibility for new product and market planning, marketing services and technical customer service. **Ted Nemzek** was named senior director of sales for the audio/visual products group.

• **Electro-Voice** has appointed **Michael V. Torlone** as market development manager, music electronics. The announcement was made by **Jim Long**, E-V director of marketing. Torlone will be responsible for the continued growth of E-V's music-market electronics business.

• At **Altec Lansing, Frank McMullen** has been appointed Director of Manufacturing by **Dave Merrey**, President. His responsibilities include purchasing, production control, production, shipping, quality control and customer service.

• **Neve** has moved its Western Regional sales and support office into expanded facilities in the new Mercedes-Benz building in Hollywood, CA. The new office is a modern complex that includes complete conference facilities, an enlarged parts department including a greater inventory of supplies for the entire range of Neve consoles, an expanded lab and service area, and a complete demonstration area.

• **University Sound, Inc.** has announced the appointment of **Robert W. Sandell** as company president. The announcement was made by **Robert Pabst**, head of Mark IV Audio. Sandell is responsible for all of the operations of University's business and for implementing University's strategic plan as one of the Mark IV companies.

• **Sonopress**, an audio cassette duplication company based in Atlanta, GA, has expanded with six million dollars in a new plant addition and equipment for cassette shell manufacturing. The company is building an adjoining 70,000 square foot facility to its Weaverville, NC plant. It will provide space for warehousing, offices, and cassette shell production equipment.

• **Lindsay Allen** has been promoted to Manager of Professional Audio Tape Products at **Ampex Corporation** Magnetic Tape Division. He succeeds **Warren Simmons** who is retiring after more than twenty years at Ampex. Lindsay is responsible for developing worldwide product strategies, implementing product marketing programs, and developing product support programs for the company's line of professional audio tape.

• **WaveFrame Corporation** has announced the promotion of **Gus Skinas** to the position of Senior Product Manager, Professional Audio, and the appointment of **Dave Frederick** to the position of Product Manager, Music Applications. Gus Skinas will focus on development of WaveFrame products for professional audio applications, especially recording and editing. Dave Frederick will focus of development of AudioFrame applications for music composers and editors. Also announced were the appointments of **Mike Buffington** as Director of Field Engineering, **Craig Damon** as European Support Manager, and **Arthur "Midget" Sloatman** as Systems Engineer.

• **Solid State Logic** has announced the appointments of **Piers Plaskitt** as Chief Executive Officer, and **Dave Collie** as Product Development Manager. In addition, Solid State Logic's Los Angeles office has undergone a major expansion of its facilities, with the provision of an additional 1,500 square feet of space for extra office, stores, and demonstration studio use.

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